



► Deployment Guide for the
Polycom® VVX™ 1500 D
Business Media Phone

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Polycom, Inc.
4750 Willow Road
Pleasanton, CA 94588-2708
USA

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About This Guide

This guide explains how to deploy the Polycom VVX 1500 D Business Media Phone in an H.323 environment. In this environment, you will be able to conduct both voice and video calls and conferences.

In this guide, you will find everything you need to set up the Polycom VVX 1500 D phone in an H.323 environment. Be sure to verify that your network is prepared for configuring the phone.

The following related documents for the Polycom VVX 1500 D phone are available:

- Quick Start Guide, which describe how to assemble the phone
- User Guide, which describe the basic and advanced features available on the phone
- SIP Administrator's Guide, which describes how to configure, customize, manage, and troubleshoot Polycom SoundPoint IP, SoundStation IP, and VVX phone systems
- Technical Bulletins, which describe workarounds to existing issues and provide expanded descriptions and examples
- Release Notes, which describe the new and changed features and fixed problems in the latest version of the software

For support or service, please contact your Polycom® reseller or go to Polycom Technical Support at <http://www.polycom.com/support/voice/>.

Polycom recommends that you record the phone model numbers, software (both the BootROM and SIP), and partner platform for future reference.

Polycom VVX 1500 D MAC Address: _____

BootROM version: _____

SIP Software version: _____

Partner Platform: _____

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Overview

This guide provides information for system administrators who want to deploy the Polycom VVX 1500 D Business Media Phone in an H.323 environment. It also describes how to upgrade the phone's software when it is connected in an H.323 environment. The phone must be running SIP application 3.2.2 (phone software) or later.

The topics in this guide include:

- Setting up the Polycom VVX 1500 D phone in one of two ways:
 - Changing the phone's configuration through a web interface. Refer to [Modifying Phone's Configuration Using the Web Interface](#) on page 2-7.
 - Setting up a provisioning server and changing the configuration files to work in an H.323 environment. Refer to [Setting Up the Provisioning Server](#) on page 2-10, [Deploying Phones From the Provisioning Server](#) on page 2-12, and [Configuration Files](#) on page A-1.
- Troubleshooting any issues with the phone. Refer to [Troubleshooting the Polycom VVX 1500 D Phone](#) on page 3-1.

The topics in this chapter include:

- [Supported Phone Features](#)
- [Unsupported Phone Features](#)



Polycom strongly recommends that you connect the Polycom VVX 1500 D to the LAN using the included network cable. A LAN connection enables access to a provisioning server as well as a time server.

A provisioning server facilitates software upgrades, language support, configuration management, and diagnostic logging, which is helpful if troubleshooting is required. Access to a time server enables display of the current date and time on the phone's display.

For more information, refer to [Setting Up the Provisioning Server](#) on page 2-10.

Supported Phone Features

The following features are supported on the Polycom VVX 1500 D:

- Touch screen for data entry
- Six line appearances
- Place a call (to an extension or H.323 URL)
- Hold/Resume
- Consultative and blind transfer
- Video on/off
- Single-button voicemail dialing
- Message Waiting Indicator
- Shared line appearance (call forking - no busy lamp field)
- LDAP/Active Directory integration
- Local call directory
- Placed Call/Missed Call/Received Call lists
- Three-way audio conference with management options (join, split, mute)
- Mid-call video/audio negotiation
- Electronic hookswitch with supported headsets
- Enhanced feature keys and configurable soft keys
- Audio call recording and playback (USB flash drive required)
- Digital picture frame (USB flash drive required)
- Multiple language support (provisioning server required)
- Full interactive browser

Unsupported Phone Features

The following features are not supported on the Polycom VVX 1500 D:

- Transferring or forwarding H.323 calls
- Joining of remaining parties when H.323 conference host exits
- Multiple H.323 servers
- SRST (SIP server redundancy)
- Far-end camera control
- Conveying microphone mute status between endpoints

Setting Up the Polycom VVX 1500 D Phone

The Polycom VVX 1500 D phone is designed to be used like a regular phone on a public switched telephone network (PSTN). It is a business media phone that combines advanced business telephony, one-touch video, real-time presence, and integrated business applications into one seamless, lifelike communications experience.

The phone provides a powerful, yet flexible IP communications solution, delivering excellent video and voice quality. The high-resolution color graphic display supplies content for call information, multiple languages, directory access, and system status. The phone supports advanced functionality, including multiple call and flexible line appearances, HTTPS secure provisioning, custom ring tones, and local audio and video conferencing.

With the Polycom VVX 1500 D phone, you can now have connectivity to any native H.323 video environment using the H.323 protocol.

For the package parts list and instructions on how to assemble the phone, refer to the *Polycom VVX 1500 Quick Start Guide*, which is in the phone package.

This chapter provides basic instructions for setting up Polycom VVX 1500 D phones when managed in an H.323 environment:

1. Connect the phone to the local area network (LAN) using the supplied network cable. Refer to [Connecting the Phone to the Network](#) on page 2-2.

The phone will boot up once power is supplied. This process may take some time.

2. Decide how you want the phones to behave. Refer to [Configuring the Phones](#) on page 2-3.
3. Once the bootup process is complete, you can change the phone's configuration through a web interface. Refer to [Modifying Phone's Configuration Using the Web Interface](#) on page 2-7.

4. If you choose to manage a group of phones through a provisioning server, you need to:
 - Set up a provisioning server and download the appropriate files. Refer to [Setting Up the Provisioning Server](#) on page 2-10.
 - Create configuration files to support the required features on the phones. Refer to [Deploying Phones From the Provisioning Server](#) on page 2-12.



For installation and maintenance of the Polycom VVX 1500 D phone, the use of a provisioning server is strongly recommended. This allows for flexibility in installing, upgrading, maintaining, and configuring the phone. Configuration, log, language, and directory files are normally located on this server. Allowing the phone write access to the server is encouraged.

The phone is designed such that, if it cannot locate a provisioning server when it boots up, it will operate with internally saved parameters. This is useful for occasions when the provisioning server is not available, but is not intended to be used for long-term operation of the phones.

Because of the large number of optional installations and configurations that are available, this section focuses on one particular way that the SIP application and the required external systems might initially be installed and configured in your network.

5. (Optional.) Upgrade the phone software after changing the configuration files and deploying the phones.

