



**Asterisk Business Edition™  
Version C.1.4  
Digium Partner Certification**



**Interoperability Report  
Polycom® SoundPoint IP 450  
Firmware Version 3.1.0.0147**



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](http://www.asterisk.org)  
[www.digium.com](http://www.digium.com)  
[www.asterisknow.org](http://www.asterisknow.org)

© Digium®, Inc. 2009  
All rights reserved.

No part of this publication may be copied, distributed, transmitted, transcribed, stored in a retrieval system, or translated into any human or computer language without the prior written permission of Digium, Inc.

This document describes test setups, configurations, test plans, and test results that Digium has performed or validated to determine the level of interoperability between the named Digium products and those of a partner or other vendor, in cooperation with the partner or vendor. This document does not necessarily describe all features or usage scenarios of the products; only those which Digium believes are essential for basic interoperability, and those additional features that Digium and the partner or vendor have agreed to describe and test are included. These tests typically are of a functional nature to assure static interoperability, and do not include or purport to be dynamic, stress, or performance tests under loads or changing conditions unless otherwise indicated. Thus, these tests may not be representative of “real-world” conditions you may encounter. Digium, Inc. has made reasonable efforts to ensure that the information contained in this document is accurate at the time of its release, for the versions of each product described and tested or validated as described herein. However, since products are often revised over time, Digium cannot guarantee accuracy of the information contained herein after the date of release of this document. Digium welcomes input on how to improve its documentation, but Digium’s liability for any errors in this document is limited to the correction of such errors at its sole discretion. This document has been prepared for use by professional and properly trained personnel, and the user assumes full responsibility when using it.

In no event will Digium or its suppliers, distributors, employers, agents, or officers be liable for any loss of data, loss of income, loss of opportunity or profits, or cost of recovery or for any other special, incidental, consequential, or indirect damages arising from the use of this document or any information herein, however caused and under any theory of liability. This limitation will apply even if Digium has been advised of the possibility of such damage. In no event shall Digium's liability for any errors or omissions in this document exceed the amount paid for the Digium Products or Services at issue, or \$1000.00 (One thousand U.S. Dollars), whichever is less.

Asterisk, Digium, Switchvox, and AsteriskNOW are registered trademarks of Digium, Inc. Asterisk Business Edition, AsteriskGUI, and Asterisk Appliance are trademarks of Digium, Inc. Any other trademarks mentioned in the document are the property of their respective owners.

# TABLE OF CONTENTS

---

|  |           |
|--|-----------|
| <b>Section 1: Executive Summary</b> .....      | <b>5</b>  |
| 1.1 Products Tested.....                       | 5         |
| 1.1.1 Asterisk Business Edition.....           | 5         |
| 1.1.2 Partner Equipment Tested (UUTs).....     | 6         |
| 1.2 Summary of Test Results.....               | 7         |
| 1.2.1 Feature Matrix.....                      | 7         |
| <b>Section 2: Test Configuration</b> .....     | <b>8</b>  |
| 2.1 Description of Test Setup.....             | 8         |
| 2.1.1 Other Equipment Used During Testing..... | 8         |
| 2.2 Test Setup Diagram.....                    | 9         |
| <b>Section 3: Product Configuration</b> .....  | <b>10</b> |
| 3.1 Configuration of Digium Products.....      | 10        |
| <b>Section 4: Tests Performed</b> .....        | <b>12</b> |
| 4.1 Registration.....                          | 12        |
| 4.1.1 SIP Registration.....                    | 12        |
| 4.2 Basic Call Functions.....                  | 13        |
| 4.2.1 Outbound Call.....                       | 13        |
| 4.2.2 Inbound Call.....                        | 14        |
| 4.2.3 Call History.....                        | 15        |
| 4.2.4 Hold and Resume.....                     | 16        |
| 4.2.5 Attended Transfer.....                   | 17        |
| 4.2.6 Unattended Transfer.....                 | 18        |
| 4.2.7 Conferencing.....                        | 19        |
| 4.2.8 Forwarding.....                          | 20        |
| 4.3 Advanced Call Features.....                | 21        |

|   |           |
|---|-----------|
| 4.3.1 Message Waiting Indicator.....            | 21        |
| 4.3.2 Do Not Disturb.....                       | 22        |
| <b>Section 5: Glossary of Common Terms.....</b> | <b>23</b> |

# 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 | C.1.4   |         |

## 1.1.2 Partner Equipment Tested (UUTs)

| Partner | Product           | Version    | Remarks |
|---------|-------------------|------------|---------|
| Polycom | SoundPoint IP 450 | 3.1.0.0147 |         |

The Polycom SoundPoint IP 450 is a SIP desktop phone with a high-resolution, backlit display.

