TE120 Series
TE120P/TE121/TE122

User Manual

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.

VoIPon www.voipon.co.uk sales@voipon.co.uk Tel: +44 (0)1245 808195 Fax: +44 (0)1245 808299

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 a T1/E1 capable card series created for voice and data. They support industry standard protocols, including Robbed Bit Signaling (also known as CAS or Channel Associated Signaling), CCS (Common Channel Signaling), E&M, Primary Rate ISDN (PRI), and several data modes (PPP, HDLC, Cisco HDLC, and frame relay). The TE120 Series cards are 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
    - Q.SIG

- 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.
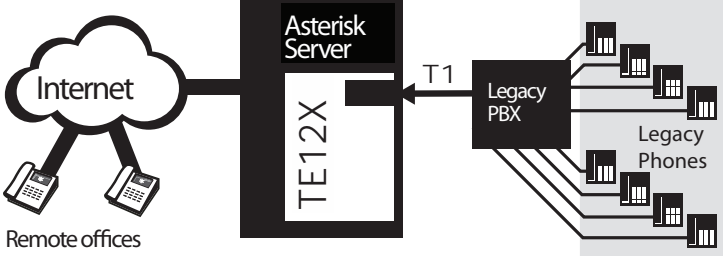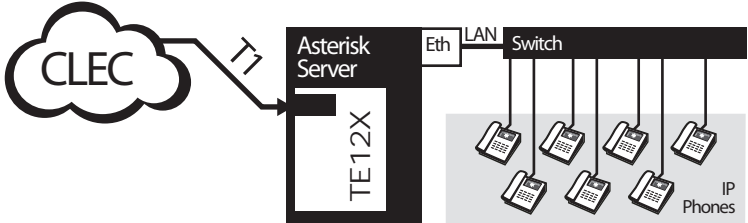
Figure 1: Sample Legacy Phone Application

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 echo cancellation solution.

If equipped and not explicitly disabled in chan_dahdi.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 also 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. It 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 contact 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:

■ TE120 Series card (TE120P/TE121/TE122)

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 span is ready for T1 mode, and with the jumper **on**, the span is ready for E1 mode.