Out of the box, the Polycom VVX 1500 D phone is running BootROM 4.2.1 and SIP 3.2.2. To determine the BootROM version running on the phone, press the **Menu** key, select **Settings > Status > Platform > Phone**, and then scroll down to BootROM. To determine the SIP application version running on the phone, press the **Menu** key, and then select **Settings > Status > Platform > Application > Main**.

Refer to [Upgrading the Phone's Software](#) on page 2-16.

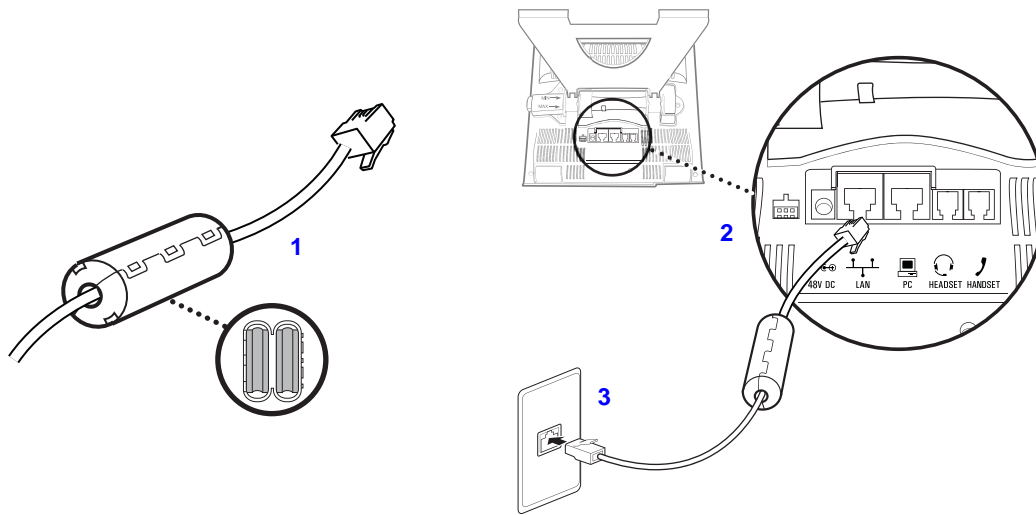
For more information on configuring the Polycom VVX 1500 D phone, refer to the latest *SIP Administrator's Guide*, which is available at http://www.polycom.com/support/video/business_media_phones/vvx1500d.html.

Connecting the Phone to the Network

Note

Connecting the Polycom VVX 1500 D phone to the network is optional, but highly recommended (for the reasons discussed at the beginning of this chapter).

The following figure shows how to connect the phone to the network.



To connect the Polycom VVX 1500 D phone to the network:

1. Clamp a ferrite clamp around the cable.
2. Connect the 25-foot Network Cable to the LAN jack on the phone.
3. Connect the other end of the 25-foot Network Cable to an available LAN port on the wall or hub/switch device port.

Note

AC power adapters are not included in the basic phone package. Using the supplied network cable, the phone can be powered from a Power over Ethernet (PoE) compliant switch or hub. Power over Ethernet is the default connection for the phone.

Configuring the Phones

Before you can allow users to place and answer calls you must make some basic configuration changes.

This section provides information for making configuration changes for:

- [Time and Date Display](#)
- [Browser](#)
- [H.323 Protocol](#)

To troubleshoot any problems with the Polycom VVX 1500 D phones on the network, refer to [Troubleshooting the Polycom VVX 1500 D Phone](#) on page 3-1.

For more information on all of the configuration parameters that you can change, refer to the latest *SIP Administrator's Guide* at http://www.polycom.com/support/video/business_media_phones/vvx1500d.html.

Time and Date Display

The phone maintains a local clock and calendar. Time and date can be displayed in certain operating modes such as when the phone is idle and during a call. The clock and calendar must be synchronized to a remote Simple Network Time Protocol (SNTP) timeserver. The time and date displayed on the phone will flash continuously to indicate that they are not accurate until a successful SNTP response is received. The time and date display can use one of several different formats and can be turned off.

For instructions on turning off the time and date display, refer to [Polycom VVX 1500 D Display](#) on page 3-2.

Browser

The Polycom VVX 1500 D phone supports a full browser. This can be launched by pressing the **App** key or it can be accessed through the **Menu** key by selecting **Applications**.

Two instances of the Browser may run concurrently:

- An instance with standard interactive user interface
- An instance that does not support user input, but appears in a window on the idle display

Configuration changes can be performed centrally at the provisioning server or locally. For detailed information on configuration file settings, refer to [Microbrowser <mb/>](#) on page A-6.

H.323 Protocol

Telephony signaling support via the H.323 family of protocols enabling direct communication with H.323 endpoints, gatekeepers, call and media servers, and signaling gateways is supported on the Polycom VVX 1500 D phone.

SIP and H.323 signaling can be supported at the same time, including bridging both types of calls during multi-party conference calls. Automatic detection of the correct or optimal signaling protocol is available when dialing from the contact or corporate directories. While SIP supports server redundancy and several transport options, only a single configured H.323 gatekeeper address per phone is supported. H.323 gatekeepers are optional, but if available, they must be used. If a gatekeeper is not configured or unavailable, calls can still be made if so configured.

Support of the SIP protocol for telephony signaling can be disabled on the Polycom VVX 1500 D such that all calls would be routed via the H.323 protocol.

Video calls are supported to the following Polycom endpoints/bridges/call servers (or gatekeepers)/media servers:

Make/Model	Protocol	Software Version
Polycom RMX 1000™	H.323	SW 1.1.1.8787
Polycom RMX 2000™	H.323	SW 4.0.2.7
Polycom HDX 9000™ series	SIP/ISDN/H.323	SW 2.5.0.6
Polycom HDX 8000™	SIP/ISDN/H.323	SW 2.5.0.6
Polycom HDX 7000™	SIP/ISDN/H.323	SW 2.5.0.6
Polycom HDX 6000™	SIP/ISDN/H.323	SW 2.5.0.6
Polycom HDX 4000™	SIP/ISDN/H.323	SW 2.5.0.6
Polycom VSX™ 8000	SIP/ISDN/H.323	SW 9.0.5.1
Polycom VSX™ 7000s and VSX™ 7000e	SIP/ISDN/H.323	SW 9.0.5.1
Polycom VSX™ 6000 and 6000a	SIP/ISDN/H.323	SW 9.0.5.1
Polycom VSX™ 5000	SIP/ISDN/H.323	SW 9.0.5.1
Polycom VSX™ 3000	SIP/ISDN/H.323	SW 9.0.5.1
Polycom V700™	SIP/ISDN/H.323	SW 9.0.5.1
Polycom V500™	SIP/ISDN/H.323	SW 9.0.5.1

Note

Refer to the *Release Notes* for the latest list of supported Polycom endpoints/bridges/call servers (or gatekeepers)/media servers and any supported third party products. Any issues (and possible workarounds) with any of the above-mentioned products are also documented in the *Release Notes*.

