



**Asterisk Business Edition™
Version B.2.5.14
Digium Partner Certification**



**Interoperability Report
Polycom VVX® 1500 D (Development)
Firmware Version 3.2.2.0481
Preliminary Bootrom 4.2.1.0334**



Digium, Inc.
445 Jan Davis Drive NW
Huntsville, AL 35806
United States
Main Number: 1.256.428.6000
Tech Support: 1.256.428.6161
U.S. Toll Free: 1.877.344.4861
Sales: 1.256.428.6262
www.asterisk.org
www.digium.com
www.asterisknow.org

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Section 1: Executive Summary

This document covers the tests executed for validation of interoperability of the partner's product(s) with Digium's Asterisk Business Edition. All relevant information is included in order to allow the replication of these test scenarios.

1.1 Products Tested

Asterisk Business Edition has been thoroughly tested for interoperability against the partner's product(s) listed below. The software versions for all tested products are included.

1.1.1 Asterisk Business Edition

| Product | Version | Remarks |
|---------------------------|----------|---------|
| Asterisk Business Edition | B.2.5.14 | |

1.1.2 Partner Equipment Tested (UUTs)

| Partner | Product | Version | Remarks |
|---------|------------|------------|---------|
| Polycom | VVX 1500 D | 3.2.2.0481 | |

A dual stack business media phone for H.323 and SIP environments, the Polycom VVX 1500 D combines one-touch video calling, integrated business applications, and advanced IP telephony.

- **Key Features and Benefits:**

- On-device, simultaneous support for H.323 and SIP for easy connectivity and flexibility
- Intuitive, color touch-screen phone interface for voice, video and applications
- Instant, one-touch business-grade video conferencing right from the desktop
- Adjustable camera, base, and display to suit the environment and provide eye-level visual interactions
- Six-line, feature-rich touch screen phone with Polycom HD Voice™ technology 1
- Highly customizable applications platform with open Polycom API, integrated full browser and USB 2.0 port for applications
- Bundled with productivity and personalization applications, including Polycom Productivity Suite, Polycom My Info Portal and Digital Photo Frame
- Integrated Gigabit Ethernet (GigE) switch for bandwidth-intensive applications
- IEEE 802.3af powered at under 11W maximum for a "green" world
- Smart motion detection allows the screen to go to power-save mode when no one is in the office

1.2 Summary of Test Results

A summary of the test results is provided below. Detailed test results are available in Section 4.

1.2.1 Feature Matrix

| Feature | Polycom VVX 1500 D |
|---------------------|--------------------|
| SIP Register | ✓ |
| Outbound Call | ✓ |
| Inbound Call | ✓ |
| Call History | ✓ |
| Hold and Resume | ✓ |
| Attended Transfer | ✓ |
| Unattended Transfer | ✓ |
| Conferencing | ✓ |
| Forwarding | ✓ |
| MWI | ✓ |
| DND | ✓ |
| Codec G.729 | ✓ |
| Codec G.722 | □ |
| DTMF Mode Inband | ✓ |

| Legend | |
|--------|----------------|
| ✓ | Pass |
| ✗ | Fail |
| □ | Not Applicable |

Section 2: Test Configuration

This section describes the test configuration and setup, and any additional equipment that was required to perform the testing. A diagram of the test setup is available in Section 2.2.

