Asterisk Business Edition™
Version C.3.1
Digium Partner Certification

Interoperability Report
Dialogic Media Gateway 1000
Firmware Version 6.0.SU3.2.001

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.3.1	

1.1.2 Partner Equipment Tested (UUTs)

Partner	Product	Version	Remarks
Dialogic	DMG1000	6.0.SU3.2.001	

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 Test Matrix

Test Case	DMG1000
PC-45	✓
PC-46	✓
PC-47	✓
PC-48	✓
PC-49	✓
PC-50	✓
PC-51	✓
PC-52	✓
PC-53	✓
PC-54	✓
PC-55	✓
PC-56	✓
PC-57	✓
PC-58	✓
PC-59	✓
PC-60	✓
PC-61	✓
PC-62	✓
PC-63	✓
PC-64	✓
PC-65	✓
PC-66	✓
PC-67	✓
PC-68	✓
PC-69	✓
PC-70	✓
PC-71	✓

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

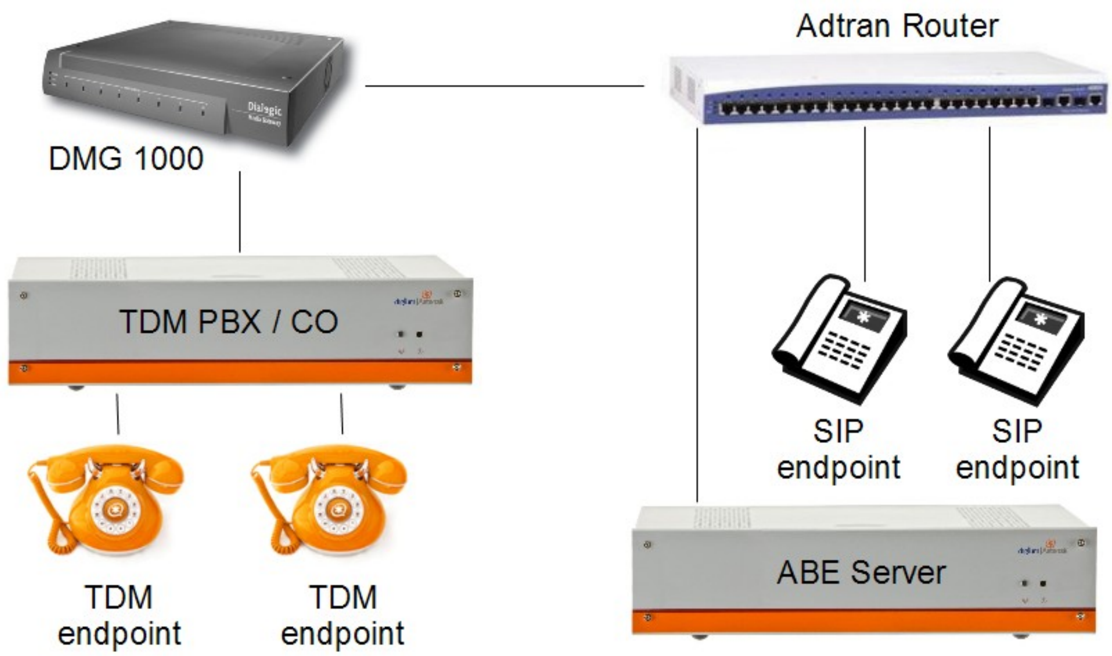
An isolated test network was created using an Adtran NetVanta switch, two PC-based Asterisk servers, and the partner media gateway (DMG1000). One Asterisk server used Asterisk Business Edition and the other acted as the “Central Office” or TDM PBX. Two SIP phones were directly connected to the network and two TDM analog phones were connected to the TDM PBX.

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 SIP to TDM calls

3.1.1 Asterisk Configuration

In extensions.conf add

```
exten => _121xxxx,1,Dial(SIP/${EXTEN:4}@10.10.11.121)
```

The first occurrence of “121” indicates the gateway used to route the call. “10.10.11.121” is the IP Address of the gateway.

3.1.2 Dialogic Media Gateway Configuration

No configuration required. Default routing table can be used.