- **Key Features and Benefits**

- Backlit 256 x 116-pixel multi-layer, grayscale graphical LCD with support of multiple languages and Asian characters
- Integrated IEEE 802.3af Power over Ethernet (PoE) support (Class 2)
- Two port 10/100 Ethernet Switch
- Dedicated RJ-9 headset port
- Built-in XHTML Microbrowser for third party applications
- Three lines for a moderate volume of calls
- Polycom HD Voice™ technology for improved voice quality
- Advanced SIP functionality, including shared lines, busy lamp field, and presence

## 1.2 Summary of Test Results

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

| Test Category          | Results | Importance | Remarks |
|------------------------|---------|------------|---------|
| Registration           | Pass    | Mandatory  |         |
| Basic Call Functions   | Pass    | Mandatory  |         |
| Advanced Call Features | Pass    | Suggested  |         |

### 1.2.1 Feature Matrix

| Feature             | SoundPoint IP 450 |
|---------------------|-------------------|
| SIP Register        | ✓                 |
| Outbound Call       | ✓                 |
| Inbound Call        | ✓                 |
| Caller ID           | ✓                 |
| Call History        | ✓                 |
| Hold and Resume     | ✓                 |
| Attended Transfer   | ✓                 |
| Unattended Transfer | ✓                 |
| Conferencing        | ✓                 |
| Forwarding          | ✓                 |
| MWI                 | ✓                 |
| DND                 | ✓                 |

## 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.

### 3.1 Configuration of Digium Products

The following are relevant portions of the Asterisk configuration files used during testing.

```
sip.conf

[general]

;*****
;*UUT*
;*****
[6450]
type=friend
username=6450
secret=6450
host=dynamic
context=testing
disallow=all
allow=ulaw
qualify=1000
subscribecontext=BLF_Enable
mailbox=6450

;*****
;*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
```

## 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 => 6450, hint, SIP/6450
exten => 7000, hint, SIP/7000
exten => 6000, hint, SIP/6000
```

## voicemail.conf

```
[default]
6450 => 6450, Polycom 450, 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. Both mandatory and suggested tests are included. Mandatory tests verify functionality which is required for interoperability. Suggested tests verify functionality which is desired, but which is not required for interoperability.

### 4.1 Registration

These mandatory tests check the registration of the phone with the Asterisk server.

#### 4.1.1 SIP Registration

| Test Case Interop-35: SIP Registration |   |
|--|---|
| Summary                                | This test is to ensure the UUT can authenticate and register to the Asterisk server successfully. |
| 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 450 (FW 3.1.0.0147) on BE-C.1.4.                                  |
| Author                                 | spimental   |

## 4.2 Basic Call Functions

These mandatory tests check the basic call functionality of the phone.

### 4.2.1 Outbound Call

| Test Case Interop-34: Outbound Call |  |
|-------------------------------------|--|
| Summary                             | This test is to ensure the UUT can place outgoing calls.   |
| Step(s)                             | <ol style="list-style-type: none"><li>1. Dial from the UUT to Phone A.</li><li>2. Verify the UUT receives ringback.</li><li>3. Verify the Phone A receives the Caller ID from the UUT.</li></ol>   |
| Expected Result(s)                  | <ul style="list-style-type: none"><li>• The UUT will receive ringback and the call will connect.</li><li>• The two callers will receive full duplex audio.</li><li>• Caller ID will be received successfully.</li><li>• The line on the UUT will display as busy/off-hook.</li></ul> |
| Pass / Fail                         | Passed   |
| Test Notes                          | Test performed on Build Polycom 450 (FW 3.1.0.0147) on BE-C.1.4.   |
| Author                              | spimental  |

## 4.2.2 Inbound Call

| Test Case Interop-33: Inbound Call |  |
|------------------------------------|--|
| Summary                            | This test is to ensure the UUT can receive incoming calls.   |
| Step(s)                            | <ol style="list-style-type: none"><li>1. Dial from Phone A to the extension set for the UUT.</li><li>2. Verify ringback.</li><li>3. Verify Caller ID is displayed and the line displays as busy/off-hook.</li></ol>  |
| Expected Result(s)                 | <ul style="list-style-type: none"><li>• The call will be received successfully.</li><li>• The two callers will receive full duplex audio.</li><li>• Caller ID will be received successfully.</li><li>• Ringback will be provided to the calling party.</li></ul> |
| Pass / Fail                        | Passed   |
| Test Notes                         | Test performed on Build Polycom 450 (FW 3.1.0.0147) on BE-C.1.4.   |
| Author                             | spimental  |