2.1 Description of Test Setup

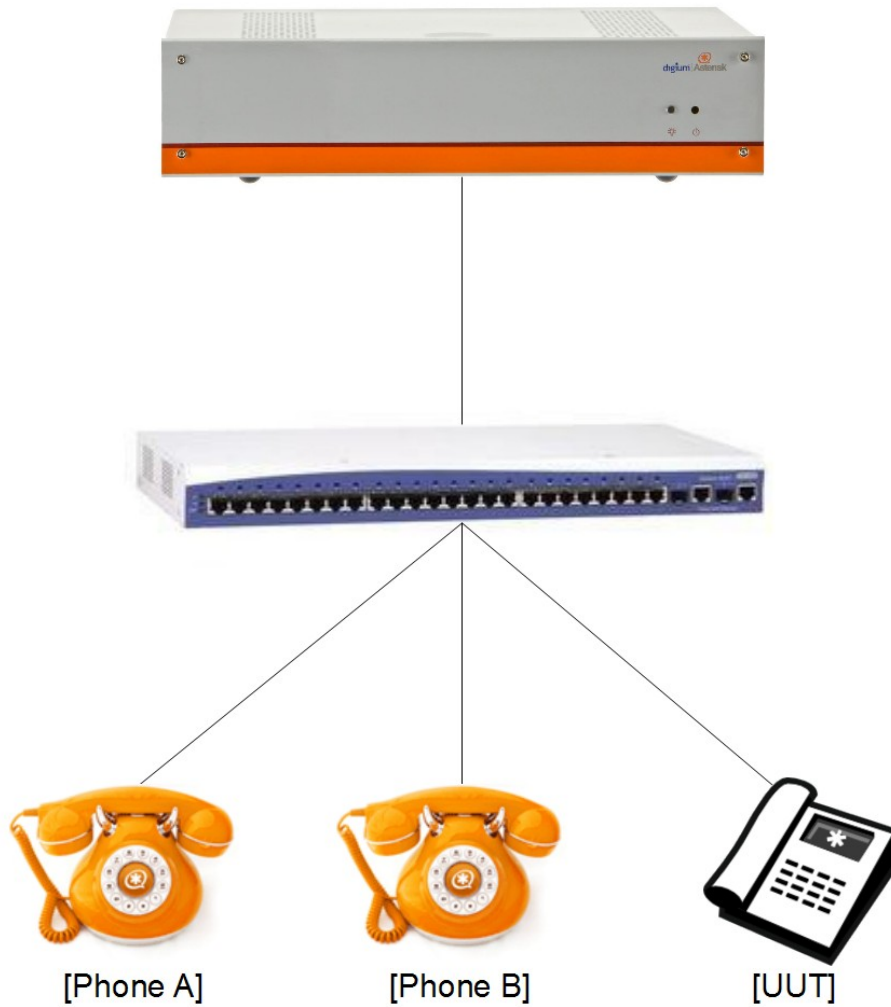
An isolated test network was created using an Adtran NetVanta switch and a PC-based server running Asterisk Business Edition. The partner phone (UUT) was connected to the test network via the Adtran switch. Each feature listed in this document was tested by placing calls to and from the UUT and the Asterisk Business Edition server. Native Bridging was disabled to ensure all traffic was directed through the Asterisk Business Edition Server.

2.1.1 Other Equipment Used During Testing

| Vendor | Product | Version | Remarks |
|--------|----------|---------|---------|
| Adtran | NetVanta | 1224st | |

2.2 Test Setup Diagram

The diagram listed below illustrates how the test equipment was connected during testing. This diagram applies to all tests within this report.



Section 3: Product Configuration

The relevant portions of the configuration for the tested products are included in this section.

/etc/asterisk/sip.conf

```
[general]

;*****
;*UUT*
;*****
[6370]
type=friend
username=6370
secret=6370
host=dynamic
context=testing
disallow=all
allow=ulaw
qualify=1000
subscribecontext=BLF_Enable
mailbox=6370

;*****
;*Phone A*
;*****
[7000]
type=friend
username=7000
secret=7000
host=dynamic
context=testing
disallow=all
allow=ulaw
qualify=yes
subscribecontext=BLF_Enable
mailbox=7000

;*****
;*Phone B*
;*****
[6000]
type=friend
username=6000
secret=6000
host=dynamic
context=testing
disallow=all
allow=ulaw
qualify=yes
subscribecontext=BLF_Enable
mailbox=6000
```

/etc/asterisk/extensions.conf

```
[testing]
exten => _6XXX,1,Dial(sip/${EXTEN},4,j)
exten => _6XXX,n,VoiceMail(${EXTEN},20,j)

exten => _7XXX,1,Dial(sip/${EXTEN},4,j)
exten => _7XXX,n,VoiceMail(${EXTEN},20,j)

exten => asterisk,1,VoiceMailmain(${CALLERID(num)},s)

exten => 8500,1,VoiceMailMain()

exten => 5001,1,Meetme(${EXTEN},i)
exten => 5001,n,Hangup()

[BLF_Enable]
exten => 6370, hint, SIP/6370
exten => 7000, hint, SIP/7000
exten => 6000, hint, SIP/6000
```

/etc/asterisk/voicemail.conf

```
[default]
6370 => 6370, Polycom VVX 1500 D, root@localhost
7000 => 7000, Polycom 7000, root@localhost
6000 => 6000, Polycom 6000, root@localhost
```

Section 4: Tests Performed

The specific tests performed for verification of functionality with the partner's product(s) are provided below.