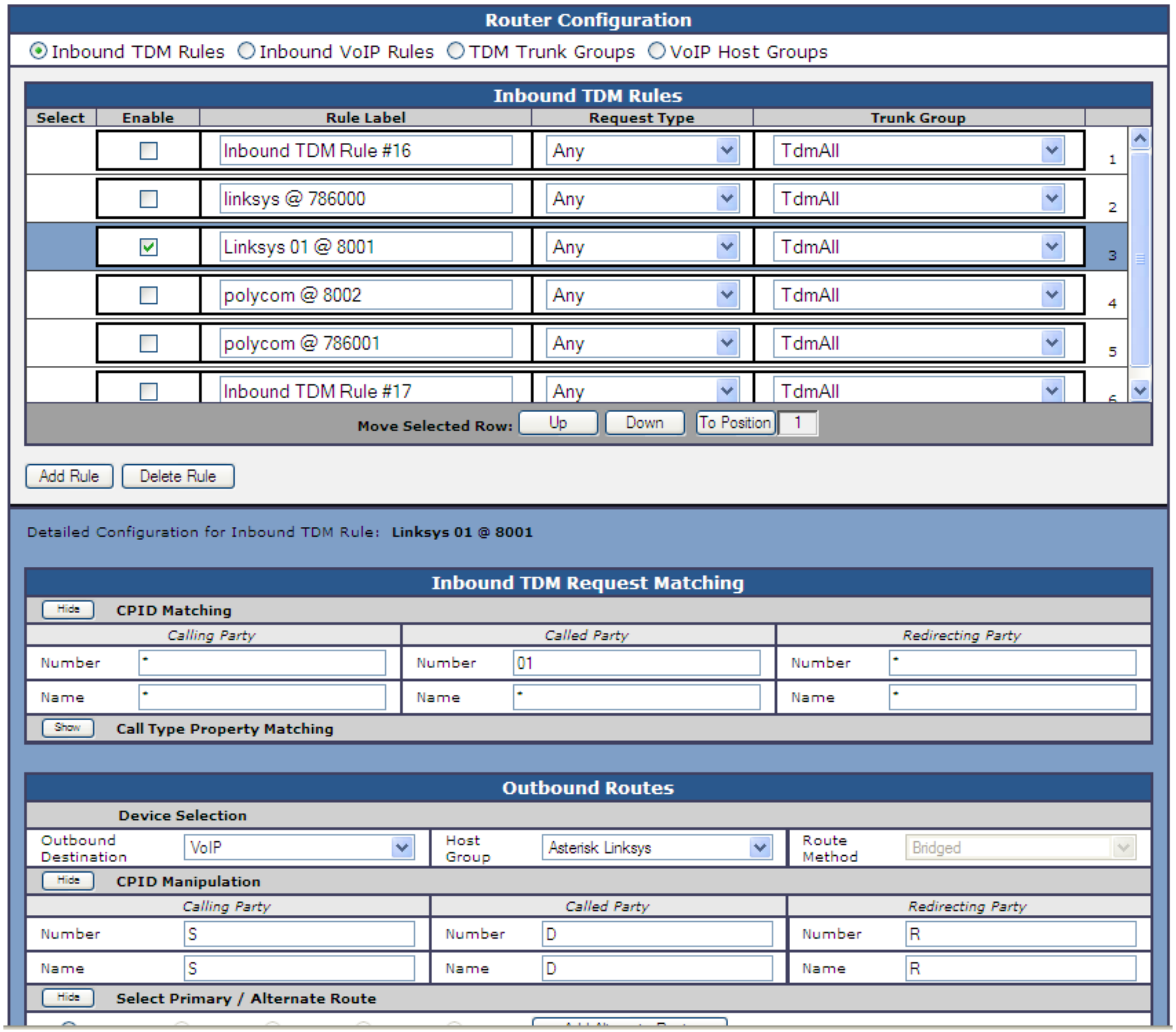
3.2 TDM to SIP calls

3.2.1 Asterisk Configuration

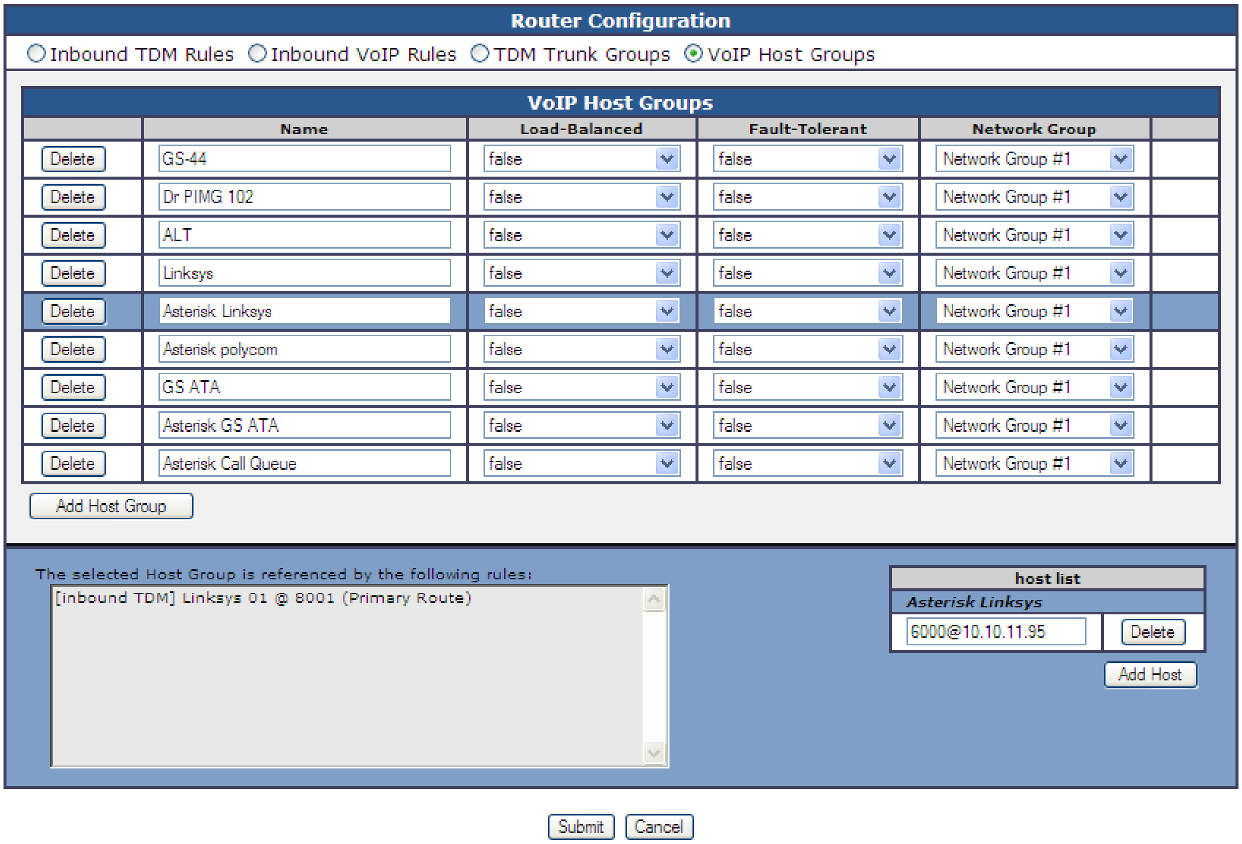
Using either the GUI or the configuration files, add extensions for the SIP phones.

