digium™ | Asterisk™

The Open Source PBX

TDM400P

User's Manual
June 28, 2005

Release 1.0

Safety Certification and Agency Approvals

Safety:
UL, CE
A-Tick (Australia)

Telco:
FCC Part 68 TIA-968-A including amendments A1 & A2
Industry Canada IC-CS-03
Australia S002 (POTS line) / S003 (POTS Extn)

Immunity:
EN55024 ITE (US, CA, AU, EU)

Emissions:
EN55022 Class A Radiated & Conducted (US FCC Part 15, CA, AU, EU)
Class A (Commercial)
Cisper

Federal Communications Commission Part 68

This equipment complies with Part 68 of the FCC rules and the requirements adopted by the ACTA. On the back of the TDM400P printed circuit board is a label that contains, among other information, a product identifier in the format US:AAAEQ##TXXXX. If requested, this number must be provided to the telephone company.

A plug and jack used to connect this equipment to the premises wiring and telephone network must comply with the applicable FCC Part 68 rules and requirements adopted by the ACTA.

The REN is used to determine the number of devices that may be connected to a telephone line. Excessive RENs on a telephone line may result in the devices not ringing in response to an incoming call. In most but not all areas, the sum of RENs should not exceed five (5.0). To be certain of the number of devices that may be connected to a line, as determined by the total RENs, contact the local telephone company. For products approved after July 23, 2001, the REN is part of the product identifier that has the format US:AAAEQ##TXXXX. The digits represented by ## are the REN without a decimal point (e.g., 03 is a REN of 0.3).

If the TDM400P causes harm to the telephone network, the telephone company may notify you in advance that temporary discontinuance of service may be required. But if advance notice is not practical, the telephone company will notify you as soon as possible. Also, you will be advised of your right to file a complaint with the FCC if you believe it is necessary.

The telephone company may make changes in its facilities, equipment, operations or procedures that could affect the operation of the equipment. If this happens, the telephone company will provide advance notice in order for you to make necessary modifications to maintain uninterrupted service.

If you experience problems with the TDM400P, contact Digium, Inc. (877.546.8963) for repair and/or warranty information. If the equipment is causing harm to the telephone network, the telephone company may request that you disconnect the equipment until the problem is resolved.

Connection to party line service is subject to state tariffs. Contact the state public utility commission, public service commission, or corporation commission for information.

Introduction to TDM400P Documentation

This manual contains product information for the TDM400P card. Be sure to refer to any supplementary documents or release notes that were shipped with your equipment. The manual is organized in the following manner:

Chapter/ Appendix	Title	Description
1	Overview	Identifies the card and type of modules you received with the TDM400P. This chapter covers applications and uses of the TDM400P in the real world.
2	Card Installation	Provides instructions for installing the card in your PC, acquiring correct drivers, and checking device compatibility.
3	Configuration	Provides examples for configuring dial plan options.
4	FXS and FXO Explained	Describes the FXO (Foreign Exchange Office) and FXS (Foreign Exchange Station) modules and their significance.
5	Troubleshooting	Explains resolutions to common problems and frequently asked questions pertaining to card installation and usage.
6	Service Personnel Requisites	Provides details for labeling the card and external PC enclosures by qualified service personnel.
A	Pin Assignments	Lists the connectors and pin assignments.
B	Specifications	Details card specifications.

C	Asterisk Configuration Commands	Describes all available configuration commands and their option settings in the zapata.conf file.
D	Dial Plan Applications	Defines configuration features in the extensions.conf file.
E	Glossary and Acronyms	Defines terms related to this product.

Symbol Definitions

Caution statements indicate a condition where damage to the unit or its configuration could occur if operational procedures are not followed. To reduce the risk of damage or injury, follow all steps or procedures as instructed.

The ESD symbol indicates electrostatic sensitive devices. Observe precautions for handling devices. Wear a properly grounded electrostatic discharge (ESD) wrist strap while handling the device.

The Electrical Hazard Symbol indicates a possibility of electrical shock when operating this unit in certain situations. To reduce the risk of damage or injury, follow all steps or procedures as instructed.

Important Safety Instructions

User Cautions

Warning
This card must be used with the PC lid screwed down.
Telecommunications network voltages exist inside the PC!
The PC must be shut down and telecommunications line connection
shall be removed before opening the PC.

Electrical Shock.
To reduce the risk of injury, damage to the unit or your equipment, do
not attempt to touch the modules while they are powered. The case
should be securely closed before power is applied to the unit.

Alarm Dialing Equipment.
If your home has specially wired alarm equipment connected to the
telephone line, ensure the installation of the TDM400P does not
disable your alarm equipment. If you have questions, consult your
telephone company or a qualified installer.

Servicing.
Do not attempt to service this card unless specifically instructed to do
so. Do not attempt to remove the card from your equipment while
power is present. Refer servicing to qualified service personnel.

Water and Moisture.
Do not spill liquids on this unit. Do not operate this equipment in a
wet environment.

Heat.
Do not operate or store this product near heat sources such as
radiators, air ducts, areas subject to direct, intense sunlight, or other
products that produce heat.

Static Electricity.
To reduce the risk of damaging the unit or your equipment, do not
attempt to open the enclosure or gain access to areas where you are
not instructed to do so. Refer servicing to qualified service personnel.

Service Personnel Cautions

Warning.
This card must be used with the PC lid screwed down.
Telecommunications network voltages exist inside the PC!
The PC must be shut down and telecommunications line connection
shall be removed before opening the PC.

Electrical Shock.
To reduce the risk of injury, damage to the unit or your equipment, do
not attempt to touch the modules while they are powered. The case
should be securely closed before power is applied to the unit.

