TE120 Series
TE120P/TE121/TE122

User Manual

Release 2.2

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Digium, Inc. has made every effort to ensure that the instructions contained in this document are adequate and error free. The manufacturer will, if necessary, explain issues which may not be covered by this documentation. The manufacturer's liability for any errors in the documents is limited to the correction of errors and the aforementioned advisory services.

This document has been prepared for use by professional and properly trained personnel, and the customer assumes full responsibility when using it.

Safety Certification and Agency Approvals

Safety:

UL 60950-1:2003, First Edition
CSA C22.2 No. 60950-1-03 1st Ed. April 1, 2003
IEC 60950-1:2001 First Edition
EN 60950

> **Note:** Canada, Finland, Norway, Sweden and the United States of America require that equipment using this product must be located in a Restricted Access Location (RAL).

Telecom:

FCC Part 68, ANSI/ITA-968-A, Including Amendment A1 and A2
Industry Canada CS-03
AS/ACIF S016: 2001
AS/ACIF S038: 2001
TBR4 November 1995 as amended by TBR4/A1 December 1997
TBR12 December 1993
TBR13 January 1996

EMC:

EN 55022:1998 Class B and 47 CFR Part 15, Subpart B Class B, Radiated and Conducted EN 55024:1998 / IEC 61000

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 your TE120 Series 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.

If your TE120 Series card 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.

The following information may be required when applying to the telephone company for service:

Reg. Number	Service Type	SOC	FIC	USOC
US: DIGDENANTE120P	1.544 Mbps – SF 1.544 Mbps - SF and B8ZS 1.544 Mbps – ESF 1.544 Mbps – ESF and B8ZS	6.0N	04DU9-BN 04DU9-DN 04DU9-1KN 04DU9-1SN	RJ-48C

If you experience problems with the TE120 Series, contact Digium, Inc. Technical Support +1.256.428.6161 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.

FCC Part 15

This device complies with part 15 of FCC rules. Operation is subject to the following two conditions: (1) This device may not cause harmful interference, and (2) This device must accept any interference received, including interference that may cause undesired operation.

Industry Canada Compliance Information

The Industry Canada label applied to the product (identified by the Industry Canada logo or the "IC:" in front of the certification/registration number) indicates that the Industry Canada technical specifications were met.

Introduction to TE120 Series Documentation

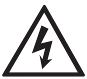
This manual contains product information for the TE120 Series of cards (TE120P, TE121, and TE122). Be sure to refer to any supplementary documents or release notes that were shipped with your equipment. The manual is organized in the following manner:

Chapter/ Appendix	Title	Description
1	Overview	Identifies your card's features. This chapter also covers applications and uses for the TE120 Series card 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 instructions for configuring your card.
4	Troubleshooting	Explains resolutions to common problems and frequently asked questions pertaining to card installation and usage.
A	Pin Assignments	Lists the connectors and pin assignments.
B	Specifications	Details card specifications.
C	Glossary and Acronyms	A list of terms and acronyms used throughout this manual.

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

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.

Save these instructions for future reference.

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.

Service Personnel Cautions

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

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.

Caution.
This card is not intended for home use. It must be used in restricted access locations and installed in UL Listed I.T.E. only.

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Chapter 1
Overview

The Digium TE120 Series cards are T1/E1 capable cards that can handle both voice and data. It supports industry standard protocols, including Robbed Bit Signaling also known as CAS (Channel Associated Signaling) and CCS (Common Channel Signaling), E&M (Digital Emulation), Primary Rate ISDN (PRI), and several data modes (PPP, HDLC, Cisco HDLC and frame relay). It is capable of running in E1, T1, or J1 modes.

Designed to be fully compatible with existing software applications and integrate fully with the Asterisk platform, the TE120 Series cards allow many advanced call features.

Data Modes:

- Cisco HDLC
- HDLC
- PPP
- Multilink PPP
- Frame Relay

Voice Modes:

- PRI CPE and PRI NET
  - NI1
  - NI2
  - EuroISDN
  - 4ESS (AT&T)
  - 5ESS (Lucent)
  - DMS100

- E&M
  - Wink
  - Feature Group B
  - Feature Group D

- FXO and FXS
  - Ground Start
  - Loop Start
  - Loop Start with Disconnect Detect

The TE120 Series cards can be used to connect your Asterisk machine to the PSTN world, your channel bank, or even another PBX. This is accomplished via a T1/E1 interface. The cards allow Asterisk software to connect to your network, creating a professional telephony environment. Figure 1 and Figure 2 show examples of the card's application.