3.2.2 Dialogic Media Gateway Configuration

Call routing configuration is shown in the following screenshot. One or more entries may be required to complete the test. The following screenshot shows how all incoming TDM calls to extension 01 are routed.



Each incoming TDM entry can route calls to one of ten VoIP HostGroups. This is shown in the following screenshot.



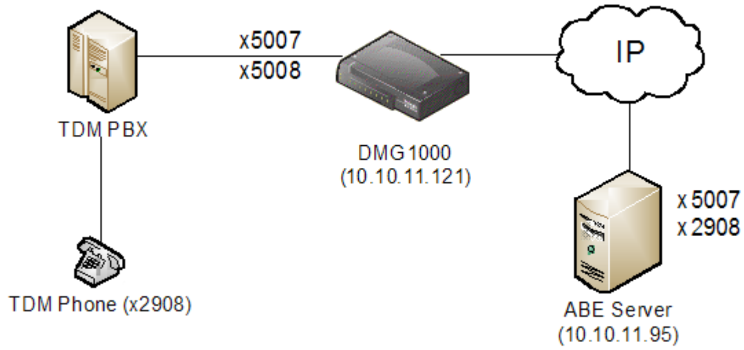
These two entries combined specify that incoming TDM calls to extension 01 are routed to extension 6000 on the Asterisk Business Edition server.

3.3 Message Waiting Indicator

The Asterisk's Server VoiceMail system can supplement legacy PBX systems. In this context, MWIs were tested in the direction of VoIP to TDM.

As a simplification, Asterisk's dial plan was expanded so that extensions spanned the available extensions on the legacy PBX.

In the figure below, the extensions are simply examples. The key to making MWIs work in this setup is to provide two extensions on the DMG routing table: one routing entry routes calls from the extension to the VoiceMailMain application to provide access for the user, and the second routing entry routes the forwarded calls to the user's voice mailbox.



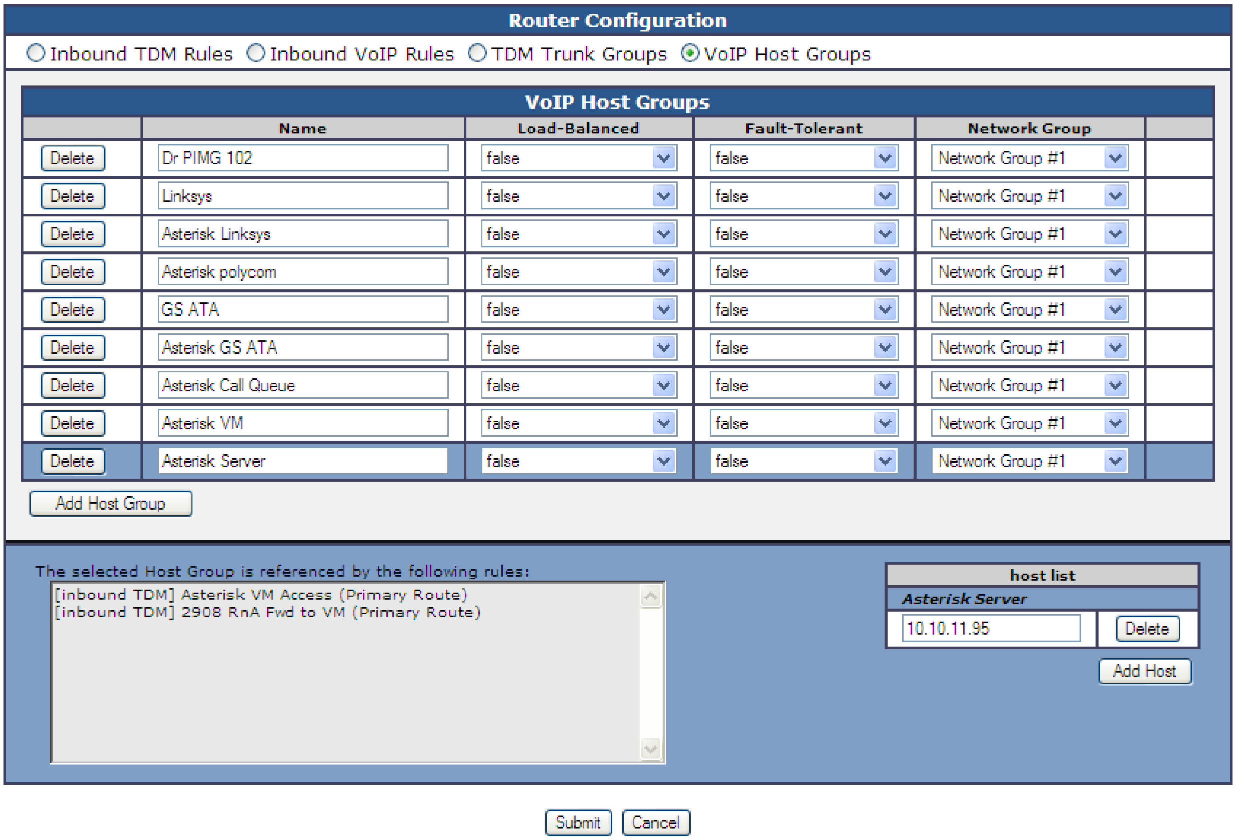
Notes:
1) From the legacy PBX, the gateway can be accessed via two extensions: 5007 and 5008
2) Extension 5007 provides Voice Mail access to users
3) The TDM phone has a mailbox on the Asterisk Server

