Asterisk Business Edition™
Version B.2.5.12
Digium Partner Certification

Interoperability Report
Siemens OpenStage 15
Firmware Version V2

Rev. A

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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.

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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.2.5.12	

1.1.2 Partner Equipment Tested (UUTs)

Partner	Product	Version	Remarks
Siemens	OpenStage 15	V2	

The Siemens OpenStage 15 is a desktop SIP phone.

- **Key Features and Benefits**
    - Supported Codecs: G.711, G.729AB, G.722 Wideband (7kHz)
    - Standard based SIP support according to RFC 3261 (with SIP software)
    - Full-duplex hands-free talking with high quality housing microphone and loudspeaker
    - Computer Telephony Integration (CTI for HFA), Third party call control (for SIP)
    - Remote administration through Web-based management and HiPath Deployment Service
    - 2 ports 10/100Base-T built-in Ethernet switch
    - Power over Ethernet (PoE) according to IEEE 802.3af or external power supply (EU, US and UK power adapters)
    - Sidecar module support for OpenStage Key Module 15
    - 3 navigation keys
    - 8 free programmable keys (with paper labels)
    - 3 fixed function keys for speaker, messages and menu with red LEDs
    - Volume keys (Loudspeaker/+/-)
    - not tiltable LCD 2 line display
    - Wall mountable

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	Siemens OpenStage 15
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

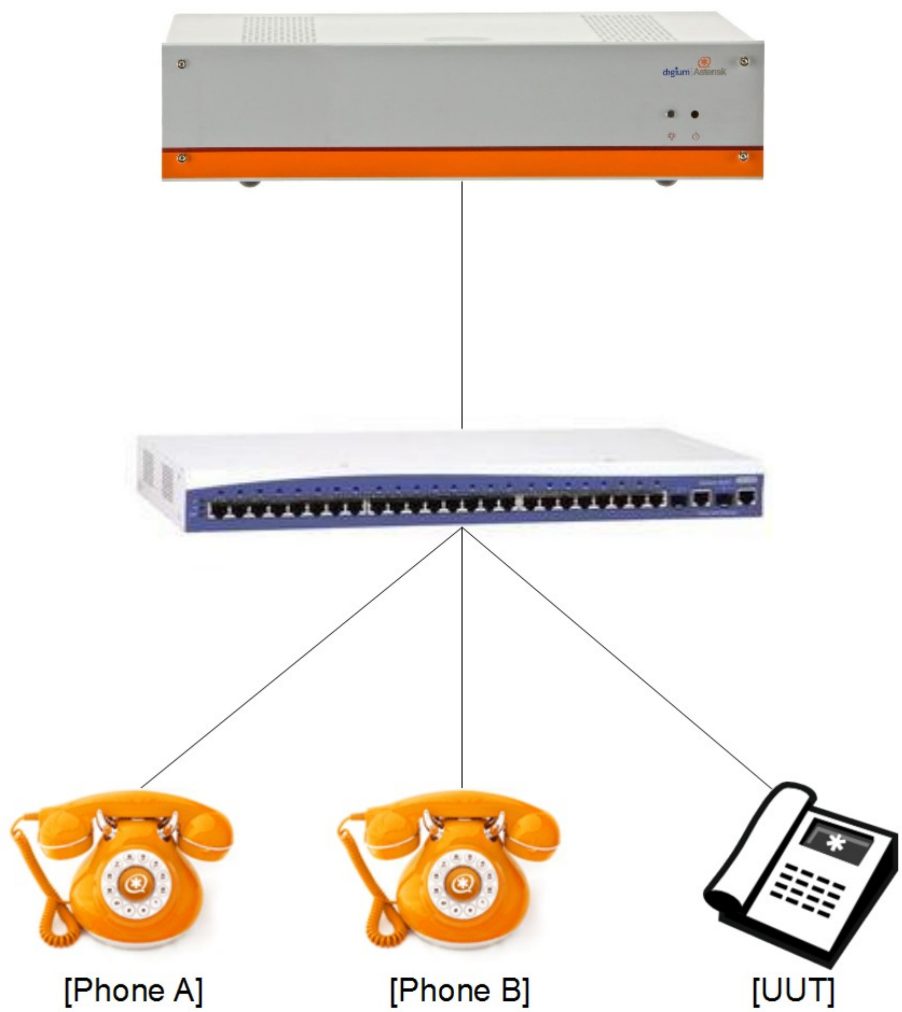
An isolated test network was created using an Adtran NetVanta switch and a PC-based server running Asterisk Business Edition. The partner phone (UUT) was connected to the test network via the Adtran switch. Each feature listed in this document was tested by placing calls to and from the UUT and the Asterisk 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,Siemens OpenStage 15,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.