Status LED

Loopback Mode LED

Port 1

T1/E1 Jumper

PCI Connector

Figure 3: TE120P Card

Status LED

VPMADT032

**Loopback
Mode LED**

Port 1

PCI Express
Connector
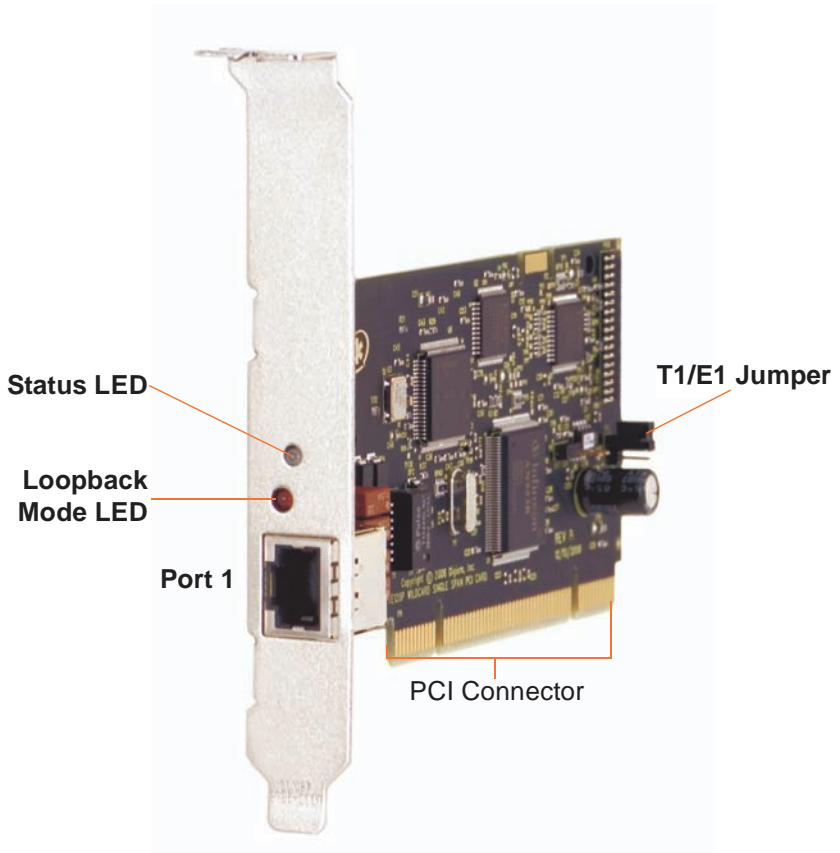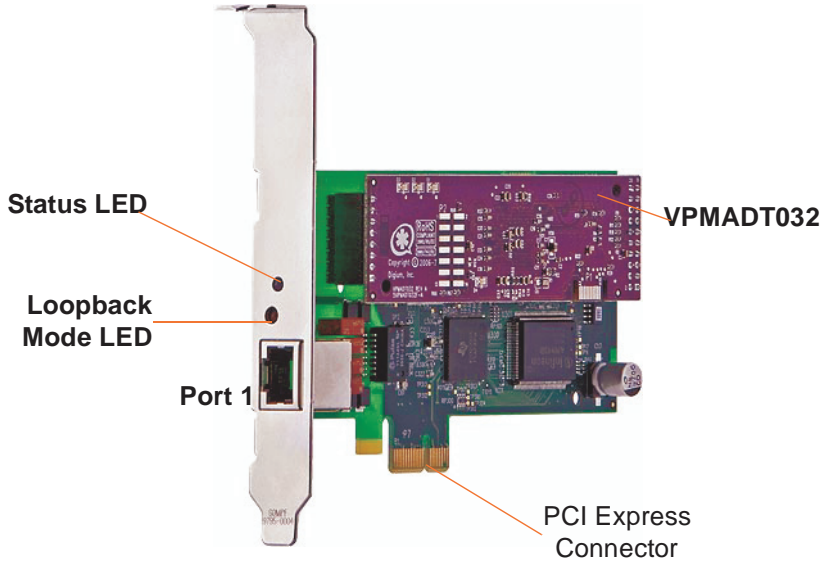
Figure 4: TE121 Card with Echo Cancellation Module
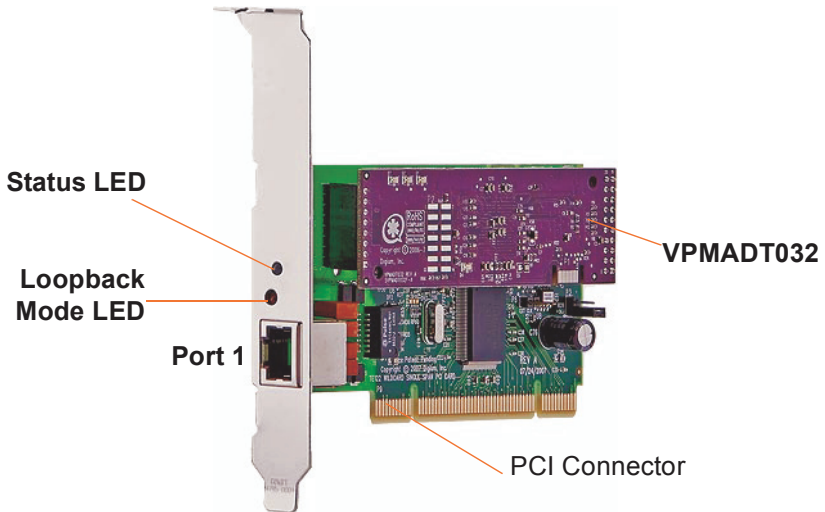
Figure 5: TE122 Card with Echo Cancellation Module

Caution.