Servicing.
Disconnect telecommunications network cable before opening the
cover or removing the card from the mother equipment

Labeling.
For safety reasons, only connect equipment with a
Telecommunications Compliance label. This includes customer
equipment previously labelled **Permitted** *or* **Certified**

Caution.
Only connect regulatory equipment (approved for use in your specific
country) to the telecommunications network voltage circuit ports.

TABLE OF CONTENTS

Chapter 1
Overview

The TDM400P card is a device used for connecting your phone network to the PSTN (Public Switched Telephone Network) world. This is accomplished through phone lines connected to the FXO (Foreign Exchange Office) ports and phones connected via the FXS (Foreign Exchange Station) ports. The card allows Asterisk PBX software to connect to your phone network, creating a Small Office Home Office (SOHO) telephony environment. See Figure 1 for an example of the card's primary application.
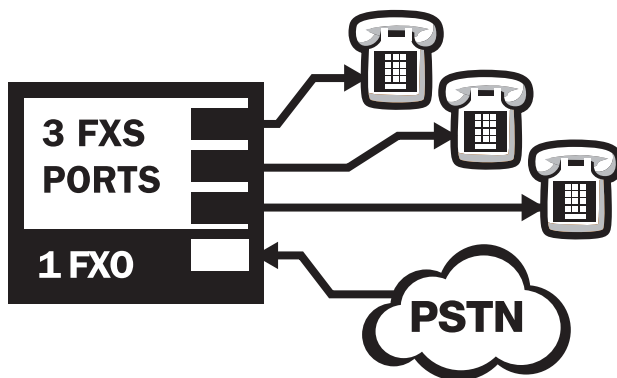
Figure 1: Sample Card Application

What is Asterisk?

Asterisk is the first Open Source PBX. Since it runs on Linux, it inherits all of the power and stability of the operating system. The name Asterisk is derived from the all-inclusive "wildcard" symbol in UNIX, because it is opening a wide range of opportunities for developers worldwide to create solutions which would otherwise be cost-prohibitive.

Asterisk allows you to create a PBX solution that rivals the features and functionality of traditional telephony switches. Current PBX solutions are expensive and proprietary. International companies are discovering that Asterisk is cost effective, low maintenance, and flexible enough to handle all of their voice and data networking. Combined with Digium hardware and a common PC, anyone can replace an existing switch or complement a PBX by adding VoIP, voicemail, conferencing, and many other capabilities. Asterisk will integrate with most standards-based IP telephone handsets and software. Analog phones and ADSI-screen phones are also supported.

Chapter 2
Card Installation

This chapter provides the following information:

Unpacking the Card

When you unpack your card, carefully inspect it for any damage that may have occurred in shipment. If damage is suspected, file a claim with the carrier and contact your reseller where the card was purchased, or Digium Technical Support (877.546.8963 or 1.877.LINUXME!). Keep the original shipping container to use for future shipment or proof of damage during shipment.

Note: Only qualified service personnel should install the card. Users should not attempt to perform this function themselves.

Shipment Inspection

The following items are included in shipment of the TDM400P:

- TDM400P card
- A combination of FXO and/or FXS modules
- Port Identification label
- User's manual

Module Identification

The TDM400P card ships with FXO and/or FXS modules. These are identified by their color. Take a moment to identify which modules were shipped with your card.

- FXO (Foreign Exchange Office) modules are Red
- FXS (Foreign Exchange Station) modules are Green

See Figure 2 on page 18 for an example of the card shown with two of each module.

Port Identification

Each card consists of four RJ11 ports located on the bracket. Each port correlates to a module slot (containing the FXO or FXS modules). The ports are numbered in sequence from one to four. The top port is Port 1 and the bottom port is Port 4. See Figure 2 on page 18 for appropriate identification of these ports.

Note: It is important to identify the type and location of your TDM400P modules. You will need this information during the Asterisk configuration.

There are multiple standard configurations in which the TDM400P card may be purchased. Each configuration consists of one to four FXS and/or FXO modules.

It is easiest to identify your card by understanding the naming scheme for each card. The first digit provides the number of FXS modules present on the card. The second digit provides the number of FXO modules present on the card. See Table 1 on page 19 for a complete list of possible configurations.
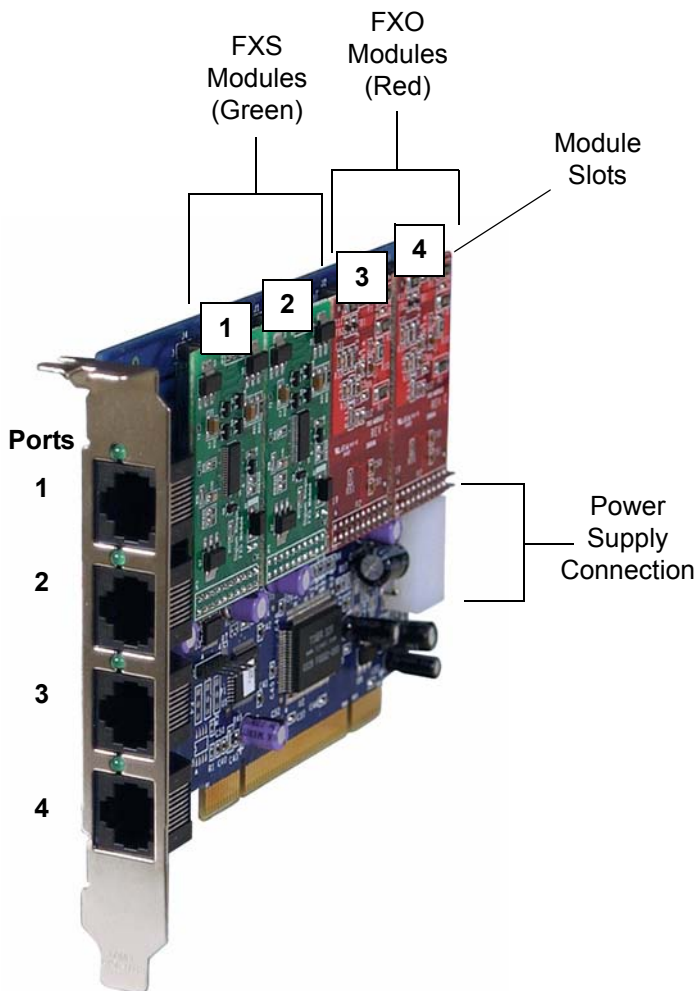
Figure 2: TDM400P Card (Model TDM22)