3.3.1 Asterisk Configuration

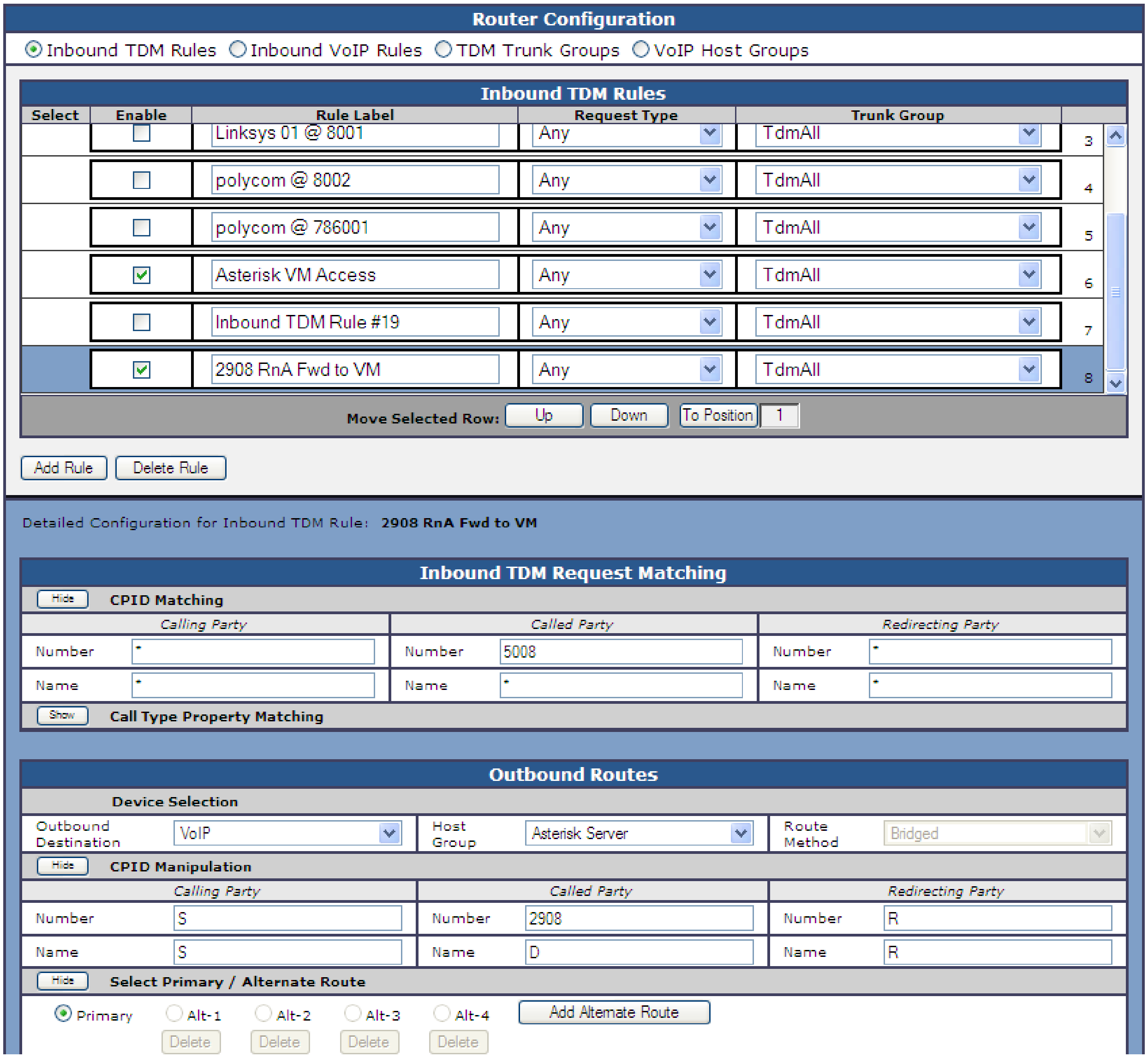
Two entries need to be added to the default context in the extensions.conf file:

```
exten => 5008, 1, VoiceMail(2908@default,u)
exten => 5007, 1, VoiceMailMain(${CALLERID(num)}@default)
```

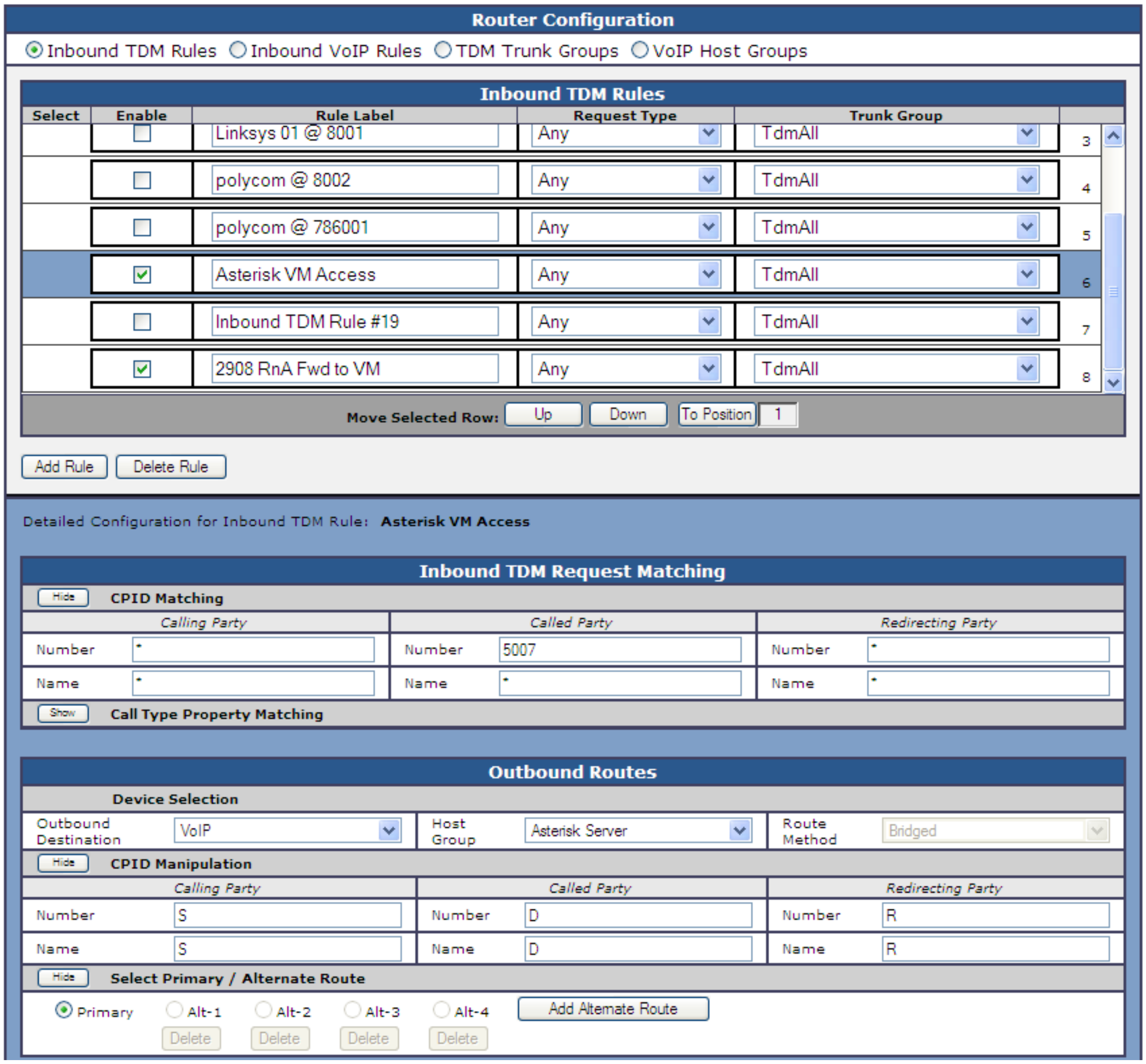
3.3.2 Dialogic Media Gateway Configuration