4.1 SIP Registration

| Test Case PC-8: SIP Registration | |
|----------------------------------|--|
| Summary | This test verifies the functionality of authenticating and registering to the Asterisk server. |
| Step(s) | Configure the phone to register to the Asterisk server. |
| Expected Result(s) | The phone authenticates and registers successfully. |
| Pass / Fail | Passed |
| Test Notes | Test performed on Build Polycom-VVX-1500-D-3.2.2.0481-BE.B.2.5.14. |
| Author | spimental |

4.2 Outbound Call

| Test Case PC-7: Outbound Call | |
|-------------------------------|--|
| Summary | This test verifies the functionality of placing outgoing calls. |
| Step(s) | <ol style="list-style-type: none">1. Dial from the UUT to Phone A.2. Verify the UUT receives ringback.3. Verify the Phone A receives the Caller ID from the UUT. |
| Expected Result(s) | <ul style="list-style-type: none">• The UUT will receive ringback and the call will connect.• The two callers will receive full duplex audio.• Caller ID will be received successfully.• The line on the UUT will display as busy/off-hook. |
| Pass / Fail | Passed |
| Test Notes | Test performed on Build Polycom-VVX-1500-D-3.2.2.0481-BE.B.2.5.14. |
| Author | spimental |

4.3 Inbound Call

| Test Case PC-6: Inbound Call | |
|------------------------------|--|
| Summary | This test verifies the functionality of receiving incoming calls. |
| Step(s) | <ol style="list-style-type: none">1. Dial from Phone A to the extension set for the UUT.2. Verify ringback.3. Verify Caller ID is displayed and the line displays as busy/off-hook. |
| Expected Result(s) | <ul style="list-style-type: none">• The call will be received successfully.• The two callers will receive full duplex audio.• Caller ID will be received successfully.• Ringback will be provided to the calling party. |
| Pass / Fail | Passed |
| Test Notes | Test performed on Build Polycom-VVX-1500-D-3.2.2.0481-BE.B.2.5.14. |
| Author | spimental |

4.4 Call History

| Test Case PC-3: Call History | |
|------------------------------|---|
| Summary | This test verifies the functionality of the Call History feature. |
| Step(s) | <ol style="list-style-type: none">1. Using the phone LCD menu navigation, clear the Call History records in the UUT. Note that most phones have history for: Placed Calls, Received Calls, and Missed Calls. Some phones with limited feature sets may only have history for: Placed Calls and Received Calls.2. Place a call from UUT to Phone A, then answer the call and hangup.3. Place a call to UUT from Phone A, then answer the call and hangup.4. Place a call to the UUT, then let it go to VoiceMail.5. Check the Call History in the UUT. |
| Expected Result(s) | <ul style="list-style-type: none">• All Call History records will be cleared from the phone.• The Call History in the UUT will show:<ul style="list-style-type: none">◦ One call placed by the UUT to Phone A◦ One call received by the UUT from Phone A◦ One missed call from Phone A |
| Pass / Fail | Passed |
| Test Notes | Test performed on Build Polycom-VVX-1500-D-3.2.2.0481-BE.B.2.5.14. |
| Author | spitts |

4.5 Hold and Resume

| Test Case PC-4: Hold and Resume | |
|---------------------------------|--|
| Summary | This test verifies the functionality of the Hold and Resume feature. |
| Step(s) | <ol style="list-style-type: none">1. Place a call to the UUT.2. Place the calling party on hold.3. Place a call from the UUT to another party.4. The UUT will end the new call and resume the call with the original party. |
| Expected Result(s) | <ul style="list-style-type: none">• A two-way voice path will be established.• The calling party will hear MoH.• A new two-way voice path will be established.• The new call is dropped, and the original call is resumed. |
| Pass / Fail | Passed |
| Test Notes | Test performed on Build Polycom-VVX-1500-D-3.2.2.0481-BE.B.2.5.14. |
| Author | spitts |

4.6 Attended Transfer

| Test Case PC-5: Attended Transfer | |
|-----------------------------------|--|
| Summary | This test verifies the functionality of the Attended Transfer feature. |
| Step(s) | <ol style="list-style-type: none">1. Place a call to the UUT from phone A.2. On the UUT, press the Transfer button, then dial the number for Phone B.3. Answer Phone B when it rings.4. Once the call to Phone B is established, press the Transfer button again. |
| Expected Result(s) | <ul style="list-style-type: none">• A two-way voice channel is established between the UUT and Phone A.• A two-way voice channel will be established between the UUT and Phone B.• Phone B is connected to Phone A. |
| Pass / Fail | Passed |
| Test Notes | Test performed on Build Polycom-VVX-1500-D-3.2.2.0481-BE.B.2.5.14. |
| Author | spitts |