Table 1: Standard Card Configurations

Card ID	Modules Included in Configuration
TDM10	1 FXS module
TDM20	2 FXS modules
TDM30	3 FXS modules
TDM40	4 FXS modules
TDM01	1 FXO module
TDM02	2 FXO modules
TDM03	3 FXO modules
TDM04	4 FXO modules
TDM11	1 FXS module and 1 FXO module
TDM22	2 FXS modules and 2 FXO modules
TDM12	1 FXS module and 2 FXO modules
TDM13	1 FXS module and 3 FXO modules
TDM21	2 FXS modules and 1 FXO module
TDM31	3 FXS modules and 1 FXO module

There are multiple standard configurations in which the TDM400P card may be purchased. Each configuration consists of one to four FXS and/or FXO modules.

It is easiest to identify your card by understanding the naming scheme for each card. The first digit provides the number of FXS modules present on the card. The second digit provides the number of FXO modules present on the card.

> **Caution.**
> *Only qualified service personnel should continue with hardware installation and configuration of the TDM400P card. Users should not attempt to perform these functions themselves.*

Hardware Installation

1. Now that you are acquainted with your card, power down your computer and unplug it from its power source.
2. Attach a static strap to your wrist and open the case.
3. Remove the bracket place holder and insert the card into a 3.3 or 5 volt PCI slot (PCI 2.2 or greater). See Figure 3.
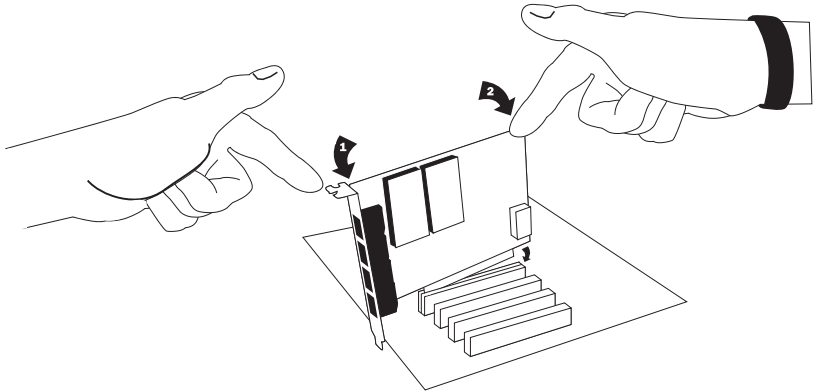
Figure 3: Insert the Card

4. If your card has any FXS modules, you will also need to connect the power cable from your computer's power supply to the back of the card. Insert a four-pin 12 volt connector (disk drive power supply cable, e.g. hard drive) into the white plastic connector on the rear of the card. See Figure 4.
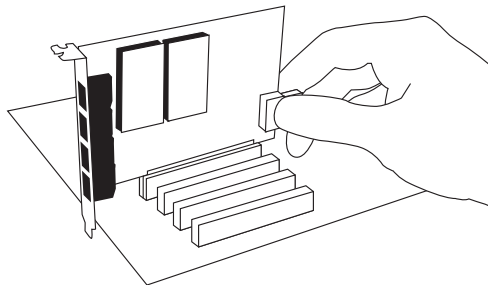
Figure 4: Connect Power for FXS Modules

5. Replace the cover to your computer.

Electrical Shock.
To reduce the risk of injury, damage to the unit or your equipment, do not attempt to apply power to the unit while the case is open.Personal injury or damage to the unit could occur if the modules are touched while powered is applied.

6. Plug all outside phone lines to the FXO (red) ports and connect all phones to the FXS (green) ports as needed.

Caution.
This unit must be connected to the Telecommunications Network in your country using an approved line cord, eg: for Australia use only line cords complying with ACA Technical Standard TS008.

Caution.
Only connect regulatory equipment (approved for use in your specific country) to the telecommunications network voltage circuit ports.

Software Installation

The card is only supported under Linux. Digium, Inc. recommends Debian, Fedora, and Red Hat, however, all other distributions are supported by Digium Technical Support. Digium hardware requires drivers and libraries that have not yet been integrated with the Linux kernel. You can obtain the source code from ftp.digium.com or cvs.digium.com.

To install your TDM400P card, you will need:

- Full Linux kernel 2.4 (or later) source code
- Bison 1.84 (or higher)
- zlib, libedit, and openssl development libraries and headers

1. Check the PCI bus compatibility. Boot the computer into Linux. After the machine is loaded, log in and execute the following:

```
# lspci
```

You will see a screen similar to the one shown in Figure 5 on page 24. Confirm your PCI bus compatibility by scanning for the information highlighted by the call out in the figure.

Figure 5: Checking PCI Compatibility Screen

A Tiger Jet or Jens Shoenfeld device should be identified. If you do not see either of these, your machine is not PCI 2.2 compatible (or higher) and the card will not work.

2. Check out the latest driver code and Asterisk software from
ftp.digium.com or cvs.digium.com. The following example is
provided to check out the code from the CVS repository:

```
# cd /usr/src
# export CVSROOT=:pserver:anoncvs@cvs.
  digium.com:/usr/cvsroot
# cvs login
# cvs checkout zaptel libpri asterisk
```

> **Note:** The password is **anoncvs**.