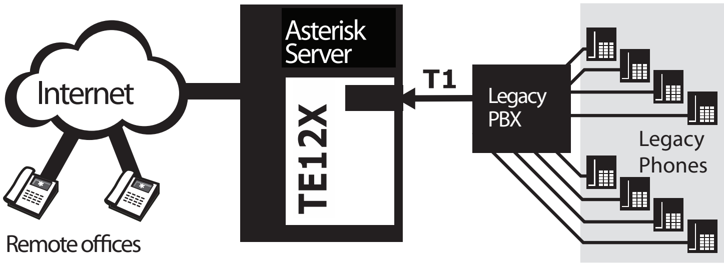


Figure 1: Sample Legacy Phone Application

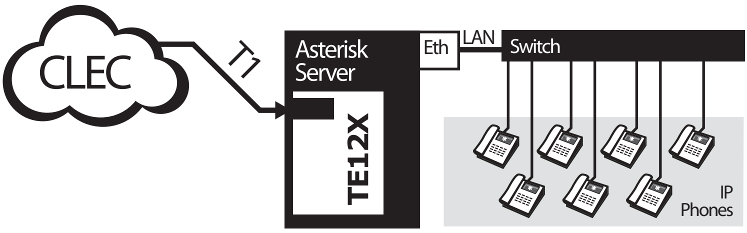


Figure 2: Sample IP Phone Application

Echo-Cancellation

Users connecting their TE120 series cards to the PSTN or other devices are likely to be placing calls that will result, at some point, in an unbalanced 4-wire/2-wire hybrid. The result of this hybrid is the reflection of a near-end echo to the calling party. Elimination of this echo is the responsibility of echo cancellation.

The TE120 series cards, unless otherwise equipped, utilize Asterisk to perform software-based echo cancellation. Asterisk maintains a number of open source echo cancelers. These open source echo cancelers provide a moderate level of echo cancellation, but are not capable of dealing with higher levels of, or more advanced, echoes.

Digium recommends that those users concerned about echo cancellation purchase the VPMADT032 hardware echo cancellation module. The VPMADT032 may be combined with both the TE121 and TE122 cards; it may not be combined with the TE120P card. The TE121 and TE122 are offered bundled with the VPMADT032 as, respectively: TE121B, TE122B.

The VPMADT032 is designed to handle up to 128ms of echo cancellation across all channels and provides a G.168 compliant and AT&T Labs certified Toll-Quality echo cancellation solution.

If equipped and not explicitly disabled in zapata.conf, the VPMADT032 will automatically operate and cancel all network echo within its tail range (1024 taps). Users of TE120P cards, which do not maintain the capability to support the VPMADT032, may purchase Digium's commercial HPEC software:

http://www.digium.com/en/products/software/hpec.php

What is Asterisk®?

Asterisk is the world's leading open source telephony engine and tool kit. Offering flexibility unheard of in the world of proprietary communications, Asterisk empowers developers and integrators to create advanced communication solutions...for free. Asterisk® is released as open source under the GNU General Public License (GPL), and it is available for download free of charge. Asterisk® is the most popular open source software available, with the Asterisk Community being the top influencer in VoIP.

Asterisk as a Switch (PBX)

Asterisk can be configured as the core of an IP or hybrid PBX, switching calls, managing routes, enabling features, and connecting callers with the outside world over IP, analog (POTS), and digital (T1/E1) connections.

Asterisk runs on a wide variety of operating systems including Linux, Mac OS X, OpenBSD, FreeBSD and Sun Solaris and provides all of the features you would expect from a PBX including many advanced features that are often associated with high end (and high cost) proprietary PBXs. Asterisk's architecture is designed for maximum flexibility and supports Voice over IP in many protocols, and can interoperate with almost all standards-based telephony equipment using relatively inexpensive hardware.

Asterisk as a Gateway

It can also be built out as the heart of a media gateway, bridging the legacy PSTN to the expanding world of IP telephony. Asterisk's modular architecture allows it to convert between a wide range of communications protocols and media codecs.

Asterisk as a Feature/Media Server

Need an IVR? Asterisk's got you covered. How about a conference bridge? Yep. It's in there. What about an automated attendant? Asterisk does that too. How about a replacement for your aging legacy voicemail system? Can do. Unified messaging? No problem. Need a telephony interface for your web site? Ok.

Asterisk in the Call Center

Asterisk has been adopted by call centers around the world based on its flexibility. Call center and contact center developers have built complete ACD systems based on Asterisk. Asterisk has also added new life to existing call center solutions by adding remote IP agent capabilities, advanced skills-based routing, predictive and bulk dialing, and more.

Asterisk in the Network

Internet Telephony Service Providers (ITSPs), competitive local exchange carriers (CLECS) and even first-tier incumbents have discovered the power of open source communications with Asterisk. Feature servers, hosted services clusters, voicemail systems, pre-paid calling solutions, all based on Asterisk have helped reduce costs and enabled flexibility.

Asterisk Everywhere

Asterisk has become the basis for thousands of communications solutions. If you need to communicate, Asterisk is your answer. For more information on Asterisk visit http://www.asterisk.org or http://www.digium.com.

Chapter 2
Card Installation

This chapter provides the following information:

Note: The TE120 Series card installation instructions are written so that they will apply to any card in the series. Examples and card specific information are included as needed.

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 from which the card was purchased, or Digium Technical Support at 1.256.428.6161. 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. The installer must ensure that the equipment is permanently connected equipment, pluggable type B or connected to a socket-outlet that has been checked to ensure that it is reliably earthed in accordance with the National Electrical Code.

This card is intended for installation in a Restricted Access Location (RAL) only.

Shipment Inspection

The following items are included in shipment of the TE120 Series:

- A TE120P, TE121, or TE122 card.