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

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 X1 Slot
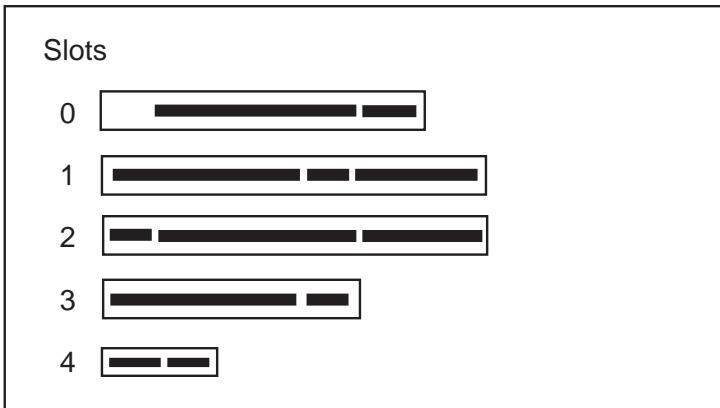
Figure 6: Motherboard Slots

The TE120 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 jumper strap **on** enables the ports for E1. Setting the jumper with the jumper 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 for an example of card installation.
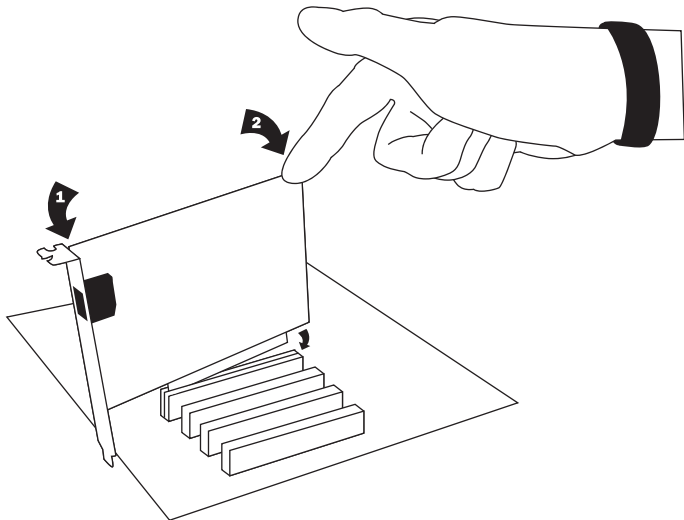
Figure 7: Insert the Card

5. Replace the cover to your computer.

6. Plug the T1 or E1 equipment cable into the RJ45 port.

Note: It is recommended that you use shielded cables.

⚠ **Caution.**
This unit must be connected to the Telecommunications Network in your country using an approved line cord (e.g. for Australia use only line cords complying with ACA Technical Standard TS008).

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

Software Installation

Digium hardware requires drivers and libraries that are not integrated with the Linux kernel. Digium hardware is only supported under Linux. Digium recommends CentOS, Debian, Red Hat, and Ubuntu distributions of Linux. However, many other distributions are supported by Digium Technical Support.

Digium's software, including drivers and application software, may be obtained from Digium's download server at:
http://downloads.digium.com

For an introduction to Asterisk, Digium's telephony software, including additional information on its configuration, setup, and features, please refer to:
http://www.asterisk.org

For the latest information on setting up and configuring DAHDI drivers for your Digium hardware product, please refer to the latest release of this manual which is available from the product-specific documentation

section at:
http://www.digium.com

To install your TE120 Series card, you will need:

- Linux 2.6 kernel headers
- Development libraries and headers for ncurses
- Development libraries and headers for zlib and openssl
- Development libraries and headers for newt
- GCC and standard software build tools

It is recommended that you use the most recent version of the Asterisk, DAHDI, and libpri software for the best results. If you have previously installed any of these, Digium recommends that you upgrade to the latest "-current" version of each.

If you are using the 1.4.x series of Asterisk, you will need Asterisk 1.4.22 or newer.

1. After the machine has booted to Linux, log in and execute the following command to list the devices detected by the PCI bus:

```
# lspci -n
```

2. Confirm that the output from **lspci** lists a device with Digium's PCI vendor ID which is "d161". The screen output should be similar to the following:

```
0000:01:00.0 0200: ISDN controller: Unknown device
d161:<card identifier>
```

Note: The output from **lspci** may or may not state "Unknown device". If it does, this does not indicate a problem.

In the PCI 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) card identifier should be listed. If a matching card identifier is not listed, then your machine is not PCI 2.2 (or higher) or PCI Express compatible, and the card will not work with your motherboard.