This will create the following directories: zaptel, libpri, and asterisk.
Complete instructions for installing Asterisk is available at
www.asterisk.org.

3. Install the zaptel drivers. Compiling the directories is straightforward.
Change to each directory and type **make install**. Compile in this
order: zaptel, libpri, and asterisk.
Type the following:

```
# cd /usr/src/zaptel
# make clean
# make install
```

> **Note:** If the build fails, it may be because you are missing one of
> the build dependencies, the kernel source, or development tools.
> Feel free to contact your reseller where the card was purchased, or

call Digium Technical Support (877.546.8963 or 1.877.LINUXME!) for assistance.

4. Load Zaptel drivers into the kernel using the program **modprobe**. The appropriate driver for the TDM400P card is **wctdm**. Users in all countries except Australia should use the following modprobe command:

```
# modprobe wctdm
```

5. Configure the card for the driver. Open the **zaptel.conf** file from the **etc** directory:

```
# nano /etc/zaptel.conf
```

6. If your card has any red FXO modules, add the following:

```
fxsks
```

This uses kewl start signalling which is loop start with disconnect supervision. For example, a TDM04B card would be configured as the following:

```
fxsks=1,2,3,4
```

> **Note:** You should have identified the type of TDM400P card when you received it. If you are not sure, refer to **Module Identification** on page 16 for assistance.

7. If your card has any green FXS modules, add the following:

```
fxoks
```

This uses kewl start signalling which is loop start with disconnect supervision. For example, a TDM40B card would be configured as the following:

```
fxoks=1,2,3,4
```

An example TDM22B card configuration would be:

```
fxoks=1,2
fxsks=3,4
```

8. Set your loadzone and default zone for your country. If you are not in the United States, you will want to change the default configuration from **US** to your own two letter country abbreviation. Save the file and return to the command line.

```
#modprobe wctdm
#ztcfg -vv
```

9. Confirm the card configuration by initiating a dmesg command. The example in Figure 6 on page 28 shows a sample of the **dmesg** screen output for a TDM04B card. The text shown may vary slightly depending the type of card installed.

```
Found a Wildcard TDM: Wildcard TDM400P REV E/F (4 modules)
Freshmaker version: 71
Freshmaker passed register test
Module 0: Installed -- AUTO FXO (FCC mode)
Module 1: Installed -- AUTO FXO (FCC mode)
Module 2: Installed -- AUTO FXO (FCC mode)
Module 3: Installed -- AUTO FXO (FCC mode)
Found a Wildcard TDM: Wildcard TDM400P REV E/F (4 modules)
Registered tone zone 0 (United States / North America)
Registered tone zone 0 (United States / North America)
ide-floppy driver 0.99.newide
```

Figure 6: dmesg Screen Shot

10. Install libpri and Asterisk.

```
#cd /usr/src/libpri
#make install
#cd /usr/src/asterisk
#make
#make install
#make samples
```

> **Note:** This step installs sample configuration files which are necessary for Asterisk. You will make changes to these configuration files to set up your dial plan and customize Asterisk to suit your particular needs.

Chapter 3
Configuration

This chapter provides sample configurations to demonstrate customizing the Asterisk software to meet your individual needs. Each section explains basic options as examples. Once you have familiarized yourself with the samples, you can edit the configuration files to meet your specific needs.

All the available options for configuring the **zapata.conf** file are explained in Appendix C—"Asterisk Configuration Commands," page 51.

> **Note:** Only qualified service personnel should install the card. Users should not attempt to perform this function themselves.

General Options

Open the **zapata.conf** file from the **/etc/asterisk/** directory.

The following is a sample configuration for a TDM22B card. You can place this at the bottom of your **zapata.conf** file.

```
;;General options
usecallerid=yes
hidecallerid=no
callwaiting=yes
threewaycalling=yes
transfer=yes
echocancel=yes
```

```
echocancelwhenbridged=yes
rxgain=0.0
txgain=0.0
;;FXS Modules
Group=1
signalling=fxo_ks
context=Internal
channel=1-2

;;FXO Modules
Group=2
echocancel=yes
signalling=fxs_ks
context=Incoming
channel=3-4
```

Voicemail

Open **voicemail.conf** and find the following line at the bottom:

```
[default]
1234 => 4242, Mark Spencer, root@localhost
```

In this example, **1234** is the mailbox number, **4242** is the password, **Mark Spencer** is the person's name, and **root@localhost** is his email address.

You can add extensions by adding the following:
```
1000 => 1234, Moose Member, moose@digium.com
2000 => 1234, John Bigelow, jbigelow@digium.com
```

Dial Plan

Open **extensions.conf**, which contains a large, complex sample dial plan. In this step, you will configure a basic dial plan to enable you to send and receive calls. Go to the bottom of the file and add the following lines:

```
[Internal]
exten => 1000,1,Dial(zap/1,20,rt)
exten => 1000,2,Voicemail(u1000)
exten => 1000,102,Voicemail(b1000)

exten => 2000,1,Dial(zap/2,20,rt)
exten => 2000,2,Voicemail(u2000)
exten => 2000,102,Voicemail(b2000)

exten => 8500,1,VoiceMailMain
exten => 8501,1,MusicOnHold

exten => _9.,1,Dial(zap/g2/www${EXTEN:1})
exten => _9.,2,Congestion

[Incoming]
exten => s,1,Answer
exten => s,2,Dial(zap/g1,20,rt)
exten => s,3,Voicemail(u1000)
exten => s,103,Voicemail(b1000)
```

In this example there are two internal extensions (1000 and 2000), a number to check voicemail (8500), a number to listen to musiconhold, (8501), and a prefix to dial to get an outside line (9). It is configured for

incoming calls over the FXO rings phones 1 and 2, and voicemail is routed to mailbox 1000.