Identifying Features

Your TE120 Series card has one RJ45 port and two status LEDs. The port is used for connecting T1, E1, or J1 cables. The two LEDs serve as a status LED and an amber loop-back LED. The card includes a strap for selecting either T1 or E1 line mode. See Figure 3 on page 23 to locate these features.

The TE121 and TE122 cards may also be combined with Digium's hardware-based echo canceler, model VPMADT032. See Figure 4 on page 24 for an example of the TE121 card shown with the echo cancellation module.

T1/E1 Selection

The T1/E1 mode, in most cases, is set at the distributor before shipment. You may want to check the setting to be certain it is set for your specific use. With the jumper **off,** the card is ready for T1 mode and with the jumper **on**, the card is ready for E1 mode.

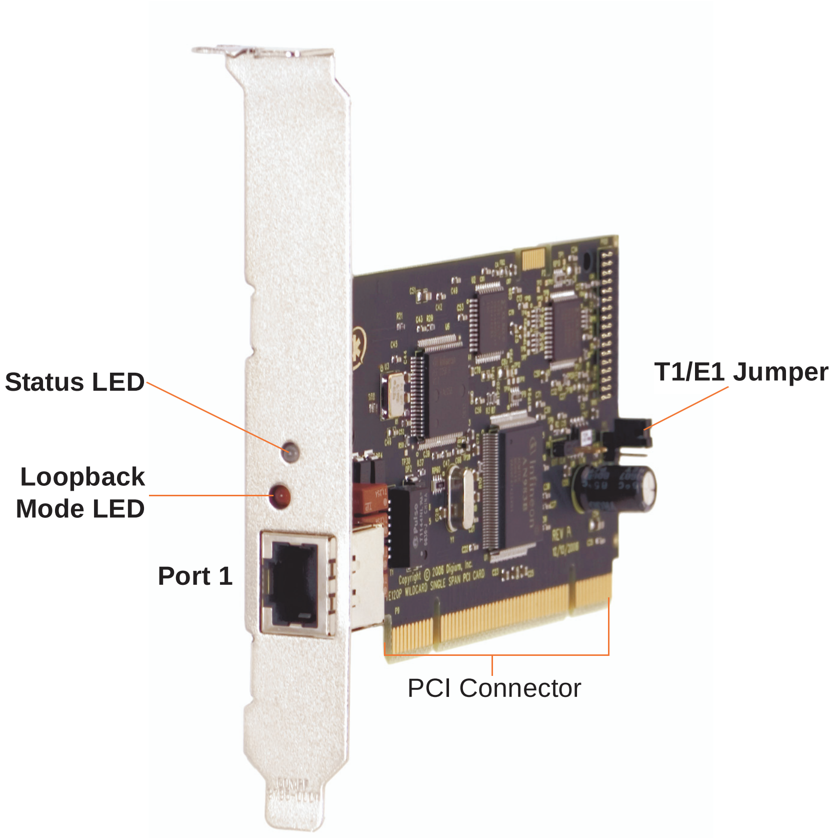


Figure 3: TE120P Card

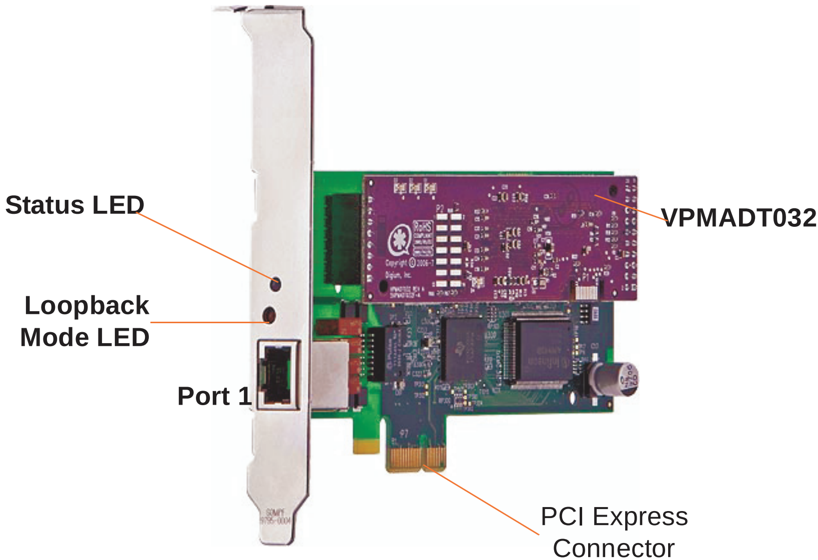


Figure 4: TE121 Card with Echo Cancellation Module

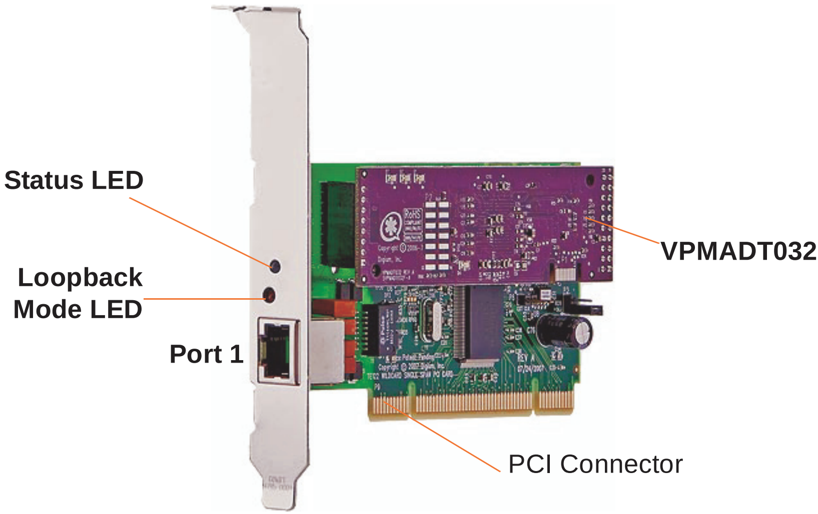


Figure 5: TE122 Card with Echo Cancellation Module

Caution.
Only qualified service personnel should continue with hardware installation and configuration of a TE120 Series card. Non-qualified personnel should not attempt to perform these functions.

Slot Compatibility

Check the type of card you received to be sure it is compatible with your PCI slot. To determine which slot you have, identify it by comparing it to those shown in Figure 6 on page 26.

Slot Number:

0: AGP Pro Slot

1: 64-bit 5.0 volt PCI Slot

2: 64-bit 3.3 volt PCI Slot

3: 32-bit 5.0 volt PCI Slot