3. Download the latest version of libpri. Substitute the version of libpri for the X.X in the command line below. libpri is available for download from:
http://downloads.digium.com/pub/telephony/libpri

```
# wget http://downloads.digium.com/pub/telephony/
libpri/libpri-X.X.current.tar.gz
```

4. Expand the downloaded file, compile its contents, and install the libraries. Substitute the version of libpri for the X.X and X.X.X in the

command lines below.

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

Note: There is no coorelation between the versioning of libpri and Asterisk. The libpri 1.4 branch will function with the Asterisk 1.6 branch.

5. Download the latest DAHDI drivers with tools. DAHDI is available for download from:
http://downloads.digium.com/pub/telephony/dahdi-linux-complete

```
# wget http://downloads.digium.com/pub/telephony/
dahdi-linux-complete/dahdi-linux-complete-
current.tar.gz
```

6. Expand the downloaded file, compile its contents, and install the drivers and tools. Substitute the version of DAHDI for the X.X.X in the command lines below.

```
# tar -zxvf dahdi-linux-complete-current.tar.gz
# cd dahdi-linux-complete-X.X.X+X.X.X
# make
# make install
# make config
```

Note: Executing 'make config' will install an init script and symlinks which will allow you to start and stop DAHDI as a service.

Installing Asterisk

If you wish to use Asterisk with your new hardware, you can follow the instructions below.

1. Download the latest release version of Asterisk, either 1.4.22 (or later) or 1.6.0.1 (or later). Substitute the version of Asterisk for the X.X in the command below. Asterisk is available for download from: http://downloads.digium.com/pub/telephony/asterisk

```
# wget http://downloads.digium.com/pub/telephony/
asterisk/asterisk-X.X-current.tar.gz
```

2. Expand the downloaded file, compile its contents, and install the application. Substitute the version of Asterisk for the the X.X and X.X.X in the command lines below.

```
# tar -zxvf asterisk-X.X-current.tar.gz
# cd asterisk-X.X.X/
# ./configure
# make menuselect
# make
# make install
```

3. If this is the first Asterisk installation on this system, you should install the sample configuration files. To do this, run:

```
# make samples
```

Note: Running this command will overwrite, after making a backup copy, any older Asterisk configuration files that you have in the /etc/ asterisk directory.

If your installation has failed, it may be because you are missing one or more of the build dependencies, the kernel headers, or the development tools. Please contact your reseller where the card was purchased, or call Digium Technical Support at 1.256.428.6161 for assistance.

4. 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

You will need to modify the **chan_dahdi.conf** file which is located in the /etc/asterisk directory in order to configure the essential features of your card. This file is the configuration layer between DAHDI and Asterisk.

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

Echocancel:

Echo Cancellation is enabled in chan_dahdi.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 with 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 T1 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 => 1-15,17-31
```

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

Configuring T1/E1 Lines

1. Begin by opening the **system.conf** file from the **/etc/dahdi** directory.

2. Specify the two letter country code for your loadzone and defaultzone. This will preload tone zone data and specify a default tone zone for your interfaces.

The following is a typical setup for a telco in the US:
```
loadzone = us
defaultzone = us
```

3. Configure the SPAN Map.

You will need to define a span. The SPAN map includes defining the SPAN number, timing, line build out, framing, and coding. Configuration details for each of these items is explained in this section.

```
span => <Number>,<Timing>,<Line Build
Out>,<Framing>,<Coding>[,Yellow]
```

Number:

This is the port the T1/E1 is plugged into. The TE120 Series cards only have one port. If you have a single TE120 Series card installed in the system, you should specify 1 as the span number.

Timing:

This determines how timing is handled by the card.

0 - Card provides its own timing
1 - Receives timing from remote end
2 - Receives secondary backup timing from remote end
3 - Receives tertiary backup timing from remote end
4 - Receives quaternary backup timing from remote end

Only one span can be defined to take timing, and it defines timing for the rest of the card's spans.

Line Build Out:

For most setups the line build out is 0.

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:

T1 utilizes framing set for D4 (SF) or ESF. E1 utilizes CAS or CCS.

Coding:

T1 coding can be AMI or B8ZS. E1 coding can be AMI or HDB3. E1 can also have the extra flag CRC4 at the end for CRC4 checking.

Yellow:

The optional yellow flag can be added at the end for transmitting a yellow alarm when no channels are open.

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
```

4. Specify the channel definitions. The format is:
```
<device> = <channel list>
```

A list of valid devices are specified in the sample system.conf file.