Testing Your Configuration

1. Start Asterisk by typing:

```
asterisk
```

2. Connect to Asterisk and view the output by typing:

```
asterisk -vvvvr
```

3. Dial tone should be present on phones connected to the FXS ports. Test your configuration by placing an outgoing call, placing a call from extension 1 to 2, or receiving an incoming call. Successful completion of these tasks indicates your configuration is working properly.
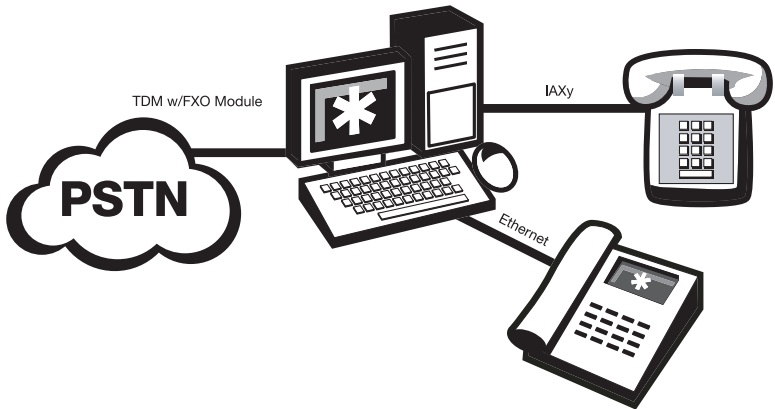
Figure 7: Sample Application

Note: More detailed information is provided on troubleshooting in the Asterisk User's Manual. A copy may be requested by contacting Digium Technical Support (877.546.8963 or 1.877.LINUXME!) or visiting the website at www.digium.com. You may also contact your distributor or reseller where the card was purchased for assistance.

Chapter 4
FXS and FXO Explained

Identification

There are multiple standard configurations in which the TDM400P card may be purchased. Each configuration consists of one to four FXS and/or FXO modules. These modules are identified by their color.

- **FXS** -Foreign Exchange Station (Green Modules for TDM400P)
- **FXO** -Foreign Exchange Office (Red Modules for TDM400P)

This chapter provides an in-depth review of the two module types and their uses within your Asterisk server.

   Note: Only qualified service personnel should install the card. Users should not attempt to perform this function themselves.

FXS Module

The FXS module allows the TDM400P card to initiate and send ringing voltage to an FXO device.

FXO Module

The FXO module allows the TDM400P card to terminate analog telephone lines (POTS).

Because of the modular design, you can activate additional ports at any time with more FXS or FXO daughter cards. The FXO module passes all the call features any standard analog telephone line will support. The

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phone receiving the call is the last FXO device in the chain. When it receives voltage from an FXS device, the phone rings.

Using Your TDM400P Card

Connect the outside line to an FXO port on your Asterisk server to receive voltage from the outside lines.

Connect the phones to FXS ports on your Asterisk server. When the FXO module in your Asterisk Server receives the voltage, it will then generate voltage using the FXS module and send it to your analog phone.

Chapter 5
Troubleshooting

This chapter provides frequently asked questions as identified from Digium Technical Support and possible resolutions. Multiple resources are available to obtain more information about Asterisk and Digium products. These resources are listed on page 41.

The FXO module never seems to hang-up the line. How do I set it to hang-up?

Set **busydetect=yes** and **busycount=10** in the **zapata.conf** for your channels. This will cause the line to hang-up by listening for the busy tone. Upon editing **zapata.conf**, you will need to restart Asterisk.

My card is acting strange. There is static on the line, poor sound quality, and random hang-ups. How can I solve this?

There are three possible causes for these problems.

1. Check for shared interrupts by entering the following:

```
cat /proc/interrupts
```
or
```
lspci -vb
```
If a conflict exists, try moving the card to another PCI slot.

2. Check to see if X windows is running by entering the following:
```
ps aux|grep X
```
If it is, stop X windows since it may cause a conflict with Asterisk.

3. Check to see if your hard drives are running with dma levels set. Perform an **hdparm** on your hard drive interface.

If you are still having problems contact your reseller where the card was purchased, or Digium Technical Support (877.546.8963 or 1.877.LINUXME!).

There is a slight echo. How can I adjust the sound quality?

There are several options available to correct this. Each involves editing the **zapata.conf** file. Be sure to restart Asterisk upon completion.

1. Adjust **echocancel=yes** to one of the following values: **32, 64, 128, or 256**.

2. You can also set **echotraining=yes**.

3. You can also adjust the **rxgain** and the **txgain**, although it is only recommended to shift between **-5** and **5**.

How can I enable more features?

To view all of the options available to add to your dial plan, type the following command from within Asterisk:

```
show applications
```

Digium also offers services to help configure and add features you might need. Contact Digium Technical Support (877.546.8963 or 1.877.LINUXME!) for more information.

Where can I ask even more questions?

There are several places to inquire for more information about Asterisk Digium products:

1. Digium Technical Support (877.546.8963 or 1.877.LINUXME!) is available 8am-6pm Central Time, Monday - Friday.
2. Asterisk forums (forums.digium.com).
3. Asterisk users mailing list (asterisk.org).
4. IRC channel **#asterisk** on (irc.freenode.net).

Chapter 6
Service Personnel Requisites

This chapter explains required installation measures that must be followed by qualified service personnel. Additional information is provided in the preceding chapters for identifying, installing, and configuring the TDM400P card. This chapter contains the following sections:

- **Card Labeling** on page 44
- **Labeling the External PC** on page 45

⚠ **Caution.**
Only qualified service personnel should install the card. Users should not attempt to perform this function themselves.