4: PCI Express Slot

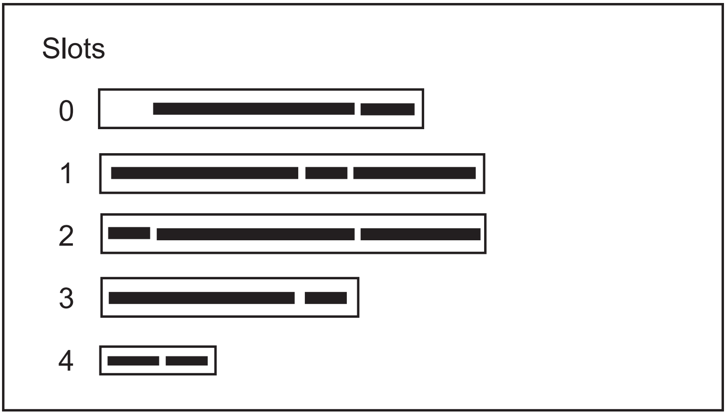


Figure 6: Motherboard PCI Slots

The TE120 Series and TE122 cards are 32-bit 33MHz cards keyed for universal 3.3 volt or 5.0 volt operation and works in any PCI 2.2 (or greater) compliant slot. This means that in the motherboard shown in Figure 6, the TE120 Series and TE122 cards will fit into Slots 1, 2, or 3 (PCI slots) but **will not** fit into Slot 0 (AGP slot).

The TE121 card is a PCI Express card. Slot 4, illustrated above, is a 1 lane (X1) PCI Express compliant slot. The TE121 will work in any PCI Express compliant slot, including lane lengths X1, X4, X8, and X16. This means that in the motherboard shown in Figure 6, the TE121 will only fit into Slot 4. The TE121 **can not** be used in Slots 0 through 3.

Hardware Installation

1. Now that you are acquainted with the TE120 Series cards, power down your computer and unplug it from its power source.
2. Attach a static strap to your wrist and open the case.
3. Check the jumper setting to ensure it matches your equipment configuration. Setting the jumper with the strap **on** enables the ports for E1. Setting the jumper with the strap **off** enables the ports for T1.
4. Remove the bracket place holder and insert the card into a PCI or PCI Express slot. See Figure 7.

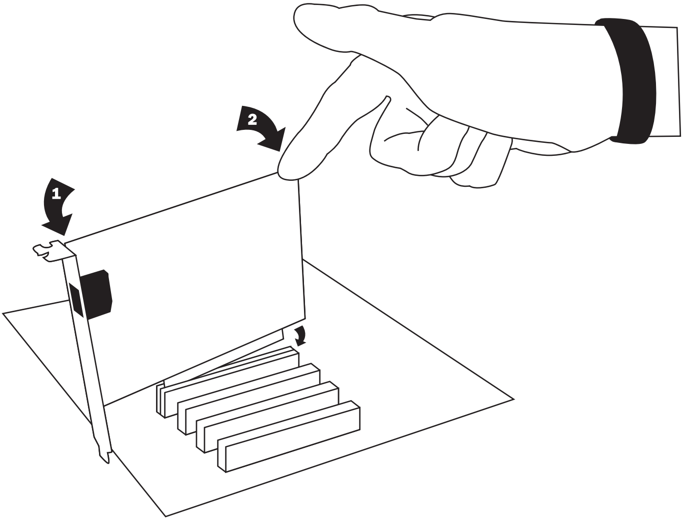


Figure 7: Insert the Card

5. Replace the cover to your computer.
6. Plug the T1 or E1 equipment cable into the RJ45 port.

Caution.
This unit must be connected to the Telecommunications Network in your country using an approved line cord.