The following is a typical setup for a T1 PRI in the US:
```
bchan = 1-23
dchan = 24
```

5. DAHDI uses modular echo cancellers that are configured per channel. The echo cancellers are compiled and installed as part of the dahdi-linux package. You can specify the echo canceller to be used for each channel. The default behavior is for there to be no echo canceller on

any channel. So, it is very important that you specify one in the system.conf file if you do not have hardware echo cancellers and need echo cancellation. The format is:

echocanceller = <echocanceller name>,<channel(s)>

A list of valid echo cancellers are specified in the sample system.conf file.

The following is a typical setup for a T1 PRI in the US using software-based echo cancellation:

echocanceller = mg2,1-23

First Example: Channel Bank

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

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

```
/etc/dahdi/system.conf:
loadzone = us
defaultzone = us
span = 1,0,0,esf,b8zs
fxoks = 1-24
echocanceller = mg2,1-24


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

Second Example: E&M Line

In the E&M Line configuration, the **system.conf** is set for the card to take timing from the telco on E&M with wink while **chan_dahdi.conf** mirrors the configuration. Feat_D is a type of E&M with wink that accepts DID, but there are many E&M options; E&M_W, E&M, Feat_B, etc.

```
/etc/dahdi/system.conf:
loadzone = us
defaultzone = us
span = 1,1,0,esf,b8zs
e&m = 1-24
echocanceller = mg2,1-24

/etc/asterisk/chan_dahdi.conf:
group = 1
echocancel = yes
context = incoming
signalling = feat_d
channel = 1-24
```

VoIPon www.voipon.co.uk sales@voipon.co.uk Tel: +44 (0)1245 808195 Fax: +44 (0)1245 808299

Third Example: PRI

By configuring the card for a T1 PRI line in **system.conf**, you acquire 23 bearer (B) channels for voice on the first 23 channels, and 1 delta (D) channel for signalling information on the 24th channel. In the **chan_dahdi.conf** file, define **pri_cpe** as the signalling type to act as the client side. Define the switch type you are connecting to as **national**. You will then have 23 voice channels for Asterisk.

PRI T1:
```
/etc/dahdi/system.conf:
loadzone = us
defaultzone = us
span = 1,1,0,esf,b8zs
bchan = 1-23
dchan = 24
echocanceller = mg2,1-23

/etc/asterisk/chan_dahdi.conf:
group = 1
echocancel = yes
signalling = pri_cpe
switchtype = national
context = incoming
channel = 1-23
```

PRI E1:

```
/etc/dahdi/system.conf:
loadzone = es
defaultzone = es
span = 1,1,0,ccs,hdb
bchan = 1-15,17-31
dchan = 16
echocanceller = mg2,1-15,17-31

/etc/asterisk/chan_dahdi.conf:
group = 1
echocancel = yes
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 **system.conf** is configured as follows:

```
loadzone = us
defaultzone = us
span = 1,0,0,esf,b8zs
nethdlc = 1-24
```

Instructions for Cisco HDLC:

1. Compile the Linux kernel with Cisco HDLC support. The Linux kernel menuconfig menu may look similar to the following:

Device Drivers --->
   [*] Network device support --->
     [*] Wan interfaces support --->
       <M> Generic HDLC support
         <M> Cisco HDLC support

Note: Digium recommends using Linux kernel version 2.6.8 or later. The HDLC implementation in Linux kernel versions prior to 2.6.8 may not be reliable or function at all.