### 4.2.3 Call History

| Test Case Interop-30: Call History |   |
|------------------------------------|---|
| Summary                            | This test verifies the operation of the Call History feature.   |
| Step(s)                            | <ol style="list-style-type: none"><li>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.</li><li>2. Place a call from UUT to Phone A, then answer the call and hangup.</li><li>3. Place a call to UUT from Phone A, then answer the call and hangup.</li><li>4. Place a call to the UUT, then let it go to VoiceMail.</li><li>5. Check the Call History in the UUT.</li></ol> |
| Expected Result(s)                 | <ul style="list-style-type: none"><li>• All Call History records will be cleared from the phone.</li><li>• The Call History in the UUT will show:<ul style="list-style-type: none"><li>◦ One call placed by the UUT to Phone A</li><li>◦ One call received by the UUT from Phone A</li><li>◦ One missed call from Phone A</li></ul></li></ul>   |
| Pass / Fail                        | Passed  |
| Test Notes                         | Test performed on Build Polycom 450 (FW 3.1.0.0147) on BE-C.1.4.  |
| Author                             | spitts  |

## 4.2.4 Hold and Resume

| Test Case Interop-31: Hold and Resume |  |
|---------------------------------------|--|
| Summary                               | This test verifies the operation of Hold and Resume.   |
| Step(s)                               | <ol style="list-style-type: none"><li>1. Place a call to the UUT.</li><li>2. Place the calling party on hold.</li><li>3. Place a call from the UUT to another party.</li><li>4. The UUT will end the new call and resume the call with the original party.</li></ol> |
| Expected Result(s)                    | <ul style="list-style-type: none"><li>• A two-way voice path will be established.</li><li>• The calling party will hear MoH.</li><li>• A new two-way voice path will be established.</li><li>• The new call is dropped, and the original call is resumed.</li></ul>  |
| Pass / Fail                           | Passed   |
| Test Notes                            | Test performed on Build Polycom 450 (FW 3.1.0.0147) on BE-C.1.4.   |
| Author                                | spitts   |



## 4.2.5 Attended Transfer

| Test Case Interop-32: Attended Transfer |  |
|---|--|
| Summary                                 | This test verifies the functionality of the Attended Call Transfer Feature.  |
| Step(s)                                 | <ol style="list-style-type: none"><li>1. Place a call to the UUT from phone A.</li><li>2. On the UUT, press the Transfer button, then dial the number for Phone B.</li><li>3. Answer Phone B when it rings.</li><li>4. Once the call to Phone B is established, press the Transfer button again.</li></ol> |
| Expected Result(s)                      | <ul style="list-style-type: none"><li>• A two-way voice channel is established between the UUT and Phone A.</li><li>• A two-way voice channel will be established between the UUT and Phone B.</li><li>• Phone B is connected to Phone A.</li></ul>  |
| Pass / Fail                             | Passed   |
| Test Notes                              | Test performed on Build Polycom 450 (FW 3.1.0.0147) on BE-C.1.4.   |
| Author                                  | spitts   |

## 4.2.6 Unattended Transfer

| Test Case Interop-29: Unattended Transfer |   |
|---|---|
| Summary                                   | This test verifies the functionality of the Unattended Call Transfer Feature.   |
| Step(s)                                   | <ol style="list-style-type: none"><li>1. Place a call to the UUT from Phone A.</li><li>2. On the UUT, press the Transfer button, then dial the number for Phone B.</li><li>3. Press the transfer button before Phone B answers.</li><li>4. Answer Phone B.</li><li>5. Verify that the call to Phone B is established.</li></ol> |
| Expected Result(s)                        | <ul style="list-style-type: none"><li>• A two-way voice channel is established between the UUT and Phone A.</li><li>• Phone B is connected to Phone A.</li><li>• All lines on UUT will show as on-hook when the UUT transfers the call.</li></ul>   |
| Pass / Fail                               | Passed  |
| Test Notes                                | Test performed on Build Polycom 450 (FW 3.1.0.0147) on BE-C.1.4.  |
| Author                                    | spitts  |