The screenshot shown above includes the VoIP HostGroup entry which contains the Asterisk Business Edition Server IP address.



The screenshot shown above utilizes the “Asterisk Server” VoIP HostGroup. This routing table entry routes the incoming TDM call to extension 5008 to the voice mailbox for extensions 2908.



The routing table entry shown above provides voicemail access to TDM phone users by routing calls to extension 5007 to the Asterisk Business Edition Server. As explained earlier, the Asterisk Business Edition Server will determine the Calling Party information and provide access to the correct mailbox.

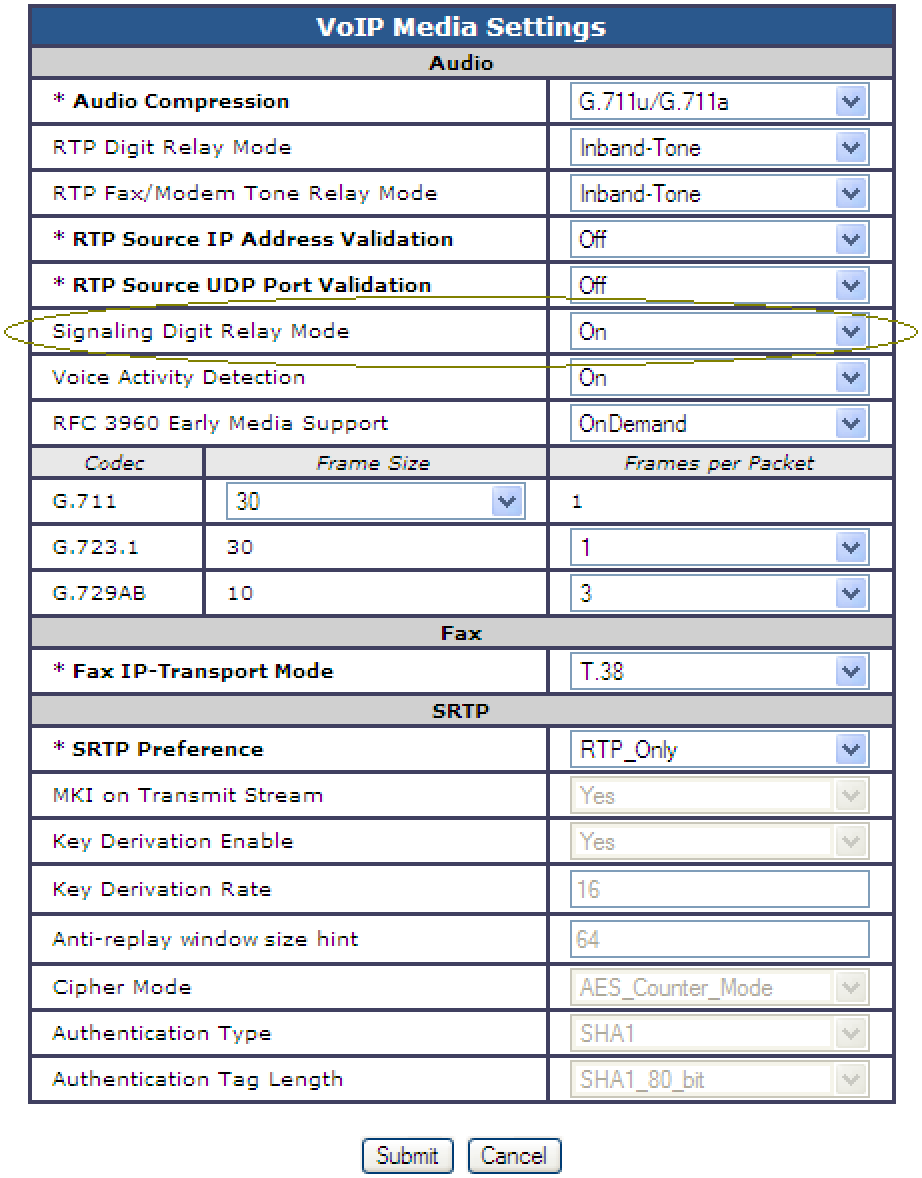
3.4 DTMF

3.4.1 Asterisk Configuration

No configuration is required.