Configuration changes can be performed centrally at the provisioning server or locally. For detailed information on configuration file settings, refer to:

- [Protocol <volpProt/>](#) on page [A-3](#)
- [User Preferences <up/>](#) on page [A-4](#)
- [Video Settings <video/>](#) on page [A-4](#)
- [Call Handling Configuration <call/>](#) on page [A-5](#)
- [Security <sec/>](#) on page [A-5](#)
- [Registration <reg/>](#) on page [A-7](#)

The following information should be noted:

- If the phone has only the H.323 protocol enabled, it cannot be used to answer SIP calls.
- If the phone has only the SIP protocol enabled, it cannot be used to answer H.323 calls.
- If both SIP and H.323 protocols are disabled by mistake, the phone will continue to work as a SIP-only phone; however, the phone is not registered (you are able to send and receive SIP URL calls).
- The protocol that is used to place a call is stored in the placed call list of the user's phone.
- The protocol to be used when placing a call from the user's local contact directory is unspecified by default. The user can select SIP or H.323.
- The protocol that is used when placing a call from the user's corporate directory depends on the order of the attributes in the corporate directory. If only SIP_address is defined, then the SIP protocol is used. If only H323_address is defined, then the H.323 protocol is used. If both are defined, then the one that is defined first is used. For example, if dir.corp.attribute.4.type is **SIP_address** and dir.corp.attribute.5.type is **H323_address**, then the SIP protocol is used.
- By default, the user is presented with protocol routing choices when a call could be placed with more than one protocol from its current context. The user must choose between SIP and H.323 to place a call by pressing the appropriate soft key.
- Calls made using H.323 cannot be forwarded or transferred.
 - The **Transfer** and **Forward** soft keys are not displayed during an H.323 call on a Polycom VVX 1500 D phone. The **Forward** soft key is not displayed on the idle display of a Polycom VVX 1500 D phone if the primary line is an H.323 line.
 - If a Polycom VVX 1500 D user presses the **Transfer** soft key during an H.323 call on a Polycom VVX 1500 D phone, no action is taken.
 - The auto-divert field in the local contact directory entry is ignored when a call is placed to that contact using H.323.
 - If a conference host ends a three-way conference the call and one of the party is connected by H.323, that party is not transferred to the other party.

Modifying Phone's Configuration Using the Web Interface

You can make changes to the phone's configuration through the web interface to the phone. The factory default username and password for the web interface is **Polycom** and **456** respectively (the username is case sensitive).

Note

Blind transfer, call recording, picture frame, corporate directory (LDAP Integration), and phone server redundancy cannot be enabled/disabled through the web interface. You cannot change the displayed language through the web interface. You must make changes for these features through the configuration files. Refer to [Deploying Phones From the Provisioning Server](#) on page 2-12.

Note

Changes made through the web interface are written to the override file (highest priority). These changes remain active until **Settings > Advanced > Admin Settings > Reset to Default > Reset Local Config...** is performed.

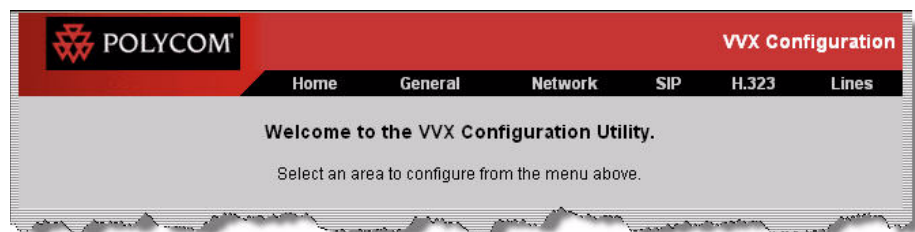
To configure the phone through the web interface:

>> Using your chosen browser, do the following:

- a To get the phone's IP address, press the **Menu** key, and then selecting **Status > Platform > Phone**. Scroll down to see the IP address.
- b Enter the phone's IP address as the browser address.

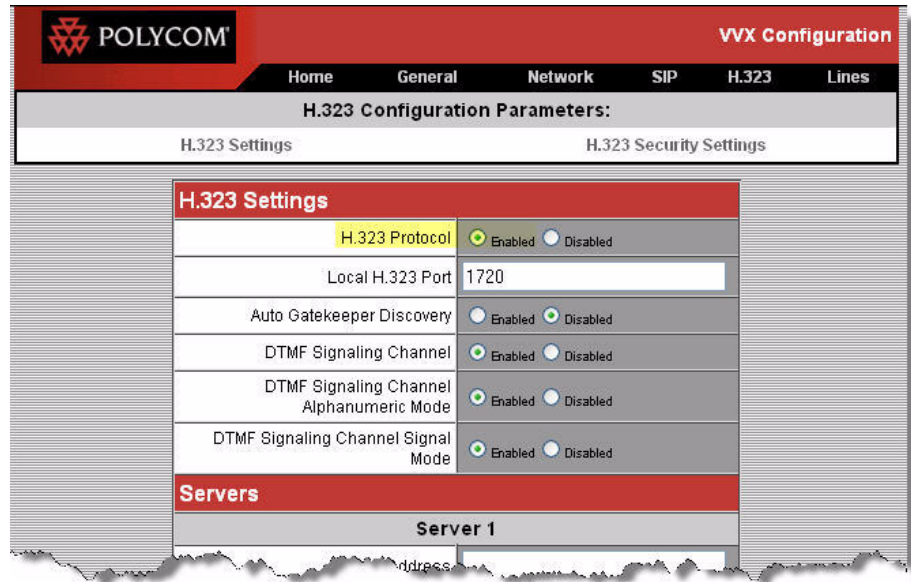
You will be prompted for the web interface username and password.

