Asterisk Business Edition™
Version C.2.3.2
Digium Partner Certification

Interoperability Report
AMTELCO XDS H.100
2/4-Wire E&M Line Interface Board
(8-port) 257A059

Rev. A

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

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.

TABLE OF CONTENTS

Section 1: Executive Summary....4
1.1 Products Tested....4
1.1.1 Asterisk Business Edition....4
1.1.2 Partner Equipment Tested (UUTs)....5
1.2 Summary of Test Results....6

Section 2: Test Configuration....7
2.1 Description of Test Setup....7
2.2 Hardware Configuration....7
2.3 Test Setup Diagram....8

Section 3: Product Configuration....9

Section 4: Tests Performed....11
4.1 Basic Call – No Answer....11
4.2 Basic Call – Call Answer and Music-on-Hold....12
4.3 Busy Call Handling – Test 1....14
4.4 Busy Call Handling – Test 2....15
4.5 Congested Call Handling....16
4.6 Incomplete Call Address Handling....17
4.7 Radio Interface Call Handling....18
4.8 Radio Interface M-Lead Control....19

Section 5: Glossary of Common Terms....20

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	Remarks
AMTELCO	XDS H.100 2/4-Wire E&M Line Interface Board (8-port) 257A059	Referenced as “AMTELCO E&M Line Interface Board” throughout this report

The AMTELCO E&M Line Interface Board contains eight ports, each capable of 2-wire or 4-wire operation, using type I or type V signaling protocols. Signalling protocols and port types are programmed on a per-port basis. Individual ports may also be programmed for Radio Interface operation, allowing independent control of the E&M signalling leads on the application level. On-board DSP resources include DTMF detection and generation, as well as energy detection, North American / European call progress tones, and a 1004Hz calibration tone. Thirty-two independent bi-directional voice resource channels are also included.

1.2 Summary of Test Results

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

Test	XDS H.100
Basic Call – No Answer	✓
Basic Call – Call Answer and Music-on-Hold	✓
Busy Call Handling – Test 1	✓
Busy Call Handling – Test 2	✓
Congested Call Handling	✓
Incomplete Call Address Handling	✓
Radio Interface Call Handling	✓
Radio Interface M-Lead Control	✓

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 2.3.

2.1 Description of Test Setup

The test setup utilizes a Digium TDM410P (or equivalent) and an AMTELCO E&M Line Interface Board running in an Asterisk environment to provide functional testing of the AMTELCO E&M Line Interface Board.

The Asterisk environment is created by installing TDM410P drivers and the Asterisk system on a computer that is using the Linux operating system. When the TDM410P driver and Asterisk installation is complete, the AMTELCO channel driver is installed to provide access to the AMTELCO E&M Line Interface Board.