3.4.2 Dialogic Media Gateway Configuration

Under Configuration -> VoIP -> Media, the DMG should be configured as shown in the following screenshot.



Section 4: Tests Performed

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

4.1.1 Test Case PC-45

PC-45: SIP to TDM call	
Step(s)	1. Place a call originating at the SIP endpoint to a TDM endpoint.
Expected Result(s)	• Call is connected. • Voice test passes. • DTMF test passes. • CPID is correct.
Pass / Fail	Passed
Test Notes	Test performed on Build Dialogic-DMG1000-6.0.SU3.2.001-BE.C.3.1.
Author	spimental

4.1.2 Test Case PC-46

PC-46: SIP supervised transfer to TDM	
Step(s)	1. Establish a SIP/TDM call. 2. Initiate a transfer from the SIP endpoint to a second TDM endpoint. 3. Once the second TDM endpoint answers, complete the transfer from the SIP endpoint.
Expected Result(s)	• SIP endpoint successfully disconnects. • TDM endpoints are connected. • Voice test passes. • DTMF test passes. • CPID is correct.
Pass / Fail	Passed
Test Notes	Test performed on Build Dialogic-DMG1000-6.0.SU3.2.001-BE.C.3.1.
Author	spimental

4.1.3 Test Case PC-47

PC-47: SIP supervised transfer to TDM; TDM disconnects before transfer is complete	
Step(s)	1. Establish a SIP/TDM call . 2. Initiate a transfer from the SIP endpoint to a second TDM endpoint. 3. Once the second TDM endpoint answers, disconnect the first TDM endpoint.
Expected Result(s)	• First TDM endpoint successfully disconnects . • SIP endpoint and second TDM endpoints remain connected. • Voice test passes. • DTMF test passes. • CPID is correct.
Pass / Fail	Passed
Test Notes	Test performed on Build Dialogic-DMG1000-6.0.SU3.2.001-BE.C.3.1.
Author	spimental

4.1.4 Test Case PC-48

PC-48: SIP unsupervised transfer to TDM	
Step(s)	1. Establish a SIP/TDM call. 2. Initiate an unsupervised transfer from the SIP endpoint to a second TDM endpoint. 3. Immediately complete the transfer by dialing.
Expected Result(s)	• SIP endpoint successfully disconnects. • TDM endpoints are connected. • Voice test passes. • DTMF test passes. • CPID is correct.
Pass / Fail	Passed
Test Notes	Test performed on Build Dialogic-DMG1000-6.0.SU3.2.001-BE.C.3.1.
Author	spitts

4.1.5 Test Case PC-49

PC-49: SIP unsupervised transfer to TDM; first TDM disconnects during unsupervised transfer	
Step(s)	1. Establish a SIP/TDM call. 2. Initiate an unsupervised transfer from the SIP endpoint to a second TDM endpoint. 3. Complete the transfer from the SIP endpoint by dialing and simultaneously disconnecting the first TDM endpoint.
Expected Result(s)	• All endpoints are disconnected
Pass / Fail	Passed
Test Notes	Test performed on Build Dialogic-DMG1000-6.0.SU3.2.001-BE.C.3.1.
Author	spitts

4.1.6 Test Case PC-50

PC-50: SIP supervised transfer to SIP	
Step(s)	1. Establish a SIP/TDM call. 2. Initiate a transfer from the SIP endpoint to a second SIP endpoint. 3. Once the second SIP endpoint answers, complete the transfer from the SIP endpoint
Expected Result(s)	• First SIP endpoint successfully disconnects. • TDM and second SIP endpoints are connected. • Voice test passes. • DTMF test passes. • CPID is correct.
Pass / Fail	Passed
Test Notes	Test performed on Build Dialogic-DMG1000-6.0.SU3.2.001-BE.C.3.1.
Author	spitts

4.1.7 Test Case PC-51

PC-51: SIP supervised transfer to SIP; TDM disconnects before transfer is complete	
Step(s)	1. Establish a SIP/TDM call. 2. Initiate a transfer from the SIP endpoint to a second SIP endpoint. 3. Once the second SIP endpoint answers, disconnect the first TDM endpoint.
Expected Result(s)	• TDM endpoint successfully disconnects. • First and second SIP endpoints remain connected. • Voice test passes. • DTMF test passes. • CPID is correct.
Pass / Fail	Passed
Test Notes	Test performed on Build Dialogic-DMG1000-6.0.SU3.2.001-BE.C.3.1.
Author	spitts

4.1.8 Test Case PC-52

PC-52: SIP unsupervised transfer to SIP	
Step(s)	1. Establish a SIP/TDM call. 2. Initiate an unsupervised transfer from the SIP endpoint to a second SIP endpoint. 3. Immediately complete the transfer by dialing.
Expected Result(s)	• First SIP endpoint successfully disconnects. • TDM and second SIP endpoints are connected. • Voice test passes. • DTMF test passes. • CPID is correct.
Pass / Fail	Passed
Test Notes	Test performed on Build Dialogic-DMG1000-6.0.SU3.2.001-BE.C.3.1.
Author	spitts

4.1.9 Test Case PC-53

PC-53: SIP unsupervised transfer to SIP; TDM disconnects during the unsupervised transfer	
Step(s)	1. Establish a SIP/TDM call. 2. Initiate an unsupervised transfer from the SIP endpoint to a second SIP endpoint. 3. Complete the transfer from the first SIP endpoint by dialing and simultaneously disconnecting the TDM endpoint.
Expected Result(s)	• All endpoints are disconnected.
Pass / Fail	Passed
Test Notes	Test performed on Build Dialogic-DMG1000-6.0.SU3.2.001-BE.C.3.1.
Author	spimental