A web page similar to the one shown below appears.



- c To make changes to the phone's H.323 settings, select **H.323** from the menu tab.

A web page similar to the one shown below appears.



- d Make the desired configuration changes.
For example, verify that **H.323 Protocol** is enabled.
- e Select the **Submit** button.
The phone will reboot.

Note

You must select the **Submit** button—and reboot the phone—for each page where you make changes; otherwise, any changes you make on that page will be lost.

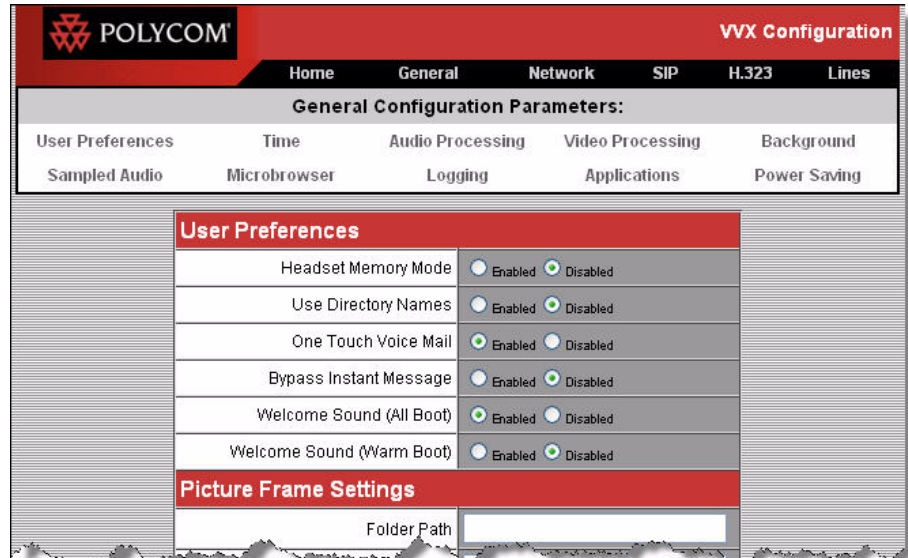
- f** After the reboot is complete, select **Lines** from the menu tab.
A web page similar to the one shown below appears.

The screenshot shows the Polycom VVX Configuration web interface. At the top, there is a red header with the Polycom logo and the text "VVX Configuration". Below the header is a navigation menu with tabs for "Home", "General", "Network", "SIP", "H.323", and "Lines". The "Lines" tab is selected, and the "Line Parameters:" section is visible. Below this, there are tabs for "Line 1", "Line 2", "Line 3", "Line 4", "Line 5", and "Line 6". The "Line 2" tab is selected, and the configuration form for Line 2 is displayed. The form is divided into several sections: "SIP Settings", "H.323 Settings", and "Identification".

Line Parameters:	
Line 1	Line 2
Line 2	
SIP Settings	
SIP Protocol	<input type="radio"/> Enabled <input checked="" type="radio"/> Disabled
H.323 Settings	
H.323 Protocol	<input checked="" type="radio"/> Enabled <input type="radio"/> Disabled
Gatekeeper Address	172.24.31.144
Port	
Registration Time	
Identification	
Display Name	Bob Smith
Address	0222465
Authentication User ID	
Authentication Password	
Label	0222465
Type	

- g** Make the desired configuration changes.
For example, under **Line 2**, enable **H.323 Protocol**, set the **Gatekeeper Address** to **172.23.31.144**, set **Display Name** to **Bob Smith**, set **Address** to **0222465**, and set **Label** to **0222465**.
- h** If you make any changes, scroll down to the bottom of the section.
- i** Select the **Submit** button.
The phone will reboot.

- j After the reboot is complete, select **General** from the menu tab.
A web page similar to the one shown below appears.



- k Make the desired configuration changes.
For example, under **Call Settings** (scroll down on page shown above), change the **Preferred Protocol** for Protocol Routing to **H.323**.
- l If you make any changes, scroll down to the bottom of the section.
- m Select the **Submit** button.

You can now instruct your users to start making H.323 calls.

Setting Up the Provisioning Server

The protocol that will be used to transfer files from the provisioning server depends on several factors including the phone model and whether the BootROM or SIP application stage of provisioning is in progress. By default, the phones are shipped with FTP enabled as the provisioning protocol.

You can use any computer on the network as the provisioning server (providing the application image and configuration files for the Polycom VVX 1500 D phone). For more information, refer to [Setting Up an FTP Server](#) on page B-1.

For organizational purposes, configuring a separate log file directory, override directory, contact directory, and license directory is recommended, but not required. The different directories can have different access permissions. For example, for LOG, CONTACTS, and OVERRIDES, allow full access (read and write) and for all others, read-only access.

Note The instructions below are written using FTP server as the provisioning server, but they also apply to any TFTP or HTTP server.

Note The factory default username and password for the phone is **PlcmSplp**.

To set up the provisioning server:

1. Create a directory to store the application image and the configuration files.

For example, C:\test\VVX1500 .

2. Extract all files from the distribution zip file to the newly created directory.

Maintain the same folder hierarchy.

The distribution zip file contains many files, but only the following are required:

- 2345-17960-001.sip.ld
- phone1.cfg
- sip.cfg
- 000000000000.cfg
- SoundPointIPWelcome.wav

Note If you want your users to be able to change the language displayed on their phone, all of the **SoundPointIP-dictionary.xml** files are also required.

Refer to the *SIP Release Notes* for a detailed description of each file in the distribution. The Release Notes can be found at http://www.polycom.com/support/video/business_media_phones/vvx1500d.html .

3. Start the FTP server and configure it to accept requests from the phone.

In the FTP server software on a computer, create a unique user with names and passwords. For example, username **user** with password **password**.

Refer to your server documentation to set up the server.

Deploying Phones From the Provisioning Server

You will need to create two configuration files for each phone – **phone1[MACAddress].cfg** and **[MACAddress].cfg** – and one configuration file for all phones in your installation – **sip[XXXX].cfg**.

For more information on why to create another configuration file, refer to the “Configuration File Management on Polycom VVX 1500 D Phones” white paper at

http://www.polycom.com/global/documents/support/technical/products/voice/white_paper_configuration_file_management_on_soundpoint_ip_phones.pdf.