Caution.
This unit must be connected only to the appropriate Telecommunications Network port (as approved for use in your specific country).

Software Installation

The TE120 Series cards are only supported on a Linux system. Digium, Inc. recommends Debian, Fedora, and Red Hat. Digium hardware requires drivers and libraries that are integrated with the Linux kernel. You can obtain the source code from downloads.digium.com. Detailed instructions are provided in this section.

To install software for your TE120 Series card, you will need:

- Full Linux kernel 2.6 (or later) source code
- Development libraries and headers for libncurses (only necessary for Asterisk 1.2; or for Zaptel 1.4 and Asterisk 1.4).
- Development libraries and headers for zlib and openssl.
- If you are using the 1.2.x series of Asterisk and Zaptel, you will need Asterisk 1.2.26 or newer, and Zaptel 1.2.23 or newer. If you are using the 1.4.x series of Asterisk and Zaptel, you will need Asterisk 1.4.17 or newer and Zaptel 1.4.8 or newer.

1. Check your **lspci** PCI device listing. Boot the computer into Linux. After the machine has loaded, log in and execute the following:

```
# lspci -n | grep d161
```

Confirm your **lspci** PCI device listing by scanning for the following information in the output screen:

```
0000:01:00.0 0200: d161:<card identifier>
```

In the device listing shown above, <card identifier> will be populated with one of the identifiers listed in the table below.

Table 1: Card Identifiers

Model	Identifier
TE120P	0120
TE121	8000
TE122	8001

A Digium TE120 Series (TE120P/TE121/TE122) ISDN Controller should be identified. If a controller is not identified, then your machine is not PCI 2.2 (or higher) or PCI Express compatible and the card will not work with your equipment. Please contact Digium’s technical support for assistance.

2. Download the latest branch of libpri that matches the branch of Zaptel and Asterisk which you are using. If you are using the 1.2.x branch, then download the 1.2.x branch of libpri. Likewise if you are using the 1.4.x branch, then obtain that version of libpri. Libpri is available from http://downloads.digium.com/pub/telephony/libpri.
3. Expand the downloaded tarballs. Substitute the version of libpri you are using with the X.X in the command lines below.:

```
# tar -zxvf libpri-1.X.X.tar.gz
# cd libpri-1.X.X/
# make
# make install
```

4. Download the latest Zaptel drivers (1.2.23 or later). If you are using the 1.4 branch of Zaptel, you should use 1.4.8 or later. They are accessible via http from http://downloads.digium.com/pub/telephony/zaptel/.
5. Expand the downloaded tarball and install the drivers. Substitute the version of Zaptel you are using with the XX in the command lines below.

```
#tar -zxvf zaptel-1.X.X.tar.gz
#cd zaptel-1.X.X
#make clean
#./configure (applies to 1.4.X only)
#make menuselect (applies to 1.4.X only if you wish
to customize the install)
#make
#make install
```

Note: If you don't already have configuration files installed, you can type `make samples` to install the default sample configuration files.

Installing Asterisk

If you wish to use Asterisk with your new hardware, you can follow the instructions below. If you are using the 1.2.x series of Asterisk and Zaptel, you will need Asterisk 1.2.26 or newer, and Zaptel 1.2.23 or newer. If you are using the 1.4.x series of Asterisk and Zaptel, you will need Asterisk 1.4.17 or newer and Zaptel 1.4.8 or newer.

1. Download the latest released version of Asterisk, either 1.2.26 (or later), or 1.4.17 (or later). Asterisk can be downloaded via http from http://downloads.digium.com/pub/telephony/asterisk.
2. Expand the downloaded tarballs. Substitute the version of Asterisk you are using with the X.X in the command lines below.

```
# tar -zxvf asterisk-1.X.X.tar.gz
# cd asterisk-1.X.X/
# make clean
# ./configure (applies to 1.4.X only)
# make menuselect (appies to 1.4.X only if you wish
to customize the install)
# make
# make install
```

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 at 1.256.428.6161 for assistance.

Note: Complete instructions for installing Asterisk are available at www.asterisk.org.

Chapter 3
Configuration

The TE120 Series cards have a variety of configuration options. This chapter provides configurations for PRI, channel bank, E&M wink, and finally, data mode. These sample configurations are provided to assist you in familiarizing yourself with the flexibility of editing the configuration files to meet your specific needs. The list of possible configurations is too expansive to cover in this user manual.

Configuring Card Features

Configure Zapata.conf, which is the layer between zaptel and Asterisk, to configure the essential card features.

Switchtype:

```
national:   National ISDN 2 (default)
dms100:     Nortel DMS100
4ess:       AT&T 4ESS
5ess:       Lucent 5ESS
euroisdn:   EuroISDN
ni1:        Old National ISDN 1
```

Echocancel:

Echo Cancellation is enabled in zapata.conf by preceding the channel variable with a variable called echocancel and its length in taps (# of milliseconds multiplied by 8); for example:

```
echocancel=yes

channel => 1-23
```

By default, and when setting to "yes," echo cancellation is enabled and set to 16 ms (128 taps). Echo cancellation is explicitly disabled by setting:

```
echocancel=no
```

Digium does not recommend that users set echo cancellation to "no."