4.1.10 Test Case PC-54

PC-54: TDM to SIP call	
Step(s)	1. Place a call originating from a TDM endpoint to a SIP endpoint.
Expected Result(s)	• Call is connected. • Voice test passes. • DTMF test passes. • CPID is correct.
Pass / Fail	Passed
Test Notes	Test performed on Build Dialogic-DMG1000-6.0.SU3.2.001-BE.C.3.1.
Author	spitts

4.1.11 Test Case PC-55

PC-55: TDM supervised transfer to SIP	
Step(s)	1. Establish a TDM/SIP call. 2. Initiate a transfer from the TDM endpoint to second a SIP endpoint. 3. Once the second SIP endpoint answers, complete the transfer from the TDM endpoint.
Expected Result(s)	• TDM endpoint successfully disconnects. • SIP endpoints are connected. • Voice test passes. • DTMF test passes. • CPID is correct.
Pass / Fail	Passed
Test Notes	Test performed on Build Dialogic-DMG1000-6.0.SU3.2.001-BE.C.3.1.
Author	spitts

4.1.12 Test Case PC-56

PC-56: TDM supervised transfer to SIP; SIP disconnects before transfer is complete	
Step(s)	1. Establish a TDM/SIP call. 2. Initiate a transfer from the TDM endpoint to a second SIP endpoint. 3. Once the second SIP endpoint answers, disconnect the first SIP endpoint.
Expected Result(s)	• First SIP endpoint successfully disconnects. • TDM endpoints and second SIP endpoints remain connected. • Voice test passes. • DTMF test passes. • CPID is correct.
Pass / Fail	Passed
Test Notes	Test performed on Build Dialogic-DMG1000-6.0.SU3.2.001-BE.C.3.1.
Author	spimental

4.1.13 Test Case PC-57

PC-57: TDM unsupervised transfer to SIP	
Step(s)	1. Establish a TDM/SIP call. 2. Initiate an unsupervised transfer from the TDM endpoints to a second SIP endpoints. 3. Immediately complete the transfer by dialing.
Expected Result(s)	• TDM endpoints successfully disconnects. • SIP endpoints are connected. • Voice test passes. • DTMF test passes. • CPID is correct.
Pass / Fail	Passed
Test Notes	Test performed on Build Dialogic-DMG1000-6.0.SU3.2.001-BE.C.3.1.
Author	spimental

4.1.14 Test Case PC-58

PC-58: TDM unsupervised transfer to SIP; the first SIP phone disconnects during the unsupervised transfer	
Step(s)	1. Establish a TDM/SIP call 2. Initiate an unsupervised transfer from the TDM endpoint to a second SIP endpoint. 3. Complete the transfer from the TDM endpoint by dialing and simultaneously disconnecting the first SIP endpoint.
Expected Result(s)	• All endpoints are disconnected
Pass / Fail	Passed
Test Notes	Test performed on Build Dialogic-DMG1000-6.0.SU3.2.001-BE.C.3.1.
Author	spimental

4.1.15 Test Case PC-59

PC-59: TDM supervised transfer to TDM	
Step(s)	1. Establish a TDM/SIP call. 2. Initiate a transfer from the TDM endpoint to a second TDM endpoint. 3. Once the second TDM endpoint answers, complete the transfer from the TDM endpoint.
Expected Result(s)	• First TDM endpoint successfully disconnects. • SIP and second TDM endpoints are connected. • Voice test passes. • DTMF test passes. • CPID is correct.
Pass / Fail	Passed
Test Notes	Test performed on Build Dialogic-DMG1000-6.0.SU3.2.001-BE.C.3.1.
Author	spimental

4.1.16 Test Case PC-60

PC-60: TDM supervised transfer to TDM; SIP disconnects before transfer is complete	
Step(s)	1. Establish a TDM/SIP call. 2. Initiate a transfer from the TDM endpoint to a second TDM endpoint. 3. Once the second TDM endpoint answers, disconnect the first SIP endpoint.
Expected Result(s)	• SIP endpoint successfully disconnects. • First and second TDM endpoints remain connected. • Voice test passes. • DTMF test passes. • CPID is correct.
Pass / Fail	Passed
Test Notes	Test performed on Build Dialogic-DMG1000-6.0.SU3.2.001-BE.C.3.1.
Author	spimental

4.1.17 Test Case PC-61

PC-61: TDM unsupervised transfer to TDM	
Step(s)	1. Establish a TDM/SIP call. 2. Initiate an unsupervised transfer from the TDM endpoint to a second TDM endpoint. 3. Immediately complete the transfer by dialing.
Expected Result(s)	• First TDM endpoint successfully disconnects. • SIP and second TDM endpoints are connected. • Voice test passes. • DTMF test passes. • CPID is correct.
Pass / Fail	Passed
Test Notes	Test performed on Build Dialogic-DMG1000-6.0.SU3.2.001-BE.C.3.1.
Author	spimental

4.1.18 Test Case PC-62

PC-62: TDM unsupervised transfer to TDM; SIP disconnects during the unsupervised transfer	
Step(s)	1. Establish a TDM/SIP call. 2. Initiate an unsupervised transfer from the TDM endpoint to a second TDM endpoint. 3. Complete the transfer from the first TDM endpoint by dialing and simultaneously disconnecting the SIP endpoint.
Expected Result(s)	• All endpoints are disconnected.
Pass / Fail	Passed
Test Notes	Test performed on Build Dialogic-DMG1000-6.0.SU3.2.001-BE.C.3.1.
Author	spimental