Note

Use an XML editor to create/edit the configuration files. In this way, all configuration files that you create will be valid XML files.

If the configuration files are not valid, they will not load on the phone. An error message will be logged to the provisioning server.

For more detailed information on specific parameters in configuration files, refer to [Configuration Files](#) on page A-1.

Deploy Polycom VVX 1500 D phones by:

1. [Creating a Per-Phone Configuration File](#)
2. [Creating an Application Configuration File](#)
3. [Creating a Master Configuration File](#)
4. [Downloading the New Configuration](#)

Creating a Per-Phone Configuration File

To create a per-phone configuration file:

Note

Obtain the phone's MAC address from the barcoded label on underside of phone or on the outside of the shipping box.

1. Create a per-phone configuration file by using the **phone1.cfg** file from the distribution as a template.

For example, create `phone1_0004f2be80a6.cfg`

2. Add contents to the per-phone configuration file.

For example:

Configuration Path	Value
xml	version="1.0" encoding="UTF-8" standalone="yes"
#comment	Example Per-phone Configuration File
phone1	
reg	
reg.1.address	1301
reg.1.label	1301
reg.1.displayName	VVX1234
reg.1.protocol.SIP	0
reg.1.protocol.H323	1
reg.1.server.H323.1.port	1720
reg.1.server.H323.1.address	192.168.1.1
reg.2.address	1302
reg.2.label	1302
reg.2.displayName	VVX2345
reg.2.protocol.SIP	0
reg.2.protocol.H323	1
reg.2.server.H323.1.port	1720
reg.2.server.H323.1.address	192.168.1.1
call	
autoOffHook	
call.autoOffHook.1.enabled	1
call.autoOffHook.1.contact	1111
call.autoOffHook.1.protocol	H323

Creating an Application Configuration File

To create an application configuration file:

1. Create an application configuration file by using the **sip.cfg** file from the distribution as a template.

For example, create sip_VVXs.cfg

2. Add contents to the application configuration file.

For example:

Configuration Item	Value
version	"1.0" encoding="utf-8"...
SIP Application Configuration File	
volpProt.SIP.enable	0
volpProt.H323.enable	1
dialplan.digitmap	33xxH
up.manualProtocolRouting	0
up.manualProtocolRouting.softKeys	1
saf.1	SoundPointIPWelcome.wav
video.callRate	768
video.maxCallRate	896
call.autoAnswer.SIP	0
call.autoAnswer.H323	0
call.autoAnswer.micMute	1
call.autoAnswer.videoMute	0
call.autoAnswer.ringClass	4
call.autoRouting.preference	protocol
call.autoRouting.preferredProtocol	H323
sec.H235.mediaEncryption.enabled	1
sec.H235.mediaEncryption.offer	1
sec.H235.mediaEncryption.require	0
mb.main.home	http://apps.company.com/apps/MyApps.aspx
mb.main.idleTimeout	300
mb.idleDisplay.home	http://apps.company.com/apps/Idle.aspx
mb.idleDisplay.refresh	30
mb.limits.nodes	256
mb.limits.cache	200

Creating a Master Configuration File

To create a master configuration file:

1. Create a master configuration file by using the 000000000000.cfg file from the distribution as a template.

For example, create 0004f2be80a6.cfg

2. Add contents to the master configuration file.

For example:



Warning

The order of the configuration files listed in CONFIG_FILES is significant:

- The files are processed in the order listed (left to right).
- The same parameters may be included in more than one file.
- The parameter found first in the list of files will be the one that is effective.

This provides a convenient means of overriding the behavior of one or more phones without changing the baseline configuration files for an entire system.

Downloading the New Configuration

To download the new configuration to the phones:

1. To reboot the phones, press the **Menu** key, and select **Settings > Advanced > Reboot Phone...**

When prompted, enter the administrator password and press the **Enter** soft key.

The phone will reboot. The following messages will appear on the screen:

- Waiting for network to initialize
- Downloading new application
- Checking new application
- Saving new application
- Checking new application
- Loading application
- Running *<application filename>*
- Welcome

The reboot process will take some time.

2. Ensure that the configuration process completed correctly.

For example, on the phone, press the **Menu** key, and then select **Status > Platform > Application** to see the SIP application version and **Status > Platform > Configuration** to see the configuration files downloaded to the phone.

Monitor the provisioning server event log and the uploaded event log files (if permitted). All configuration files used by the provisioning server are logged. To make changes to the information that is logged, refer to [Event Logging <log/>](#) on page A-5.

You can now instruct your users to start making H.323 calls.

Upgrading the Phone's Software

To upgrade the phone's software, you will need to:

- Download the latest SIP application to the provisioning server from the Polycom VVX 1500 D support page—where you found this deployment guide. Refer to [Locating the Latest Software](#) on page 2-16.
- Download the new SIP application to the phone. Refer to [Downloading New Software to the Phone](#) on page 2-17.

For more information on upgrading the SIP application, refer to section "Upgrading the SIP Application" in the latest *SIP Administrator's Guide* at http://www.polycom.com/support/video/business_media_phones/vvx1500d.html.

Locating the Latest Software

To determine the latest software version, go to the VoIP SIP Software Release matrix on the Polycom Customer Support web site at http://downloads.polycom.com/voice/voip/sip_sw_releases_matrix.html.

Download the latest software and release notes from http://www.polycom.com/support/video/business_media_phones/vvx1500d.html to the provisioning server.

Downloading New Software to the Phone

To download the new SIP application to the phone, you need to:

- Extract all files from the distribution zip file as you did in [Setting Up the Provisioning Server](#) on page 2-10.
- Verify that the configuration files you created for [Deploying Phones From the Provisioning Server](#) on page 2-12 will work with the new SIP application. New and changed configuration parameters are described in the *Release Notes*.

Follow the instructions in [Deploying Phones From the Provisioning Server](#) to download the new SIP application; you will not need to change the master configuration files unless you change the names of the per-phone or application configuration files.

Troubleshooting the Polycom VVX 1500 D Phone

This chapter contains general troubleshooting information to help you solve any problems you might encounter when you use the Polycom VVX 1500 D phone in an H.323 environment. The phone can provide feedback in the form of on-screen error messages, status indicators, and log files for troubleshooting issues.