4.7 Unattended Transfer

| Test Case PC-2: Unattended Transfer | |
|-------------------------------------|---|
| Summary | This test verifies the functionality of the Unattended Transfer feature. |
| Step(s) | <ol style="list-style-type: none">1. Place a call to the UUT from Phone A.2. On the UUT, press the Transfer button, then dial the number for Phone B.3. Press the transfer button before Phone B answers.4. Answer Phone B.5. Verify that the call to Phone B is established. |
| Expected Result(s) | <ul style="list-style-type: none">• A two-way voice channel is established between the UUT and Phone A.• Phone B is connected to Phone A.• All lines on UUT will show as on-hook when the UUT transfers the call. |
| Pass / Fail | Passed |
| Test Notes | Test performed on Build Polycom-VVX-1500-D-3.2.2.0481-BE.B.2.5.14. |
| Author | spitts |

4.8 Conferencing

| Test Case PC-1: Conferencing | |
|------------------------------|--|
| Summary | This test verifies the functionality of phone-managed conferencing. |
| Step(s) | <ol style="list-style-type: none">1. Place a call from the UUT to Phone A.2. On the UUT, press the Conference button, then dial the number for Phone B.3. Once the call is established to Phone B, press the Conference button again. |
| Expected Result(s) | <ul style="list-style-type: none">• A two-way voice path will be established from the UUT to Phone A.• A two-way voice path will be established from the UUT to Phone B.• A conference will be established that bridges the UUT, Phone A, and Phone B. |
| Pass / Fail | Passed |
| Test Notes | Test performed on Build Polycom-VVX-1500-D-3.2.2.0481-BE.B.2.5.14. |
| Author | spitts |

4.9 Forwarding

| Test Case PC-9: Forwarding | |
|----------------------------|--|
| Summary | This test verifies the functionality of the Call Forwarding feature. |
| Step(s) | <ol style="list-style-type: none">1. Place a call from Phone A to the UUT, verify the voice path, and then end the call.2. On the UUT, select Forwarding, then enable and enter the extension for Phone B.3. Place a call from Phone A to the UUT.4. On the UUT, select Forwarding, then select disable.5. Place a call from Phone A to the UUT. |
| Expected Result(s) | <ul style="list-style-type: none">• UUT rings, then a two-way voice path will be established when the UUT is answered.• Phone B rings, then a two-way voice path will be established when Phone B is answered.• UUT rings, then a two-way voice path will be established when the UUT is answered. |
| Pass / Fail | Passed |
| Test Notes | Test performed on Build Polycom-VVX-1500-D-3.2.2.0481-BE.B.2.5.14. |
| Author | spimental |

4.10 Message Waiting Indicator

| Test Case PC-10: Message Waiting Indication | |
|---|---|
| Summary | This test verifies the functionality of the Message Waiting Indicator feature. |
| Step(s) | <ol style="list-style-type: none">1. Place a call from Phone A to the UUT.2. Do not answer the call. Let it go to VoiceMail.3. Leave a message for the UUT and end the call.4. Press the Messages button on the UUT.5. Enter the VoiceMailBox number and Secret for the UUT.6. Delete the voicemail once it has been reviewed.7. Verify that the MWI LED turns off. |
| Expected Result(s) | <ul style="list-style-type: none">• Phone A will enter into the VoiceMail menu.• The MWI LED on the UUT will start flashing and a message waiting symbol will be displayed on the UUT LCD.• The UUT will dial into VoiceMail.• The UUT will have 1 message from Phone A.• Once the message is deleted, the MWI indicator will turn off. |
| Pass / Fail | Passed |
| Test Notes | Test performed on Build Polycom-VVX-1500-D-3.2.2.0481-BE.B.2.5.14. |
| Author | spitts |

4.11 Do Not Disturb

| Test Case PC-11: Do Not Disturb | |
|---------------------------------|--|
| Summary | This test verifies the functionality of the Do Not Disturb feature. |
| Step(s) | <ol style="list-style-type: none">1. Place a call from Phone A to the UUT.2. End the call.3. Select Do Not Disturb on the UUT.4. Place a call from Phone A to the UUT.5. Disable Do Not Disturb on the UUT.6. Place a call from Phone A to the UUT. |
| Expected Result(s) | <ul style="list-style-type: none">• UUT rings, then a two-way voice path will be established when the UUT is answered.• UUT will not ring and the call will go to VoiceMail.• A two-way voice path will be established from the UUT to Phone A. |
| Pass / Fail | Passed |
| Test Notes | Test performed on Build Polycom-VVX-1500-D-3.2.2.0481-BE.B.2.5.14. |
| Author | spitts |