4.1.1 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 Siemens-OpenStage-15-V2-BE.B.2.5.12.
Author	spimental

4.1.2 Outbound Call

Test Case PC-7: Outbound Call	
Summary	This test verifies the functionality of placing outgoing calls.
Step(s)	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)	• 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 Siemens-OpenStage-15-V2-BE.B.2.5.12.
Author	spimental

4.1.3 Inbound Call

Test Case PC-6: Inbound Call	
Summary	This test verifies the functionality of receiving 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 Siemens-OpenStage-15-V2-BE.B.2.5.12.
Author	spimental

4.1.4 Call History

Test Case PC-3: Call History	
Summary	This test verifies the functionality 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 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)	• 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 Siemens-OpenStage-15-V2-BE.B.2.5.12.
Author	spitts

4.1.5 Hold and Resume

Test Case PC-4: Hold and Resume	
Summary	This test verifies the functionality of the Hold and Resume feature.
Step(s)	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 Siemens-OpenStage-15-V2-BE.B.2.5.12.
Author	spitts

4.1.6 Attended Transfer

Test Case PC-5: Attended Transfer	
Summary	This test verifies the functionality of the Attended 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 Siemens-OpenStage-15-V2-BE.B.2.5.12.
Author	spitts

4.1.7 Unattended Transfer

Test Case PC-2: Unattended Transfer	
Summary	This test verifies the functionality of the Unattended 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 transfers the call.
Pass / Fail	Passed
Test Notes	Test performed on Build Siemens-OpenStage-15-V2-BE.B.2.5.12.
Author	spitts

4.1.8 Conferencing

Test Case PC-1: Conferencing	
Summary	This test verifies the functionality 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 Siemens-OpenStage-15-V2-BE.B.2.5.12.
Author	spitts

4.1.9 Forwarding

Test Case PC-9: Forwarding	
Summary	This test verifies the functionality of the Call Forwarding feature.
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 Siemens-OpenStage-15-V2-BE.B.2.5.12.
Author	spimental

4.1.10 Message Waiting Indicator

Test Case PC-10: Message Waiting Indication	
Summary	This test verifies the functionality of the Message Waiting Indicator feature.
Step(s)	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 Siemens-OpenStage-15-V2-BE.B.2.5.12.
Author	spitts

4.1.11 Do Not Disturb

Test Case PC-11: Do Not Disturb	
Summary	This test verifies the functionality of the Do Not Disturb feature.
Step(s)	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 Siemens-OpenStage-15-V2-BE.B.2.5.12.
Author	spitts

4.1.12 Codec G.729

Test Case PC-14: Codec G.729	
Summary	This test verifies the functionality of the G.729 codec.
Step(s)	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)	• 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 Siemens-OpenStage-15-V2-BE.B.2.5.12.
Author	spimental

4.1.13 Codec G.722

Test Case PC-15: Codec G.722	
Summary	This test verifies the functionality of the G.722 codec.
Step(s)	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)	• 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 Siemens-OpenStage-15-V2-BE.B.2.5.12.
Author	spimental

4.1.1 DTMF Mode Inband

Test Case PC-16: DTMF Mode Inband	
Summary	This test verifies the functionality of the inband DTMF mode.
Step(s)	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)	• 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 Siemens-OpenStage-15-V2-BE.B.2.5.12.
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