2.2 Hardware Configuration

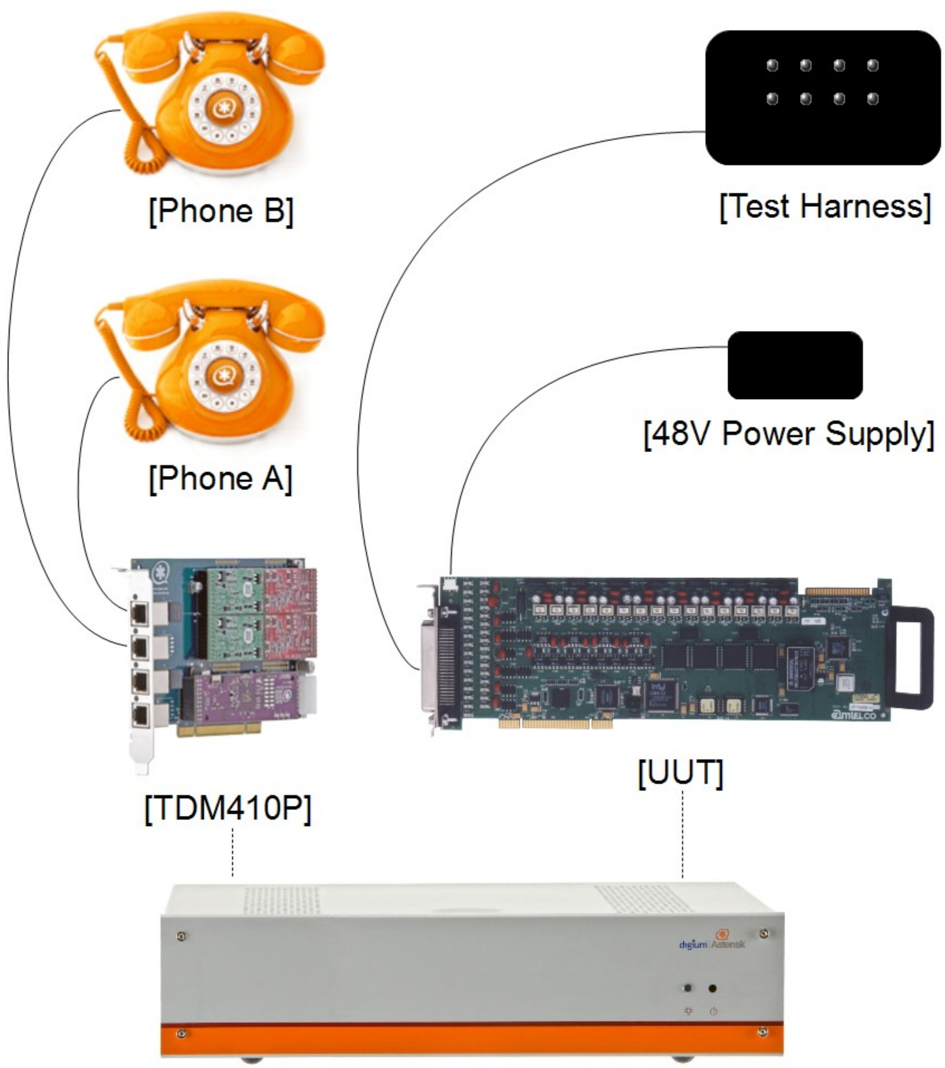
The test setup requires the following hardware configuration:

- Digium TDM410P with FXS modules on ports 1 and 2
- Two analog telephones connected to ports 1 and 2 on the Digium TDM410P
- AMTELCO E&M Line Interface Board
- 48V power supply connected to J2 on the AMTELCO E&M Line Interface Board (Attach the 4-pin red power supply connector to J2 with the yellow wire toward the back of the card)
- Test harness connected to J1 on the AMTELCO E&M Line Interface Board

The test harness connects even ports to odd ports on the AMTELCO E&M Line Interface Board. This allows a call placed from a telephone connected to port 1 on the TDM410P to travel into the first E&M port, out the second E&M port, and then to terminate on a telephone connected to port 2 on the TDM410P. The test harness also has LEDs used to monitor and verify the states of the signalling leads on the AMTELCO E&M Line Interface Board.

2.3 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.

> **Note:** In the example files listed below, “PE00951” refers to an AMTELCO E&M Line Interface Board with serial number 951. The numeric portion of the string should be replaced with the serial number of the actual board being tested.

/etc/asterisk/extensions.conf

```
[Internal]

exten => _20[1-3],1,Dial(AMTELCO/PE00951:1/${EXTEN})

exten => 204,1,Dial(AMTELCO/PE00951:1/20)

exten => 205,1,Dial(AMTELCO/PE00951:3)

exten => 206,1,Dial(AMTELCO/PE00951:4)

exten => 207,1,AmExec(M_Lead,AMTELCO/PE00951:4,Close)
exten => 207,n,Wait(1)
exten => 207,n,AmExec(M_Lead,AMTELCO/PE00951:4,Open)
exten => 207,n,Wait(1)
exten => 207,n,AmExec(M_Lead,AMTELCO/PE00951:4,Close)
exten => 207,n,Wait(1)
exten => 207,n,AmExec(M_Lead,AMTELCO/PE00951:4,Open)
exten => 207,n,Hangup()

[Incoming]

exten => _20[1-2],1,Dial(DAHDI/${EXTEN:2:1})
```