For detailed information on error messages, log files, testing the phone hardware and phone issues – along with likely causes and corrective actions – refer to the “Troubleshooting” chapter of the latest *SIP Administrator’s Guide*.

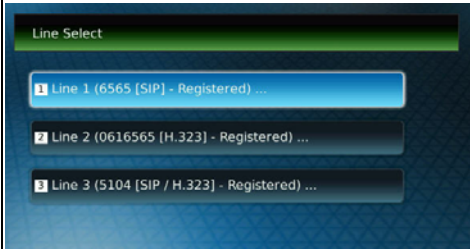
This chapter presents phone issues, likely causes, and corrective actions for the Polycom VVX 1500 D phone in an H.323 environment. Issues are grouped as follows:

- [Calling](#)
- [Polycom VVX 1500 D Display](#)

Review the latest *Release Notes* for the SIP application for known problems and possible workarounds.

Calling

Symptom	Problem	Corrective Action
The line icon shows an unregistered line icon.	The phone line is unregistered.	Verify that the appropriate configuration parameters are set correctly. Verify that the call server is functioning correctly.
The Polycom VVX 1500 D selects a lower-quality codec (H.261) when the called endpoint also supports H.264 .	The Polycom VVX 1500 D selects which audio/video codecs to transmit based on the far-end endpoint’s order of codecs in their supported list.	Verify that the appropriate configuration parameters are set correctly. Check the value of <code>voIpProt .SDP .answer .useLocalPreferences .</code>

Symptom	Problem	Corrective Action
The user could not place a call.	The call was placed using the incorrect protocol.	<p>Verify that the user is using a line that is registered to use the appropriate protocol. Do the following:</p> <ol style="list-style-type: none"> 1. Press the Menu key. 2. Select Status > Lines and then select the line you used to make the call. <p>The following figure shows three registered lines and the protocol used for each line.</p> 

Polycom VVX 1500 D Display

Symptom	Problem	Corrective Action
The time and date are flashing.	You have disconnected the phone from the network/LAN or there is no SNTP server configured.	<p>Do one of the following:</p> <ul style="list-style-type: none"> • Reconnect the phone to the network. • Disable the time and date display on the phone per the instructions below. <ol style="list-style-type: none"> 1. Press the Menu key, and then select Settings > Basic > Preferences > Time & Date. The Time & Date screen displays. Scroll to Disable, and then press Select soft key. 2. Press the Menu key again to return to the idle screen. The idle screen no longer displays the time and date.

Configuration Files

This appendix provides detailed descriptions of certain configuration files used by the SIP application running on the Polycom VVX 1500 D phone.

This appendix contains information on:

- [Master Configuration File \(MAC-address.cfg or 000000000000.cfg\)](#)
- [Application Configuration File \(sip.cfg\)](#)
- [Per-Phone Configuration File \(phone1.cfg\)](#)

The application configuration files dictate the behavior of the phone once it is running the executable specified in the master configuration file.

For detailed information—including permitted values, default values, and a brief description—on the complete set of configuration parameters available to you in the configuration files, refer to the “Configuration Files” appendix of the latest *SIP Administrator’s Guide*.

Caution

Configuration files should only be modified by a knowledgeable system administrator. Applying incorrect parameters may render the phone unusable. The configuration files which accompany a specific release of the SIP software must be used together with that software. Failure to do this may render the phone unusable.

Note

In the tables in the subsequent sections, “Null” should be interpreted as the empty string, that is, attributeName="" when the file is viewed in an XML editor.

To enter special characters in a configuration file, enter the appropriate sequence using an XML editor:

- & as &
- " as "
- ' as '
- < as <
- > as >

Note

You can also make changes to the configuration files through the web interface to the phone. Using your chosen browser, enter the phone's IP address as the browser address. For more information, refer to [Modifying Phone's Configuration Using the Web Interface](#) on page 2-7.

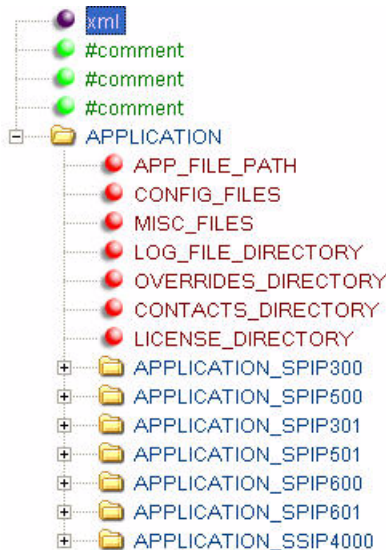
Changes made through the web interface are written to the override file (highest priority). These changes remain active until **Settings > Advanced > Admin Settings > Reset to Default > Reset Local Config...** is performed.

Master Configuration File

Master configuration files contain the following XML attributes:

- APP_FILE_PATH
- CONFIG_FILES
- MISC_FILES
- LOG_FILE_DIRECTORY
- CONTACTS_DIRECTORY
- OVERRIDES_DIRECTORY
- LICENSE_DIRECTORY

For example:



```

<?xml version="1.0" encoding="utf-8" standalone="yes"
Default Master SIP Configuration File
Edit and rename this file to <Ethernet-address>.cfg for each phone.
$RCSfile: 000000000000.cfg,v $ $Revision: 1.18 $

sip.ld
phone1_0004f2be80a6.cfg, phone1.cfg, sip_VVXs.cfg, sip.cfg
    
```


Application Configuration File

The configuration file **sip.cfg** contains SIP protocol and core configuration settings that would typically apply to an entire installation and must be set before the phones will be operational, unless changed through the local web server interface or local menu settings on the phone.