⚠ **Caution.**
Only connect regulatory equipment (approved for use in your specific country) to the telecommunications network voltage circuit ports.

⚠ **Warning.**
Disconnect telecommunications network cables before opening the cover or removing the card from the mother equipment.

⚠ **Warning.**
For safety reasons, only connect equipment with a telecommunications compliance label. This includes customer equipment previously labelled **permitted** *or* **certified**.

Card Labeling

A blank label sticker is provided with the card. This label should be completed in indelible ink with the appropriate port information (as described below) and placed on the TDM400P card. There are four ports available on the card which need to be identified on the external label. Refer to Chapter 2—"Card Installation" for instructions on identifying the type of card you have and the modules.

For FXS modules, identify the corresponding port with **PHONE**. For FXO modules, identify the corresponding port with **LINE**. If there are not four modules present on the card, identify the remaining empty ports with **VOID**. Table 2 provides an example for a model TDM21 card (consisting of 2 FXS modules and 1 FXO module). Similarly, Figure 8 on page 45 shows an actual card with the label installed in the lower, outside portion of the bracket.

Table 2: Card Label

1	PHONE
2	PHONE
3	LINE
4	VOID

Figure 8: Actual Label on Card

Labeling the External PC

On the external PC enclosure, two labels need to be applied by the installer in indelible and clear markings. These labels can be created by the distributor or installer, or can be obtained upon request from Digium Technical Support (877.546.8963 or 1.877.LINUXME!).

Apply the label shown in Figure 9 on the outside of the PC enclosure, near the screws that secure the casing to the PC. Do not install on the bottom of the PC.

WARNING
SHUT DOWN THE PC
AND DISCONNECT FROM TELEPHONE LINE
BEFORE OPENING THE COVER.

CAUTION
ONLY CONNECT APPROVED EQUIPMENT
TO THE TELECOMMUNICATIONS NETWORK
VOLTAGE CIRCUIT PORTS.

Figure 9: External Caution Label

Appendix A
Pin Assignments

All four ports on the TDM400P bracket are 6-pin RJ11 ports. The pin assignments are identified in Table A-1.

Table A-1: RJ11 Telco Port Connector

	Pin	Description
Pin 1 — Pin 6	1	Not used
	2	Not used
	3	Tip
	4	Ring
	5	Not used
	6	Not used

Appendix B
Specifications

This appendix provides specifications, required environmental conditions, and maximum power consumption for the TDM400P card.

Physical.

Size:	5.242" × 4.02" × 0.59" (13.3 x 10.2 x 1.5 cm)
	PCB size, does not include the PCI bracket
Weight:	1.06 oz (30 gm) with no modules loaded
	Each module adds 0.058 oz (1.6 gm)

Interfaces.

Local Loop Access: RJ11

PCI Bus: 3.3V or 5V bus slot, half-length slot minimum size, 33MHz minimum bus speed, compliant with PCI 2.2 or greater.

Additional Power: four-pin 12V connector for FXS power supply (required only if FXS modules are installed)

Environment.

Temperature: 0 to 50° C (32 to 122° F) operation
                    -20 to 65° C (4 to 149° F) storage

Humidity: 10 to 90% non-condensing

Hardware and Software Requirements.
500-Mhz Pentium III or better
64MB RAM
Available PCI Slot (as described previously)

Table B-1: Maximum Power Consumption

Model	Power
3.3V All models	360mW
5V All models	1W
12V TDM04	0W
TDM11 into 3REN	2.4W
TDM22 into 3REN	3.6W
TDM40 into 3REN	5W

Note: 3.3 and 5 volt power is taken from the PCI slot. 12 volt power is taken only from the four-pin hard disk drive connector.

Appendix C
Asterisk Configuration Commands

This appendix provides the configuration options available in the zapata.conf file which is created during the initial installation of Asterisk. To change any of the following features, edit the zapata.conf file with your text editor, saving the file upon completion. The following lists the command, description, and the available options:

adsi
ADSI (Analog Display Services Interface) can be enabled on a per-channel basis if you have (or may have) ADSI compatible CPE equipment.
yes / no

busycount
If enabled, it is also possible to specify how many busy tones to wait before hanging up.
positive integer

busydetect
On trunk interfaces (FXS) and E&M interfaces (E&M, Wink, Feature Group D, etc.), it can be useful to perform busy detection either in an effort to detect hangup or for detecting busy signals.
yes / no

cadence

You can define your own custom ring cadences, up to 8 pairs. If the silence is negative, it indicates where the callerid spill is to be placed. Also, if you define any custom cadences, the default cadences will be turned off.

Syntax is: cadence=ring,silence [,ring,silence[...]]

The default is based on the country code.

callerid

CallerID can be set to **asreceived** *or a specific number if you want to override it. Note that* **asreceived** *only applies to trunk interfaces.*

asreceived or a specific number

callgroup, pickupgroup

Supports ring groups (a.k.a. call groups) and pickup groups. If a phone is ringing and it is a member of a group which is one of your pickup groups, then you can answer it by picking up and dialing *8#.

For simple offices, just make these both the same positive integer

callprogress, progzone

On trunk interfaces (FXS), it can be useful to attempt to follow the progress of a call through ringing, busy, and answering. If enabled, call progress attempts to determine answer, busy, and ringing on phone lines. This feature is HIGHLY EXPERIMENTAL and can easily detect false answers, so DO NOT rely on it being accurate.