## 4.2.7 Conferencing

| Test Case Interop-28: Conferencing |  |
|------------------------------------|--|
| Summary                            | This test verifies the operation of phone-managed conferencing.  |
| Step(s)                            | <ol style="list-style-type: none"><li>1. Place a call from the UUT to Phone A.</li><li>2. On the UUT, press the Conference button, then dial the number for Phone B.</li><li>3. Once the call is established to Phone B, press the Conference button again.</li></ol>                  |
| Expected Result(s)                 | <ul style="list-style-type: none"><li>• A two-way voice path will be established from the UUT to Phone A.</li><li>• A two-way voice path will be established from the UUT to Phone B.</li><li>• A conference will be established that bridges the UUT, Phone A, and Phone B.</li></ul> |
| Pass / Fail                        | Passed   |
| Test Notes                         | Test performed on Build Polycom 450 (FW 3.1.0.0147) on BE-C.1.4.   |
| Author                             | spitts   |

## 4.2.8 Forwarding

| Test Case Interop-36: Forwarding |  |
|----------------------------------|--|
| Summary                          | This test verifies the operation of Call Forwarding.   |
| Step(s)                          | <ol style="list-style-type: none"><li>1. Place a call from Phone A to the UUT, verify the voice path, and then end the call.</li><li>2. On the UUT, select Forwarding, then enable and enter the extension for Phone B.</li><li>3. Place a call from Phone A to the UUT.</li><li>4. On the UUT, select Forwarding, then select disable.</li><li>5. Place a call from Phone A to the UUT.</li></ol> |
| Expected Result(s)               | <ul style="list-style-type: none"><li>• UUT rings, then a two-way voice path will be established when the UUT is answered.</li><li>• Phone B rings, then a two-way voice path will be established when Phone B is answered.</li><li>• UUT rings, then a two-way voice path will be established when the UUT is answered.</li></ul>   |
| Pass / Fail                      | Passed   |
| Test Notes                       | Test performed on Build Polycom 450 (FW 3.1.0.0147) on BE-C.1.4.   |
| Author                           | spimental  |

## 4.3 Advanced Call Features

These tests check the advanced call features of the phone. Support for these features is suggested, but not mandatory. Future releases of the product(s) may support these features correctly.

### 4.3.1 Message Waiting Indicator

| Test Case Interop-8: Message Waiting Indication |   |
|---|---|
| Summary   | This test verifies the operation of Message Waiting Indicator.  |
| Step(s)   | <ol style="list-style-type: none"><li>1. Place a call from Phone A to the UUT.</li><li>2. Do not answer the call. Let it go to VoiceMail.</li><li>3. Leave a message for the UUT and end the call.</li><li>4. Press the Messages button on the UUT.</li><li>5. Enter the VoiceMailBox number and Secret for the UUT.</li><li>6. Delete the voicemail once it has been reviewed.</li><li>7. Verify that the MWI LED turns off.</li></ol> |
| Expected Result(s)                              | <ul style="list-style-type: none"><li>• Phone A will enter into the VoiceMail menu.</li><li>• The MWI LED on the UUT will start flashing and a message waiting symbol will be displayed on the UUT LCD.</li><li>• The UUT will dial into VoiceMail.</li><li>• The UUT will have 1 message from Phone A.</li><li>• Once the message is deleted, the MWI indicator will turn off.</li></ul>   |
| Pass / Fail                                     | Passed  |
| Test Notes                                      | Test performed on Build Polycom 450 (FW 3.1.0.0147) on BE-C.1.4.  |
| Author  | spitts  |

### 4.3.2 Do Not Disturb

| Test Case Interop-7: Do Not Disturb |  |
|-------------------------------------|--|
| Summary                             | This test verifies the operation of Do Not Disturb.  |
| Step(s)                             | <ol style="list-style-type: none"><li>1. Place a call from Phone A to the UUT.</li><li>2. End the call.</li><li>3. Select Do Not Disturb on the UUT.</li><li>4. Place a call from Phone A to the UUT.</li><li>5. Disable Do Not Disturb on the UUT.</li><li>6. Place a call from Phone A to the UUT.</li></ol> |
| Expected Result(s)                  | <ul style="list-style-type: none"><li>• UUT rings, then a two-way voice path will be established when the UUT is answered.</li><li>• UUT will not ring and the call will go to VoiceMail.</li><li>• A two-way voice path will be established from the UUT to Phone A.</li></ul>                                |
| Pass / Fail                         | Passed   |
| Test Notes                          | Test performed on Build Polycom 450 (FW 3.1.0.0147) on BE-C.1.4.   |
| Author                              | spitts   |

## Section 5: Glossary of Common Terms

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

| Term | Definition  |
|------|---|
| 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. |
| 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  |