/etc/dahdi/system.conf

```
fxoks = 1,2
fxsks = 3,4
loadzone = us
defaultzone = us
echocanceller = mg2,1-4
```

/etc/asterisk/chan_dahdi.conf

```
; Note that the following text is added to the bottom of the default
; chan_dahdi.conf file.)

;FXS Modules
group = 1
signalling = fxo_ks
context = Internal
channel = 1-2
```

/etc/asterisk/amtelco.conf

```
[channels]

group = 1
signalling = em
context = Internal
emconfig = 4,type_v
channel = PE00951:1

signalling = manual
channel = PE00951:3,4

context = Incoming
didconfig = immediate,immediate
immediate = no
signalling = em
channel = PE00951:2
```

Section 4: Tests Performed

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

4.1 Basic Call – No Answer

Test Case PC-17: Basic Call – No Answer	
Summary	This test verifies the functionality of the following: • Asterisk can make an outbound call through an E&M port. • An E&M port can detect an incoming call, collect addressing information, and present the call to Asterisk. • An E&M port plays ringback to Asterisk when instructed by Asterisk to do so. • An outbound call on an E&M port disconnects properly when instructed by Asterisk to do so. • If an inbound E&M call is disconnected before the called party answers, the call is disposed of appropriately.
Step(s)	1. Dial 201 from TDM410P port 2. 2. Hang up TDM410P port 2.
Expected Result(s)	1. TDM410P port 1 rings; TDM410P port 2 hears ringback; E&M port 2 E-Lead LED turns on. 2. TDM410P port 1 stops ringing; E&M port 2 E-Lead LED turns off.
Pass / Fail	Passed
Test Notes	Test performed on Build AMTELCO-XDS-H.100-E&M-8-port-257A059-ABE.C.2.3.2.
Author	spimental

4.2 Basic Call – Call Answer and Music-on-Hold

Test Case PC-18: Basic Call – Call Answer and Music-on-Hold	
Summary	This test verifies the functionality of the following: • Asterisk can make an outbound call through an E&M port. • An E&M port can detect an incoming call, collect addressing information, and present the call to Asterisk. • An E&M port plays ringback to Asterisk when instructed by Asterisk to do so. • E&M ports establish audio connections appropriately. • E&M ports respond to Music-on-Hold requests appropriately. • Connected calls are disposed of properly.
Step(s)	1. Dial ‘201’ from TDM410P port 2. 2. Answer TDM410P port 1. 3. Issue a hook flash on TDM410P port 1. 4. Issue another hook flash on TDM410P port 1. 5. Hang up TDM410P port 1. 6. Hang up TDM410P port 2.
Expected Result(s)	1. TDM410P port 1 rings; TDM410P port 2 hears ringback; E&M port 2 E-Lead LED turns on. 2. E&M port 1 E-Lead LED turns on; two-way audio is present on the two TDM410P ports. 3. Music-on-Hold is played to TDM410P port 2. 4. Music-on-Hold stops on TDM410P port 2; two-way audio is present on the two TDM410P ports. 5. E&M port 1 E-Lead LED turns off; E&M port 2 E-Lead LED turns off after the port 1 E-Lead LED turns off; Congestion (reorder) is played to TDM410P port 2. 6. None

Pass / Fail	Passed
Test Notes	Test performed on Build AMTELCO-XDS-H.100-E&M-8-port-257A059-ABE.C.2.3.2.
Author	spimental

4.3 Busy Call Handling – Test 1

Test Case PC-19: Busy Call Handling – Test 1	
Summary	This test verifies that a call to an E&M port that is in use returns a busy indication to Asterisk so that Asterisk plays busy to the originating call.
Step(s)	1. Dial '201' from TDM410P port 2. 2. Answer TDM410P port 1. 3. Issue a hook flash on TDM410P port 2. 4. Dial '201' from TDM410P port 2. 5. Hang up TDM410P ports 1 and 2.
Expected Result(s)	1. TDM410P port 1 rings; TDM410P port 2 hears ringback; E&M port 2 E-Lead LED turns on. 2. E&M port 1 E-Lead LED turns on; two-way audio is present on the two TDM410P ports. 3. Music-on-Hold is played to TDM410P port 1. 4. Busy is played to TDM410P port 2. 5. E&M ports 1 and 2 E-Lead LEDs turn off.
Pass / Fail	Passed
Test Notes	Test performed on AMTELCO-XDS-H.100-E&M-8-port-257A059-ABE.C.2.3.2.
Author	spimental

