Asterisk Business Edition™
Version C.2.3.2
Digium Partner Certification

Interoperability Report
Polycom SoundPoint IP 670
Firmware Version 3.1.2

Rev. A

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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	C.2.3.2	

1.1.2 Partner Equipment Tested (UUTs)

Partner	Product	Version	Remarks
Polycom	SoundPoint IP 670	3.1.2	

The Polycom SoundPoint IP 670 is a SIP desktop phone with color display.

- **Key Features and Benefits**
  - Backlit 320 x 160 pixel color display
  - Advanced SIP functionality, including shared lines, busy lamp field, and presence
  - Polycom HD Voice™ technology for improved voice quality
  - Integrated Gigabit Ethernet (GigE) switch to enable bandwidth-intensive applications
  - Six lines in stand-alone mode and 34 lines with three Polycom SoundPoint IP Color Expansion Modules

    Built-in USB port to support applications, such as Local Call Recording from the Polycom Productivity Suite

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 670
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 the configurations files for the Asterisk server used during testing.

sip.conf

```
[general]

;*****
;*UUT*
;*****
[6670]
type=friend
username=6670
secret=6670
host=dynamic
context=testing
disallow=all
allow=ulaw
qualify=yes
subscribecontext=BLF_Enable
mailbox=6670

;*********
;*Phone A*
;*********
[6051]
type=friend
username=6051
secret=6051
host=dynamic
context=testing
disallow=all
allow=ulaw
qualify=1000
subscribecontext=BLF_Enable
mailbox=6051

;*********
;*Phone B*
;*********
[7000]
type=friend
```

```
username=7000
secret=7000
host=dynamic
context=testing
disallow=all
allow=ulaw
qualify=yes
subscribecontext=BLF_Enable
mailbox=7000
```

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 => 6051,hint,SIP/6051
exten => 7000,hint,SIP/7000
exten => 6670,hint,SIP/6670
```

voicemail.conf

```
[default]
6051 => 6051,Aastra 51i,root@localhost
7000 => 7000,Polycom 7000,root@localhost
6670 => 6670,Polycom 670,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 PC-8: SIP Registration	
Summary	This test is to ensure the UUT can register and authenticate to the Asterisk server successfully.
Step(s)	Configure the phone to register to the Asterisk server.
Expected Result(s)	The phone is authenticated and registers successfully.
Pass / Fail	Passed
Test Notes	Test performed on Build Polycom-IP-670-3.1.2-C.2.3.2.
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 PC-7: Outbound Call	
Summary	This test is to ensure the UUT can place outgoing calls.
Step(s)	1. Dial from the UUT to Phone A. 2. Verify the UUT receives ringback. 3. Verify that Phone A receives the Caller ID from the UUT.
Expected Result(s)	• 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-IP-670-3.1.2-C.2.3.2.
Author	spimental

4.2.2 Inbound Call

Test Case PC-6: Inbound Call	
Summary	This test is to ensure the UUT can receive incoming calls.
Step(s)	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)	• 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-IP-670-3.1.2-C.2.3.2.
Author	spimental

4.2.3 Call History

Test Case PC-3: Call History	
Summary	This test verifies the operation of the Call History feature.
Step(s)	1. Using the phone LCD menu navigation, clear the Call History records in the UUT. Note that most phone have history for: Placed Calls, Received Calls, Missed Calls. Some phone with limited feature sets may only have: 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)	• All Call History records will be cleared from the phone. • The Call History in the UUT will show: ◦ 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-IP-670-3.1.2-C.2.3.2.
Author	spimental

4.2.4 Hold and Resume

Test Case PC-4: Hold and Resume	
Summary	This test verifies the operation of Hold and Resume.
Step(s)	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)	• 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-IP-670-3.1.2-C.2.3.2.
Author	spimental

4.2.5 Attended Transfer

Test Case PC-5: Attended Transfer	
Summary	This test verifies the functionality of the Attended Call Transfer Feature.
Step(s)	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)	• 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-IP-670-3.1.2-C.2.3.2.
Author	spimental

4.2.6 Unattended Transfer

Test Case PC-2: Unattended Transfer	
Summary	This test verifies the functionality of the Unattended Call Transfer Feature.
Step(s)	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)	• 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 transfer the call.
Pass / Fail	Passed
Test Notes	Test performed on Build Polycom-IP-670-3.1.2-C.2.3.2.
Author	spimental

4.2.7 Conferencing

Test Case PC-1: Conferencing	
Summary	This test verifies the operation of phone-managed conferencing.
Step(s)	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)	• 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-IP-670-3.1.2-C.2.3.2.
Author	spimental

4.2.8 Forwarding

Test Case PC-9: Forwarding	
Summary	This test verifies the operation of Call Forwarding.
Step(s)	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)	• 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-IP-670-3.1.2-C.2.3.2.
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 PC-10: Message Waiting Indication	
Summary	This test verifies the operation of Message Waiting Indicator.
Step(s)	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)	• 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-IP-670-3.1.2-C.2.3.2.
Author	spimental

4.3.2 Do Not Disturb

Test Case PC-11: Do Not Disturb	
Summary	This test verifies the operation of Do Not Disturb.
Step(s)	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)	• 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-IP-670-3.1.2-C.2.3.2.
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 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