Users of open source Asterisk-based echo cancelers also have the following options:

```
echocancel=128 (this sets 128 taps or 16ms)
```

or

```
echocancel=256 (this sets 256 taps or 32ms)
```

Users of Digium's HPEC software have the following additional options:

`echocancel=512 (this sets 512 taps or 64ms)`

or

`echocancel=1024 (this sets 1024 taps or 128ms)`

Please note that HPEC consumes extremely high amounts of CPU MIPS that increase as the number of taps are increased. Audio quality issues may result from choosing a taps length greater than the server's ability to process the echo in real-time. If audio quality is affected, reduce the taps length or purchase a TE121 or TE122 and Digium's VPMADT032.

Users of Digium's VPMADT032 hardware echo cancellation module will have 128ms of echo cancellation performed at all times unless explicitly disabled by setting the echocancel variable equal to "no."

Signalling:

```
pri_cpe for CPE side.
pri_net for NET side.
```

If you have a T1 PRI, add these lines to the following lines of the sample file.

```
signalling=pri_cpe
switchtype=national
group=1
context=incoming
channel=>1-23
```

E1 PRI

```
signalling=pri_cpe
switchtype=euroisdn
context=incoming
channel=>1-15,17-31
```

You can also configure a channel bank of phones

```
signalling=fxo_ks
group=1
context=phones
channel=>1-24
```

E1 channel bank

```
signalling=fxo_ks
group=1
context=phones
channel=>1-24
```

Note: More detailed troubleshooting information is provided on http://www.asterisk.org.

Configuring T1/E1 Lines

1. Begin by opening the **/etc/zaptel.conf**. This is where the base configuration for your hardware is stored. If you did a **make samples** during the install, you can read through the commented example and edit it to your needs. Otherwise, continue following these instructions.
2. Next, configure your T1/E spans in the span definitions. They are in the following format:

```
span=<span num>,<timing source>,<line build out
(LBO)>,<framing>,<coding>[,yellow]
```

<span num>

Since this card only has one span, the **`<span num>`** will be 1 if it is the only Digium digital interface card in your system.

<timing source>

All T1/E1 spans generate a clock signal on their transmit side. The <timing source> parameter determines whether the clock signal from the far end of the T1/E1 is used as the master source of clock timing. If it is, our own clock will synchronise to it. T1/E1's connected directly or indirectly to a PSTN provider (telco) should generally be the first choice to sync to. The PSTN will never be a slave to you. You must be a slave to it.

Choose **1** to make the equipment at the far end of the E1/T1 link the preferred source of the master clock. Choose **2** to make it the second choice for the master clock, if the first choice port fails (the far end dies, a cable breaks, etc.). Choose **3** to make a port the third choice, and so on. If you have, for instance, 2 ports connected to the PSTN, mark those as **1** and **2**. The number used for each port should be different.

If you choose 0, the port will never be used as a source of timing. This is appropriate when you know the far end should always be a slave to you. If the port is connected to a channel bank, for example, you should always be its master. Any number of ports can be marked as 0.

Incorrect timing sync may cause clicks/noise in the audio, poor quality or failed faxes, unreliable modem operation, and dropped calls.

`<line build out>`

The line build-out (or LBO) is an integer, from the following:

0: 0 db (CSU) / 0-133 feet (DSX-1)

1: 133-266 feet (DSX-1)

2: 266-399 feet (DSX-1)

3: 399-533 feet (DSX-1)

4: 533-655 feet (DSX-1)

5: -7.5db (CSU)

6: -15db (CSU)

7: -22.5db (CSU)

<framing>

d4 or **esf** for T1

cas, or **ccs** for E1

<coding>

ami or **b8zs** for T1

ami or **hdb3** for E1

E1 can also have the extra flag CRC4 at the end for CRC4 checking.

[,yellow] (optional)

If the keyword **yellow** follows, yellow alarm is transmitted when Asterisk is not running.

The following is a typical setup for a telco in the US:

```
span=1,1,0,esf,b8zs
```

In Europe:

```
span=1,1,0,ccs,hdb3,crc4
```

3. Next, define the country zone. See the example configuration file for more details.

```
defaultzone=us
loadzone=us
```

4. If you are using Asterisk, you will need to configure it to use your new hardware. This configuration is located in **/etc/asterisk/zapata.conf**. These options are subject to change with future Asterisk versions. Examples are provided below that may work for you.

First Example: Channel Bank

The Channel Bank in this example has 24 FXS ports. In this configuration, the zaptel.conf is set for the card to provide timing to the channel bank and fxoks is set for 24 stations.

Set zapata.conf to mirror the configuration with **signalling=fxo_ks** and define it for channels 1-24.

T1 Channel Bank

```
/etc/zaptel.conf:
span=1,0,0,esf,b8zs
fxoks=1-24

/etc/asterisk/zapata.conf:
group=1
context=channelbank
signalling=fxo_ks
channel=1-24
```

E1 Channel Bank

```
/etc/zaptel.conf:
span=1,0,0,ccs,hdb3
fxoks=1-31

/etc/asterisk/zapata.conf:
group=1
context=channelbank
signalling=fxo_ks
channel=1-31
```

Second Example: E&M Line

To configure a span for E&M, the zaptel.conf must specify the span and the channel definition, while the zapata.conf specifies the signalling and incoming dialplan context for a group of channels. In the example below, the zaptel.conf shows the first span port configured to receive timing, with no line build-out (LBO), using ESF and B8ZS for framing and coding. The zapata.conf shows that group 1 has channels 1-24 configured with featd signalling and processes incoming calls with the "incoming" dialplan extensions context.