4.4 Busy Call Handling – Test 2

Test Case PC-20: Busy Call Handling – Test 2	
Summary	This test verifies that when an incoming call on an E&M port is destined to a device that is in use, the E&M port plays busy.
Step(s)	1. Dial ‘202’ from TDM410P port 2. 2. Hang up TDM410P port 2.
Expected Result(s)	1. TDM410P port 2 hears busy; E&M port 2 E-Lead LED turns on. 2. E&M port 2 E-Lead LED turns off.
Pass / Fail	Passed
Test Notes	Test performed on AMTELCO-XDS-H.100-E&M-8-port-257A059-ABE.C.2.3.2.
Author	spimental

4.5 Congested Call Handling

Test Case PC-21: Congested Call Handling	
Summary	This test verifies that when an invalid destination number is received from an incoming call on an E&M port, the E&M port passes congestion (reorder) through to Asterisk.
Step(s)	1. Dial '203' from TDM410P port 2. 2. Hang up TDM410P port 2.
Expected Result(s)	1. TDM410P port 2 hears congestion (reorder); E&M port 2 E-Lead LED turns on. 2. E&M ports 2 E-Lead LED turns off.
Pass / Fail	Passed
Test Notes	Test performed on Build AMTELCO-XDS-H.100-E&M-8-port-257A059-ABE.C.2.3.2.
Author	spimental

4.6 Incomplete Call Address Handling

Test Case PC-22: Incomplete Call Address Handling	
Summary	This test verifies that when an incomplete destination number is received from an incoming call on an E&M port, the E&M port passes congestion (reorder) through to Asterisk.
Step(s)	1. Dial ‘204’ from TDM410P port 2. 2. Hang up TDM410P port 2.
Expected Result(s)	1. E&M port 2 E-Lead LED turns on; Congestion (reorder) is played to TDM410P port 2 after approximately 8 seconds. 2. E&M port 2 E-Lead LED turns off.
Pass / Fail	Passed
Test Notes	Test performed on Build AMTELCO-XDS-H.100-E&M-8-port-257A059-ABE.C.2.3.2.
Author	spimental

4.7 Radio Interface Call Handling

Test Case PC-23: Radio Interface Call Handling	
Summary	This test verifies that calls can be placed to E&M ports configured for Radio Interface operation and that those call are disposed of properly.
Step(s)	1. Dial ‘205’ from TDM410P port 2. 2. Dial ‘206’ from TDM410P port 1. 3. Hang up both TDM410P ports. 4. Repeat steps 1 through 3.
Expected Result(s)	1. TDM410P port 2 hears silence; all E-Lead LEDs remain off. 2. Two-way audio is present between the two TDM410P ports; all E-Lead LEDs remain off. 3. None. 4. Original results are repeated.
Pass / Fail	Passed
Test Notes	Test performed on Build AMTELCO-XDS-H.100-E&M-8-port-257A059-ABE.C.2.3.2.
Author	spimental

4.8 Radio Interface M-Lead Control

Test Case PC-24: Radio Interface M-Lead Control	
Summary	This test verifies that the M-Lead Control can be manipulated via the dialplan for E&M ports configured for manual control.
Step(s)	1. Dial ‘207’ from TDM410P port 2. 2. Hang up TDM410P port 2.
Expected Result(s)	1. TDM410P port 2 hears silence; E&M port 3 E-Lead LED turns on for 1 second, off for 1 second, on for 1 second, and then off; TDM410P port 2 hears reorder after the E&M port 3 E-Lead LED turns off the second time. 2. None.
Pass / Fail	Passed
Test Notes	Test performed on Build AMTELCO-XDS-H.100-E&M-8-port-257A059-ABE.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 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