Few zones are supported at the time of this writing, but may be selected with progzone, yes / no, and two letter country code for progzone.

callreturn

Support call return (*69).

yes / no

callwaiting
Enable call waiting on FXO lines.
yes / no

callwaitingcallerid
Support caller ID on call waiting.
yes / no

canpark
Allow call parking.
yes / no

cidsignalling
Type of caller ID signalling in use.
bell =bell202 (US)
v23 = v23 (UK)
dtmf = DTMF (Denmark, Sweden, & Netherlands)

cidstart
Identifies what signals the start of caller ID.
ring = a ring signals the start
polarity = polarity reversal signals the start

echocancel
Enable echo cancellation.
yes / no, or a power of two from 32 to 256 (if you want to actually set the
number of cancellation taps)

echocancelwhenbridged
Generally, it is not necessary (and in fact undesirable) to echo cancel when the circuit path is entirely TDM. You may, however, reverse this behavior by enabling the echo cancel during pure TDM bridging.
yes / no

echotraining
In some cases, the echo canceller doesn't train quickly enough and there is echo at the beginning of the call. Enabling echo training will cause Asterisk to briefly mute the channel, send an impulse, and use the impulse response to pre-train the echo canceller so it can start out with a much closer idea of the actual echo.
Value may be yes, no, or a number of milliseconds (positive int) to delay before training.
(default = 400)

faxdetect
Upon fax detection, routes fax to a fax extension.
Both, incoming, outgoing, or no.

group
Logical groups can be assigned to allow outgoing rollover.
Groups range from 0 to 63, and multiple groups can be specified.

hanguponpolarityswitch
In some countries, a polarity reversal is used to signal the disconnect of a phone line. If this option is enabled, the call will be considered disconnected on a polarity reversal.
yes / no

hidecallerid

Hide outgoing caller ID. Override with *67 or *82.
yes / no

immediate

Specify whether the channel should be answered immediately or if the simple switch should provide dialtone, read digits, etc.
yes / no

jitterbuffers

Configure jitter buffers in zapata.
(each one is 20ms, default is 4) positive integer

mailbox

Supports stutter dialtone.
If a mailbox is specified without a voicemail context, then when message is received, taking the phone off hook will cause a stutter dialtone instead of a normal dialtone. If a mailbox is specified with a voicemail context, the same will result if a message is received in the specified voicemail context.
mailbox ex. 1234, or 1234@other.

musiconhold

Select which class of music to use for music on hold. If not specified then the default will be used. The music class is defined in musiconhold.conf file.
default, loud, random

pulsedial

Use pulse dial instead of DTMF for FXO (FXS signalled) devices.
yes / no

relaxdtmf
If you have trouble with DTMF detection, you can relax the DTMF detection parameters. Relaxing them may make the DTMF detector more likely to have talk off where DTMF is detected.
yes / no

rxgain
Receive gain.
Positive or negative double, measured in dB

sendcalleridafter
Some countries have ring tones with a set of cadences which differ from the default (e.g. UK uses ring-ring). This requires the callerid to be set with a delay, and not right after the first ring (the default).
Positive integer

threewaycalling
Support three-way calling.
yes / no

transfer
Support flash-hook call transfer (requires three way calling). Also enables call parking (overrides the **canpark** *parameter).*
yes / no

txgain
Transmit gain.
Positive or negative double, measured in dB

usecallerid
Whether or not to use caller ID.

yes / no

usedistinctiveringdetection
Indicates whether or not to allow distinctive ring detection on FXO lines.
yes / no

Appendix D
Dial Plan Applications

This appendix details the applications available for configuring the dial plan. The dial plan is customized in the extensions.conf file which is created during the initial installation of Asterisk. This file controls how all calls are handled and routed. To change any of the following options, edit the extensions.conf file, saving the file upon completion. The following defines each feature:

AgentCallbackLogin
Call agent callback login

AgentLogin
Call agent login

AGI
Executes an AGI compliant application

Answer
Answer a channel if ringing

Authenticate
Authenticate a user

BackGround
Play a file while awaiting extension

BackgroundDetect
Background a file with talk detect

Busy
Indicate busy condition and stop

Dial
Place a call and connect to the current channel

Directory
Provide directory of voicemail extensions

DISA
DISA (Direct Inward System Access)

Goto
Go to a particular priority, extension, or context

GotoIf
Conditional goto

GotoIfTime
Conditional goto on current time

Hangup
Unconditional hangup

Macro
Macro Implementation

Monitor
Monitor a channel

MusicOnHold
Play music on hold indefinitely

Playback
Play a file

Queue
Queue a call for a call queue

Record
Record to a file

SetCallerID
Set CallerID

SetCallerPres
Set CallerID presentation

SetCIDName
Set CallerID name

SetCIDNum
Set CallerID number

VoiceMail
Leave a voicemail message

VoiceMailMain
Enter voicemail system

Zapateller
Block telemarketers with SIT

Appendix E
Glossary and Acronyms

ANSI *American National Standards Institute*

An organization which proposes and establishes standards for international communications.

asynchronous

Not synchronized; not timed to an outside clock source. Transmission is controlled by start bits at the beginning and stop bits at the end of each character. Asynchronous communications are often found in internet access and remote office applications.

attenuation

The dissipation of a transmitted signal's power as it travels over a wire.

bandwidth

The capacity to carry traffic. Higher bandwidth indicates the ability to transfer more data in a given time period.

bit

The smallest element of information in a digital system. A bit can be either a zero or a one.

bps *bits per second*

A measurement of transmission speed across a data connection.

broadband

Broadband transmission shares the bandwidth of a particular medium (copper or fiber optic) to integrate multiple signals. The channels take up different frequencies on the cable, integrating voice, data, and video over one line.

channel

A generic term for an individual data stream. Service providers can use multiplexing techniques to transmit multiple channels over a common medium.

Cat5

Category of Performance for wiring and cabling. Cat 5 cabling support applications up to 100 MHz.

Cat5E

Category of Performance for wiring and cabling. Category 5 Enhanced wiring supports signal rates up to 100 MHz but adheres to stricter quality specifications.