There are many other signalling methods available, though featd is very common. See the zapata.conf sample configuration file for commented examples.

```
/etc/zaptel.conf:
span=1,1,0,esf,b8zs
e&m=1-24

/etc/asterisk/zapata.conf:
group=1
context=incoming
signalling=featd
channel=1-24
```

Third Example: PRI

By setting the card to take timing in **zaptel.conf,** you acquire 23 b channels and voice channels, with channel 24 as the data transport. For Asterisk, define PRI_CPE so that it is the client side. Define the switch type you are connecting to as **national**. There are several options for the switch type including 5ESS, 4ESS, and NI1. You will then have 23 voice channels for Asterisk.

PRI T1

```
/etc/zaptel.conf:
span=1,1,0,esf,b8zs
bchan=1-23
dchan=24

/etc/asterisk/zapata.conf
group=1
signalling=pri_cpe
switchtype=national
context=incoming
channel=1-23
```

PRI E1

```
/etc/zaptel.conf:
span=1,1,0,ccs,hdb3
bchan=1-15,17-31
dchan=16

/etc/asterisk/zapata.conf
group=1
signalling=pri_cpe
switchtype=euroisdn
context=incoming
channel=1-15,17-31
```

Fourth Example: Data Mode

Data mode is a little different than the other options. The zaptel.conf is configured as follows:

```
/etc/zaptel.conf
span=1,0,0,esf,b8zs
nethdlc=1-24
```

1. Uncomment the following line in zconfig.h of the Zaptel package:

   ```
   #define CONFIG_ZAPATA_NET
   ```

   If you are using a Linux kernel prior to 2.4.19, also uncomment this line:

   ```
   #define CONFIG_OLD_HDLC_API
   ```

Build the data tools for Zaptel by executing:

```
make data; make sethdlc-new
```

Or, for kernels prior to 2.4.19

```
make data; make sethdlc
make install
```

2. Load and configure your driver:

```
modprobe wcte12xp
```

```
ztcfg
```

3. Use sethdlc to bring up the interface:

```
sethdlc hdlc0 cisco
```

-or- for old style (make sethdlc instead of sethdlc-new) use:

```
sethdlc hdlc0 mode cisco
```

4. Assign the interface an address:

```
ifconfig hdlc0 192.168.0.1 netmask 255.255.255.0
```

5. The interface may be addressed as any other networking interface (i.e., eth0) in Linux.

Testing Your Configuration.

1. Load Zaptel drivers into the kernel using the program **modprobe**. The appropriate driver for the TE120 Series cards is **wcte12xp**. Use the following modprobe command:

   # modprobe wcte12xp

   # ztcfg -vv

   # dmesg

```
ACPI: PCI interrupt 0000:01:00.0[A] -> GSI 21 (level, low) -> IRQ
209
PCI Config reg is 02900117
wcte120p: New Reg: fe590000!
Detected REG0: 00000100
Detected REG1: 00007849
Detected REG2: 0000001d
(pre) Reg fc is 50000027
Detected REG0: 0000ffff
(post) Reg fc is 50000024
Detected REG2: 0000ffff
wcte120p: reg is a04c0004
TE120P: FALC version: 00000000
TE120P: Setting up global serial parameters for T1 FALC V1.2
TE120P: Successfully initialized serial bus for card
Found a Wildcard TE: Wildcard TE120P
```

Figure 8: Example dmesg Output

2. Run **zttool** from the command line and see if the span turns green for each span you have connected.

   zttool

3. Execute the following Asterisk command to see if the span came up successfully.

```
asterisk
asterisk -vvvr
```

Note: More detailed troubleshooting information is provided on http://www.asterisk.org.

Chapter 4
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 52.

What do the Status LED colors indicate?

- Green - Card is in-sync with the far end.
- Yellow - Card is synchronizing or is receiving a red alarm from the far end. Use a software tool such as zttool to get a textual description of the state of the card.
- Red - Card is not seeing far end, circuit is not up, or cable is bad.

I can't receive DID calls even though I have it enabled in extensions.conf.

Your telco might be sending calls with a method you are not expecting.

1. Check the method being used by attempting the following in your line context:
   ```
   _x.,1,NoOp(My DID Matches as ${EXTEN}
   ```

2. Then type `reload` in the Asterisk console and call in. You should see the DID come in on your T1/E1 line.

My D Channel seems to go up and down.

Check to be sure you have set your timing parameters correctly. Also check the common causes of problems for a T1. See the **Common Fixes for all cards**, page 50.

I have trouble dialing out. It seems that one type of dialing works (local, long distance, international) but another does not.

Check your **pridialplan** variable and be sure that you are dialing using the method your telco is expecting.

I am having trouble receiving access code information over E&M.