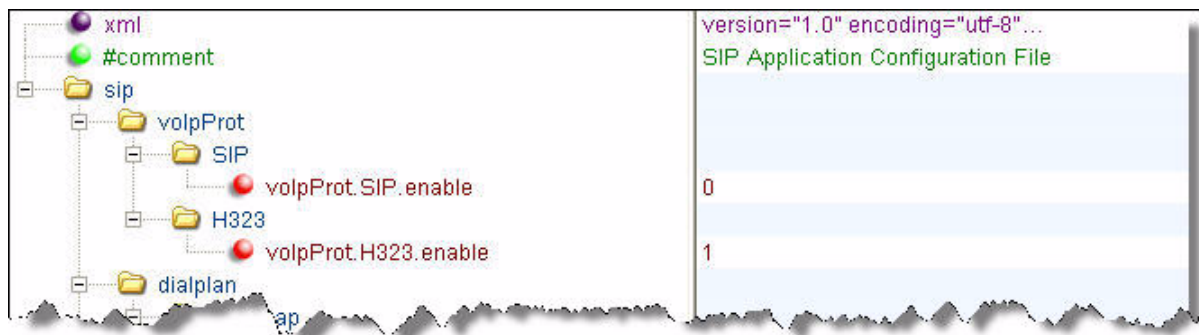
The important parameters for the Polycom VVX 1500 D phone include:

- **Protocol <volpProt/>** – Specify H.323 protocol, server, gatekeeper, and DTMF signaling parameters.
- **User Preferences <up/>** – Specify call routing parameters.
- **Video Settings <video/>** – Specify the video call rate.
- **Call Handling Configuration <call/>** – Specify call routing parameters and auto-answer parameters.
- **Event Logging <log/>** – Specify basic logging levels.
- **Security <sec/>** – Specify H.323 media encryption parameters
- **Microbrowser <mb/>** – Specify the Application browser home page, a proxy to use, and size limits.

Protocol <volpProt/>

Edit the voIpProt parameter as shown below:

- Enable the H.323 protocol.
- Disable the SIP protocol.



User Preferences <up/>

Edit the up parameter as shown below:

- Disable manual protocol routing.



Video Settings <video/>

Edit the video parameter as shown below:

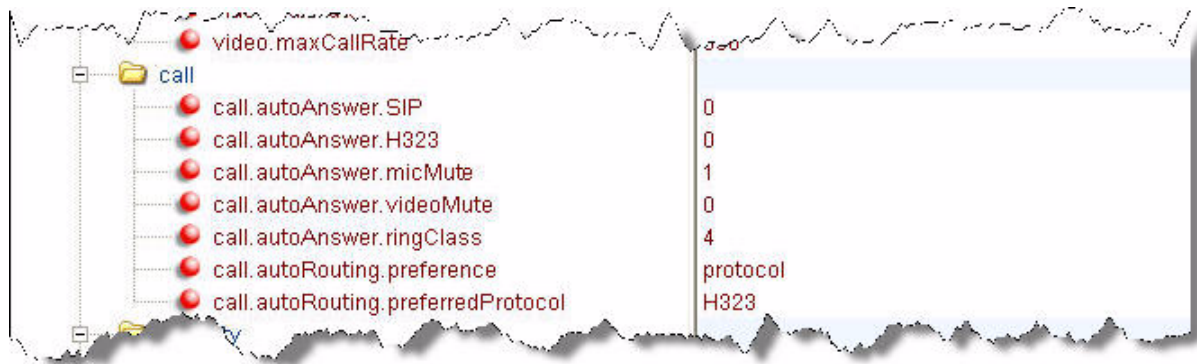
- Set the call rate to your organization's desired value.
- Set the maximum call rate to your organization's desired value.



Call Handling Configuration <call/>

Edit the call parameter as shown below:

- Set the H.323 auto-answer flag to the desired value.
- Set the microphone mute and video mute auto-answer flags to the desired values.
- Set the auto-routing preference to **protocol**.
- Set the auto-routing preferred protocol to **H323**.



Event Logging <log/>

By default, log.level.change.h323 is set to 4 (Minor error - graceful recovery).

The permitted values are 0 to 5.

Security <sec/>

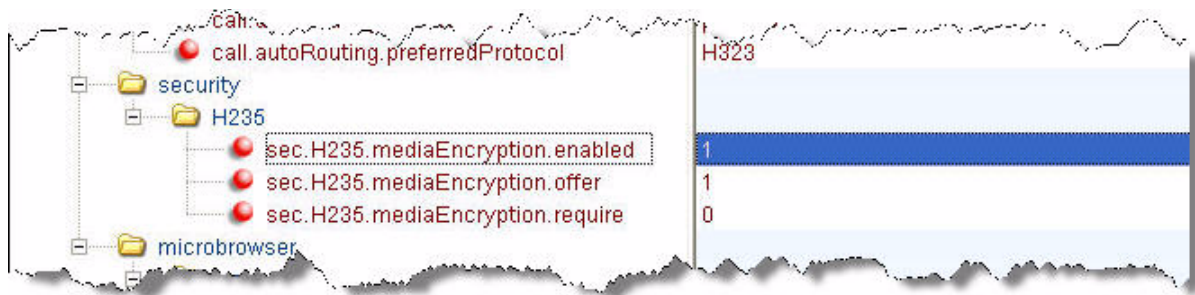
Note

The H.235 Voice Profile implementation is Polycom HDX-compatible. OpenSSL-based Diffie-Hellman key exchange and AES-128 CBC encryption algorithms are used to encrypt the RTP media.

Edit the H.235 security parameter as shown below:

- Enable media encryption.
- Enabled media encryption offered.

- Disable media encryption required.

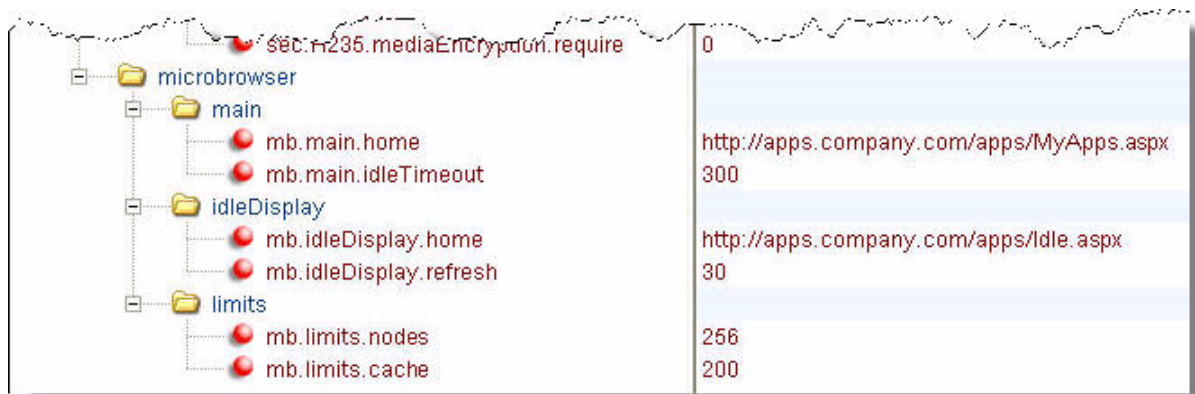


Microbrowser <mb/>

This attribute's settings control the home page, proxy and size limits to be used by the Polycom Browser when it is selected to provide services.