CLEC   *competitive local exchange carrier*

A term for telephone companies established after the Telecommunications Act of 1996 deregulated the LECs. CLECs compete with ILECs to offer local service. See also *LEC* and *ILEC*.

CO *central office*

The CO houses local switching equipment. All local access lines in a particular geographic area terminate at this facility (which is usually owned and operated by an ILEC).

CPE *customer premises equipment*

Terminal equipment which is connected to the telecommunications network and which resides within the home or office of the customer. This includes telephones, modems, terminals, routers, and television set-top boxes.

DS0 *Digital Signal, Level 0*

A voice grade channel of 64 Kbps. The worldwide standard speed for digitizing voice conversation using PCM (Pulse Code Modulation).

DS1 *Digital Signal, Level 1*

1.544 Mbps in North America (T1) and Japan (J1) -up to 24 voice channels (DS0s), 2.048 Mbps in Europe (E1) - up to 32 voice channels (DS0s). DS1/T1/E1 lines are part of the PSTN.

DS3 *Digital Signal, Level 3*

T3 in North America and Japan, E3 in Europe. Up to 672 voice channels (DS0s). DS3/T3/E3 lines are not part of the PSTN

DTMF *Dual Tone Multi-Frequency*

Push-button or touch tone dialing.

E1

The European equivalent of North American T1, transmits data at 2.048 Mbps, up to 32 voice channels (DS0s).

E3

The European equivalent of North American T3, transmits data at 34.368 Mbps, up to 512 voice channels (DS0s). Equivalent to 16 E1 lines.

EMI   *Electromagnetic Interference*

Unwanted electrical noise present on a power line

full duplex

Data transmission in two directions simultaneously.

FXO   *Foreign Exchange Office*

Receives the ringing voltage from an FXS device. Outside lines are connected to the FXO port on your TDM400P card.

FXS   *Foreign Exchange Station*

Initiates and sends ringing voltage. Phones are connected to the FXS ports on the TDM400P card.

G.711

The International Telecommunications Union recommendation for an algorithm designed to transmit and receive maelaw PCM voice and A-law at digital bit rate 64 Kbps. This algorithm is used for digital telephone sets on digital PBX.

G.729

The International Telecommunications Union standard for voice algorithm.

H.323

The International Telecommunications Union standard for multimedia communications over packet-based networks.

IAX   *Inter-Asterisk eXchange*

The protocol used by Asterisk. It is used to enable VoIP connections between Asterisk servers, and between servers and clients that also use the IAX protocol.

iLBC   *internet Low Bitrate Codec*

A free speech codec used for voice over IP. It is designed for narrow band speech with a payload bitrate of 13.33 kbps (frame length = 30ms) and 15.2 kbps (frame length = 20 ms).

ILEC   *incumbent local exchange carrier*

The LECs that were the original carriers in the market prior to the entry of competition and therefore have the dominant position in the market.

interface

A point of contact between two systems, networks, or devices.

ISO   *International Standards Organization*

LED   *light-emitting diode*

Linux

A robust, feature-packed open source operating system based on Unix that remains freely available on the internet. It boasts dependability and offers a wide range of compatibility with hardware and software. Asterisk runs exclusively on Linux.

loopback

A state in which the transmit signal is reversed back as the receive signal, typically by a far-end network element.

MGCP   *Media Gateway Control Protocol*

multiplexing

Transmitting multiple signals over a single line or channel. FDM (frequency division multiplexing) and TDM (time division multiplexing) are the two most common methods. FDM separates signals by dividing the data onto different carrier frequencies, and TDM separates signals by interleaving bits one after the other.

MUX   *multiplexer*

A device which transmits multiple signals over a single communications line or channel. See multiplexing.

PBX   *private branch exchange*

A smaller version of a phone company's large central switching office. Example: Asterisk.

PCI   *peripheral component interconnect*

A standard bus used in most computers to connect peripheral devices.

POP   *point of presence*

The physical connection point between a network and a telephone network. A POP is usually a network node serving as the equivalent of a CO to a network service provider or an interexchange carrier.

POTS   *plain old telephone service*

Standard phone service over the public switched telephone network (PSTN). This service provides analog bandwidth of less than 4 kHz.

PPP   *point-to-point protocol*

Type of communications link that connects a single device to another single device, such as a remote terminal to a host computer.

PSTN   *public switched telephone network*

A communications network which uses telephones to establish connections between two points. Also referred to as the dial network.

QoS   *quality of service*

A measure of telephone service, as specified by the Public Service Commission.

RJ11

A six pin jack typically used for connecting telephones, modems, and fax machines in residential and business settings to PBX or the local telephone CO.

SIP   *Session Initiation Protocol*

An IETF proposed standard for setting up sessions between one or more clients. It is currently the leading signaling protocol for Voice over IP, gradually replacing H.323.

T1

A dedicated digital carrier facility which transmits up to 24 voice channels (DS0s) and transmits data at 1.544 Mbps. Commonly used to carry traffic to and from private business networks and ISPs.

T3

A dedicated digital carrier facility which consists of 28 T1 lines and transmits data at 44.736 Mbps. Equivalent to 672 voice channels (DS0s).

TDM   *time division multiplexer*

A device that supports simultaneous transmission of multiple data streams into a single high-speed data stream. TDM separates signals by interleaving bits one after the other.

telco

A generic name which refers to the telephone companies throughout the world, including RBOCs, LECs, and PTTs.

tip and ring

The standard termination on the two conductors of a telephone circuit; named after the physical appearance of the contact areas on the jack plug.

twisted pair

Two copper wires commonly used for telephony and data communications. The wires are wrapped loosely around each other to minimize radio frequency interference or interference from other pairs in the same bundle.

V   *volts*

VoIP   *Voice over IP*

Technology used for transmitting voice traffic over a data network using the Internet Protocol.

Zap

Digium hardware interface.