4.12 Codec G.729

| Test Case PC-14: Codec G.729 | |
|------------------------------|---|
| Summary | This test verifies the functionality of the G.729 codec. |
| Step(s) | <ol style="list-style-type: none">1. Set codec to G.729 in sip.conf.2. Dial from the UUT to Phone A.3. Verify that the UUT receives ringback.4. Verify that Phone A receives the Caller ID from the UUT. |
| Expected Result(s) | <ul style="list-style-type: none">• The UUT will receive ringback and the call will connect.• The two callers will receive full duplex audio.• Caller ID will be received successfully. |
| Pass / Fail | Passed |
| Test Notes | Test performed on Build Polycom-VVX-1500-D-3.2.2.0481-BE.B.2.5.14. |
| Author | spimental |

4.13 Codec G.722

| Test Case PC-15: Codec G.722 | |
|------------------------------|---|
| Summary | This test verifies the functionality of the G.722 codec. |
| Step(s) | <ol style="list-style-type: none">1. Set codec to G.722 in sip.conf.2. Dial from the UUT to Phone A.3. Verify that the UUT receives ringback.4. Verify that Phone A receives the Caller ID from the UUT. |
| Expected Result(s) | <ul style="list-style-type: none">• The UUT will receive ringback and the call will connect.• The two callers will receive full duplex audio.• Caller ID will be received successfully. |
| Pass / Fail | Not Applicable |
| Test Notes | Test performed on Build Polycom-VVX-1500-D-3.2.2.0481-BE.B.2.5.14. |
| Author | spimental |

4.14 DTMF Mode Inband

| Test Case PC-16: DTMF Mode Inband | |
|-----------------------------------|--|
| Summary | This test verifies the functionality of the inband DTMF mode. |
| Step(s) | <ol style="list-style-type: none">1. Set dtmfmode=inband in sip.conf.2. Dial from the UUT to Phone A.3. Verify that the UUT receives ringback.4. Verify that Phone A receives the Caller ID from the UUT. |
| Expected Result(s) | <ul style="list-style-type: none">• The UUT will receive ringback and the call will connect.• The two callers will receive full duplex audio.• Caller ID will be received successfully. |
| Pass / Fail | Passed |
| Test Notes | Test performed on Build Polycom-VVX-1500-D-3.2.2.0481-BE.B.2.5.14. |
| Author | spimental |

Section 5: Glossary of Common Terms

The following is a glossary of common telecommunication acronyms and terms that may be used in this report.

| Term | Definition |
|-----------|---|
| Codec | Coder/Decoder, Compressor/Decompressor. Software or hardware (or a combination of both) that converts data to a code and later decodes it, e.g. telephone firmware that converts digital signals to analog, and vice versa. Also, technology (such as MPEG) that compresses data (such as sound files) for storage and decompresses it for processing. |
| DND | Do Not Disturb |
| Fast Busy | A busy signal (also referred to as a “reorder”) in telephony is an audible or visual signal to the calling party that indicates failure to complete the requested connection of that particular telephone call. |
| Gateway | A general term used by various companies to refer to the controlling interface between the PBX and the phones within a local area network. Other companies’ “gateways” are called Call Managers or Call Servers. |
| PBX | Private Branch Exchange. Originally referring to a system providing local telephone service (“public exchange”) and access to the PSTN, PBX now typically refers to whatever connection a phone user has to other users or to the outside world. In some cases, that connection is a call manager, call server, or gateway, or some other box or combination of boxes. In some IP protocols there might not even be such a box, but simply a direct access to the Internet. |
| POE | Power over Ethernet (POE) technology is a system to transmit electrical power, along with data over a standard Ethernet cable to remote devices such as IP Telephones, remote network switched, and other appliances where it would be inconvenient or more expensive to provide a separate power supply for the device. |
| SIP | Session Initiation Protocol (SIP) is the Internet Engineering Task Force's (IETF's) standard for multimedia conferencing over IP. SIP is an ASCII-based, application-layer control protocol (defined in RFC 2543) that can be used to establish, maintain, and terminate calls between two or more end points. |
| TDM | Time-Division Multiplexing. A type of digital signaling and transmission (sometimes used in digital-to-analog or analog-to digital systems) in which two or more signals or bit streams are transferred simultaneously as sub-channels in one communication channel, physically “taking turns” on the channel. Examples of TDM communications include T1, E1, and J1 digital lines. |

| Term | Definition |
|------|---|
| TFTP | Trivial (or Thin) File Transport Protocol. A simple form of FTP, TFTP uses UDP and provides no security features. It is often used by servers to download firmware or configurations to IP phones, embedded network devices, routers, and other devices whose user interfaces are simple or not included. |
| UUT | Unit Under Test. In a formal test setup, the UUT is the device that is being tested or evaluated. |
| VoIP | Voice-over Internet Protocol |