2. Install the newly compiled Linux kernel.

3. Reboot into the new Linux kernel.

4. Uncomment the following line in **linux/drivers/dahdi/
   dahdi_config.h** of the DAHDI complete source package:

   ```
   #define CONFIG_DAHDI_NET
   ```

5. Rebuild DAHDI in order to compile the **sethdlc** utility. Then execute
   the following commands from the DAHDI complete source directory:

   ```
   # make sethdlc
   # make install
   ```

6. Load and configure the driver:

   ```
   # modprobe wcte12xp
   # dahdi_cfg
   ```

7. Use **sethdlc** to bring up the interface:

   ```
   # sethdlc hdlc0 cisco
   ```

8. Assign the interface an address:

   ```
   # ifconfig hdlc0 192.168.0.1 netmask 255.255.255.0
   ```

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

Testing Your Configuration

1. Load DAHDI drivers into the kernel using the **modprobe** utility. The appropriate driver for the TE120 Series cards is **wcte12xp**. Users in all countries except Australia should use the following modprobe command:

```
# modprobe wcte12xp
# dahdi_cfg -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

Note: Output as shown above may vary depending on the TE120 Series card you use.

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

```
# dahdi_tool
```

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

```
# asterisk
# asterisk -vvvr
```

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 58.

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 dahdi_tool 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 57.

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 verify that you are dialing using the method your telco is expecting.

I am having trouble receiving DID 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:

```
*CLI> 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 span.

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. Plug in a loopback cable. (A loopback cable is a cable that has pin 1 going to pin 4 and pin 2 going to pin 5.) Plug the cable into the span and wait for its LED to turn green.

2. Stop Asterisk and edit **system.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 the tools/ directory in your DAHDI complete source directory and type:

   ```
   # make tests
   ```

   Followed by:

```
4. # ./patlooptest /dev/dahdi/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

How can I enable more features?

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

```
*CLI> core show applications
*CLI> core show functions
```

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.

Common Fixes for all cards

1. Check to see if the X Window System (*e.g.* X.Org Server) is running by entering the following:

```
# ps aux | grep X
```

If the X Window System is running, stop the application since it may cause a conflict with Asterisk.

2. Check to see if your PATA IDE hard drives are running with DMA levels set. Advance user 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.

Where can I find answers to additional questions?

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

1. Digium Technical Support at 1.256.428.6161, or Toll Free in the U.S. (1.877.344.4861), is available 7am-8pm 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
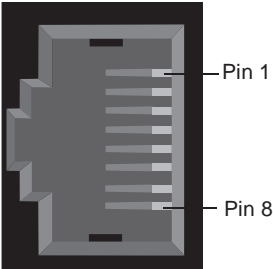
Digium is dedicated to supporting your Asterisk system by offering full technical support through our Subscription Services Program. Through this program, you can be at ease knowing that your business will always have access to the Asterisk experts. Pricing on Subscription Services may be obtained from 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
Pin 1	4	Tx
	5	Tx
Pin 8	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 TE120 Series cards.

Physical (All Cards).

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
(TE120P and TE122) - PCI Bus: 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 (TE120P and TE122) or PCI-E (TE121) Slot

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.

DAHDI   *Digium Asterisk Hardware Device Interface*

A telephony project dedicated to implementing a reasonable and affordable computer telephony platform into the world marketplace. Also, the collective name for the Digium-provided drivers for Digium telephony interface products.

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.

FXS   *Foreign Exchange Station*

Initiates and sends ringing voltage.

G.711

A recommendation by the Telecommunication Standardization Sector (ITU-T) for an algorithm designed to transmit and receive mulaw PCM voice and A-law at a digital bit rate of 64 Kbps. This algorithm is used for digital telephone sets on digital PBX.

G.723.1

A recommendation by the Telecommunication Standardization Sector (ITU-T) for an algorithm designed to transmit and receive audio over telephone lines at 6.3 Kbps or 5.3 Kbps.

G.729a

A recommendation by the Telecommunication Standardization Sector (ITU-T) for an algorithm designed to transmit and receive audio over telephone lines at 8 Kbps.

H.323

A recommendation by the Telecommunication Standardization Sector (ITU-T) for multimedia communications over packet-based networks.

IAX   *Inter-Asterisk eXchange*

The native VoIP protocol used by Asterisk. It is an IETF standard 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.