Try the other types of E&M (featd, featb, etc.) to match the method your telco is using to stream information.

I am having issues with my PRI. How can I see the messages coming across my D channel?

Enter the following command:

```
PRI debug span X
```

where x is the port from which you are connected. This command will show you the PRI messages coming across your D channel for that message.

I am still having problems and the telco tells me it is my equipment.

The first thing to do in this situation is to test your equipment.

1. Connect a loopback plug. (A loopback plug has pin 1 going to pin 4 and pin 2 going to pin 5.) Insert the plug into the span and wait for its LED to turn green.

2. Stop Asterisk and edit **zaptel.conf** by removing the lines defined for your card and replacing them with the following:
   ```
   span=>1,0,0,esf,b8zs
   clear=1-24
   ```

   Or if you have an E1 span:
   ```
   span=> 1,0,0,ccs,hdb3
   clear=1-31
   ```

3. Navigate to your zaptel source directory and type:
   `make tests`

   Followed by:
   `./patlooptest /dev/zap/1 60`

   The first argument in the patlooptest command is the device for the channel number you want to test. You should always test the first channel of a span. The second argument is the duration in seconds to run the test.

   This runs a pattern looptest for 60 seconds. If you receive any failures, it is possible you have a bad card and will need to call Digium Technical Support at 1.256.428.6161

Common Fixes for all cards

1. Check for shared interrupts by entering the following:
   `cat /proc/interrupts`

   and

   `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 X windows is running, stop the application since it may cause a conflict with Asterisk.

3. Check to see if your IDE hard drives are running with DMA levels set. Advanced users can perform an **hdparm** on your hard drive interface.

Use hdparm with caution as the man page states that hard drive corruption can occur when using incorrect settings. Please review the man page for hdparm and make sure you understand the risks before using this tool.

   Check the current mode using this command:
   `hdparm -vi /dev/[``IDE Device`**`]`**

   Use this command to set the drives into UDMA2 mode:
   `hdparm -d 1 -X udma2 -c 3 /dev/[``IDE Device`**`]`**

   If you are still having problems, contact your reseller from which the card was purchased, or Digium Technical Support at 1.256.428.6161.

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 at 1.256.428.6161 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 at 1.256.428.6161 is available 7am-7pm Central Time (GMT -6), Monday - Friday.
2. Asterisk users mailing list (asterisk.org/lists.digium.com).
3. IRC channel **#asterisk** on (irc.freenode.net).

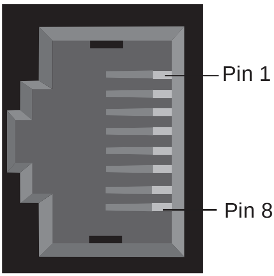
Subscription Services Program

Digium is dedicated to supporting your Asterisk system by offering full technical support through our Subscription Services Program. Through this program, you can be at ease knowing that your business will always have access to the Asterisk experts. Pricing on Subscription Services may be obtained from your nearest reseller or you may call Digium Sales for referral to your nearest reseller at +1.256.428.6000 or e-mail sales@digium.com.

Appendix A
Pin Assignments

The communication port on the TE120 Series card bracket is an 8-pin RJ45 port. The pin assignments are identified in Table A-1.

Table A-1: RJ45 Telco Port Connector

	Pin	Description
	1	Rx
	2	Rx
	3	Not used
	4	Tx
	5	Tx
	6	Not used
	7	Not used
	8	Not used

Appendix B
Specifications

This appendix provides specifications, required environmental conditions, and maximum power consumption for the TE120P card.

Physical.

Size: 4.82” × 2.175” × 0.63” (12.2 x 5.5 x 1.6 cm)
PCB size, does not include the PCI bracket

Weight: 2 oz (57g)

Interfaces.

Local Loop Access: E1, T1, J1, PRI; RJ45

PCI Bus (TE120P and TE122): 3.3V or 5V bus slot, half-length slot minimum size, 33MHz minimum bus speed, compliant with PCI 2.2 or greater.

(TE121) - PCI-E X1, compliant with PCI-E X1 1.0 or greater.

Environment.

Temperature: 0 to 50° C (32 to 122° F) operation
-20 to 70° C (4 to 158° F) storage

Humidity: 10 to 90% non-condensing

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

Table B-2: Maximum Power Consumption

Model	Power
TE120P 3.3V 5V	 1.5 Watts 0.1Watt
TE121 3.3V	 2.0 Watts
TE121B 3.3V	 3.0 Watts
TE122 3.3V 5V	 1.5 Watts 0.1Watt
TE122B 3.3V 5V	 2.5 Watts 0.1Watt

Appendix C
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.

G.711

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

G.729

An International Telecommunications Union standard for voice algorithm.

H.323

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

IAX *Inter-Asterisk eXchange*

A VoIP 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 is supported 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*

The public switched telephone network (PSTN) is the network of the world's public circuit-switched telephone networks. Originally a network of fixed-line analog telephone systems, the PSTN is now almost entirely digital, and now includes mobile as well as fixed telephones.

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 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.

Zaptel (Zap)

Zapata Telephony Project dedicated to implementing a reasonable and affordable Computer Telephony platform into the world marketplace.