4.1.19 Test Case PC-63

PC-63: SIP forward all to SIP	
Step(s)	1. Configure the first SIP endpoint to forward all to a second SIP endpoint. 2. Initiate a call from a TDM endpoint to the first SIP endpoint. The call should be forwarded to the second SIP endpoint. 3. Complete the call by answering the second SIP endpoint.
Expected Result(s)	• TDM and second SIP endpoints are connected. • Voice test passes. • DTMF test passes. • CPID is correct.
Pass / Fail	Passed
Test Notes	Test performed on Build Dialogic-DMG1000-6.0.SU3.2.001-BE.C.3.1.
Author	spimental

4.1.20 Test Case PC-64

PC-64: SIP forward Ring No Answer (RNA) to SIP	
Step(s)	1. Configure the first SIP endpoint to forward RNA to a second SIP endpoint. 2. Initiate a call from a TDM endpoint to the first SIP endpoint. The call should be forwarded after several rings to the second SIP endpoint. 3. Complete the call by answering the second SIP endpoint.
Expected Result(s)	• TDM and second SIP endpoints are connected. • Voice test passes. • DTMF test passes. • CPID is correct.
Pass / Fail	Passed
Test Notes	Test performed on Build Dialogic-DMG1000-6.0.SU3.2.001-BE.C.3.1.
Author	spimental

4.1.21 Test Case PC-65

PC-65: SIP forward busy to SIP	
Step(s)	1. Configure the first SIP endpoint to forward on busy to a second SIP endpoint. 2. Make the first SIP endpoint busy. 3. Initiate a call from a TDM endpoint to the first SIP endpoint. The call should be forwarded to the second SIP endpoint. 4. Complete the call by answering the second SIP endpoint.
Expected Result(s)	• TDM and second SIP endpoints are connected. • Voice test passes. • DTMF test passes. • CPID is correct.
Pass / Fail	Passed
Test Notes	Test performed on Build Dialogic-DMG1000-6.0.SU3.2.001-BE.C.3.1.
Author	spimental

4.1.22 Test Case PC-66

PC-66: SIP forward all to TDM	
Step(s)	1. Configure the first SIP endpoint to forward all to a second TDM endpoint. 2. Initiate a call from a TDM endpoint to the first SIP endpoint. The call should be forwarded to the second TDM endpoint. 3. Complete the call by answering the second TDM endpoint.
Expected Result(s)	• TDM endpoints are connected. • Voice test passes. • DTMF test passes. • CPID is correct.
Pass / Fail	Passed
Test Notes	Test performed on Build Dialogic-DMG1000-6.0.SU3.2.001-BE.C.3.1.
Author	spimental

4.1.23 Test Case PC-67

PC-67: SIP forward RNA to TDM	
Step(s)	1. Configure the first SIP endpoint to forward on RNA to a second TDM endpoint. 2. Initiate a call from a TDM endpoint to the first SIP endpoint. The call should be forwarded after several rings to the second TDM endpoint. 3. Complete the call by answering the second TDM endpoint.
Expected Result(s)	• TDM endpoints are connected. • Voice test passes. • DTMF test passes. • CPID is correct.
Pass / Fail	Passed
Test Notes	Test performed on Build Dialogic-DMG1000-6.0.SU3.2.001-BE.C.3.1.
Author	spimental

4.1.24 Test Case PC-68

PC-68: SIP forward busy to TDM	
Step(s)	1. Configure the first SIP endpoint to forward on busy to a second TDM endpoint. 2. Make the first SIP endpoint busy. 3. Initiate a call from a TDM endpoint to the first SIP endpoint. The call should be forwarded to the second TDM endpoint. 4. Complete the call by answering the second TDM endpoint.
Expected Result(s)	• TDM endpoints are connected. • Voice test passes. • DTMF test passes. • CPID is correct.
Pass / Fail	Passed
Test Notes	Test performed on Build Dialogic-DMG1000-6.0.SU3.2.001-BE.C.3.1.
Author	spimental

4.1.25 Test Case PC-69

PC-69: TDM forward all to SIP	
Step(s)	1. Configure a TDM endpoint to forward all to a SIP endpoint. 2. Initiate a call from another TDM endpoint to the TDM endpoint that is configured to forward the call. The call should be forwarded to the SIP endpoint. 3. Complete the call by answering the SIP endpoint.
Expected Result(s)	• TDM to SIP endpoints are connected. • Voice test passes. • DTMF test passes. • CPID is correct.
Pass / Fail	Passed
Test Notes	Test performed on Build Dialogic-DMG1000-6.0.SU3.2.001-BE.C.3.1.
Author	spimental

4.1.26 Test Case PC-70

PC-70: TDM forward RNA to SIP	
Step(s)	1. Configure a TDM endpoint to forward on RNA to a SIP endpoint. 2. Initiate a call from another TDM endpoint to the TDM endpoint that is configured to forward the call. The call should be forwarded after several rings to the SIP endpoint. 3. Complete the call by answering the SIP endpoint.
Expected Result(s)	• TDM to SIP endpoints are connected. • Voice test passes. • DTMF test passes. • CPID is correct.
Pass / Fail	Passed
Test Notes	Test performed on Build Dialogic-DMG1000-6.0.SU3.2.001-BE.C.3.1.
Author	spimental

4.1.27 Test Case PC-71

PC-71: TDM forward busy to SIP	
Step(s)	1. Configure a TDM endpoint to forward busy to a SIP endpoint. 2. Make the TDM endpoint busy. 3. Initiate a call from another TDM endpoint to the TDM endpoint that is configured to forward the call. The call should be forwarded to the SIP endpoint. 4. Complete the call by answering the SIP endpoint.
Expected Result(s)	• TDM to SIP endpoints are connected. • Voice test passes. • DTMF test passes. • CPID is correct.
Pass / Fail	Passed
Test Notes	Test performed on Build Dialogic-DMG1000-6.0.SU3.2.001-BE.C.3.1.
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