Edit the microbrowser parameters as shown below:

- Set the idle display home page URL and refresh rate.
- Set the main browser home page URL and idle timeout.



Per-Phone Configuration File

This section covers the parameters in the per-phone example configuration file **phone1.cfg**. This file would normally be used as a template for the per-phone configuration files.

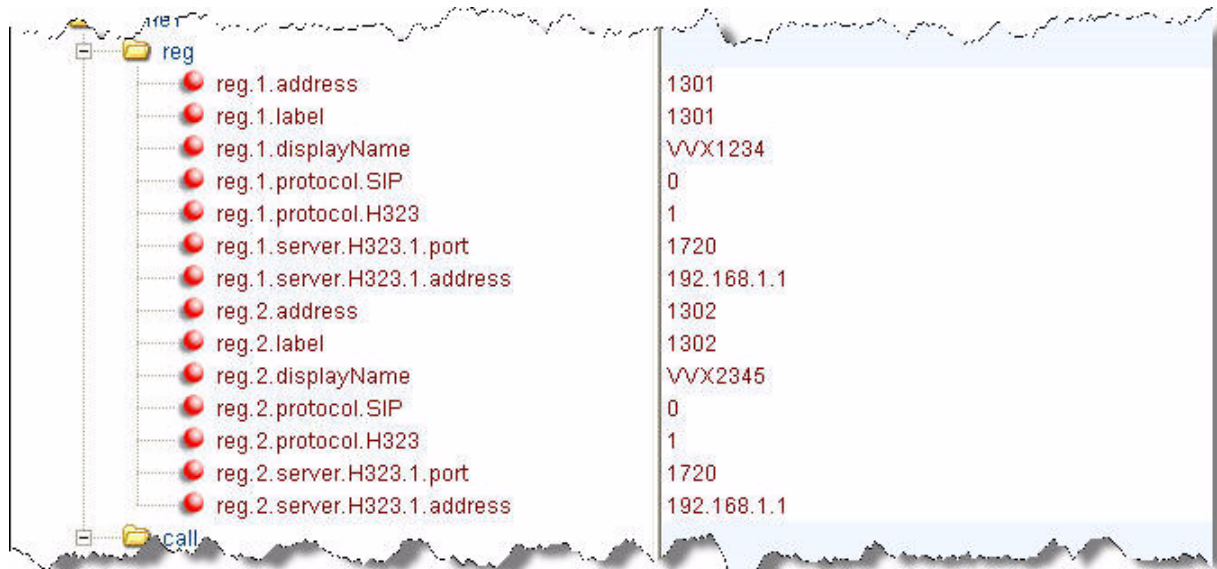
The important parameters for the Polycom VVX 1500 D phone include:

- [Registration <reg/>](#) – Specify the registration protocol.

Registration <reg/>

The Polycom VVX 1500 D phone supports six registrations (x= 1 to 6).

Edit the reg parameter as shown below (to reflect the registered lines that you want to appear on the phone).



reg.1.address	1301
reg.1.label	1301
reg.1.displayName	VVX1234
reg.1.protocol.SIP	0
reg.1.protocol.H323	1
reg.1.server.H323.1.port	1720
reg.1.server.H323.1.address	192.168.1.1
reg.2.address	1302
reg.2.label	1302
reg.2.displayName	VVX2345
reg.2.protocol.SIP	0
reg.2.protocol.H323	1
reg.2.server.H323.1.port	1720
reg.2.server.H323.1.address	192.168.1.1

Miscellaneous Administrative Tasks

This appendix provides information that may be required to administer the Polycom VVX 1500 D phone. This includes:

- [Setting Up an FTP Server](#)

For other miscellaneous tasks, refer to the “Miscellaneous Administrative Tasks” appendix of the latest *SIP Administrator’s Guide*.

Setting Up an FTP Server

A simple provisioning configuration uses File Transfer Protocol or FTP. Although FTP servers are free, they require installation, and use logins and passwords.

A free and popular server, Filezilla Server is available for Windows at <http://filezilla-project.org>. This application has been tested with the SoundStation IP 7000 phone.

To set up the server:

1. Download and install the latest version of Filezilla Server.
2. After installation, you are presented with a “Connect to Server” pop-up. Select **OK** to open the administrative user interface.
3. To configure a user, select **Edit > Users** in the status bar.
4. Select **Add**.
5. Enter the user name for the phone and select **OK**.
For example, “bob” or “bill123”
6. Select the **Password** checkbox and enter a password.
For example, “1234” or “phone&&&”
The phone will use this password to login.

7. Select **Page >Shared folders** to specify the server-side directory where the provisioning files will be located (and the log files uploaded).
 8. Select **Add** and pick the directory.
 9. To allow the phone to upload logs onto the provisioning server, select the **Shared Folders->Files->Select Write and Delete** checkboxes, and then select **OK**.
 10. Copy the configuration files from the software download into the directory specified above.
 11. Determine the IP address of the provisioning server by entering **ipconfig** at a command prompt to display the server network configuration.
- The IP Address of the provisioning server is shown.
12. Do the following to configure the phone to point to the IP address of the server:

- a Press the phone's **Menu** key.
 - b Through the menu, select **Settings >Advanced >Admin Settings >Network Configuration**.
- When prompted for the administrative password, enter **456**.
- c Scroll down to **Server Menu**, then press the **Edit** soft key.
 - d Using the right arrow key, select **FTP** as the **Server Type**.
 - e Enter the IP address from step 11 as the **Server Address**, and then press the **OK** soft key.
 - f Enter the username and password from steps 5 and 6 as the **Server User** and **Server Password**.
 - g Scroll down **Save & Reboot**, then press the **Select** soft key.

The phone will reboot.

The phone will upload two logs files to the directory specified in step 9: **<MACaddress>-app.log** and **<MACaddress>boot.log** .

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