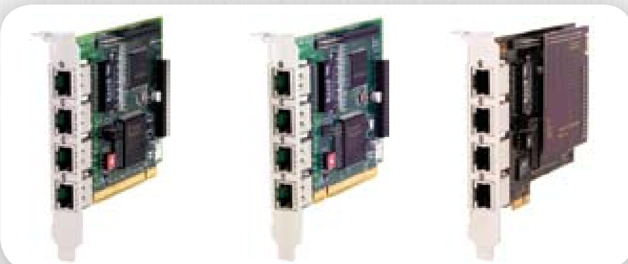
# DIGITAL TELEPHONY CARDS

Digium® cards in the TE series are high-performance, cost effective, digital telephony interfaces and support both E1 and T1 environments. The environments are selectable on a per-card or per-port basis. This feature enables signaling translation between E1 and T1 equipment, and allows inexpensive T1 channel banks to connect with E1 circuits. The bus-mastering TE cards improve I/O speed over slave-only architectures, resulting in reduced CPU usage and increased card density per server. The cards provide the power to interconnect traditional telephony systems with emerging Voice-over IP (VoIP) technologies.

The TE cards support industry standard telephony and data protocols, including Primary Rate ISDN (both N. American and Euro Standard) protocol families for voice, PPP, Cisco HDLC, and Frame Relay data modes. Both line-side and trunk-side interfaces are supported, as well as advanced call features.

The TE cards have been designed to be fully compatible with existing software applications. They are fully integrated with Digium's Asterisk® software. The open source drivers for these cards support an API for custom application development. With the combination of Digium hardware and Asterisk software, numerous telephony configurations are possible. From the traditional PBX to VoIP Gateways, Digium solutions are paving the way for a new generation of worldwide communications.

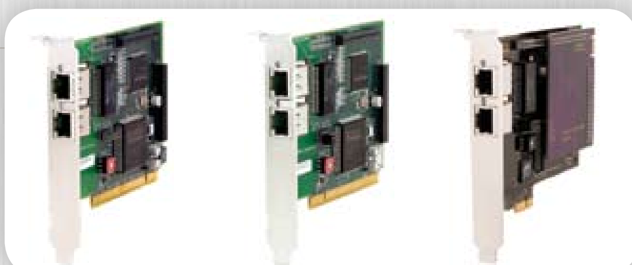


## 4 Port T1/E1 PRI

4 Digital Interface Ports Support both Voice and Data
Selectable T1, E1 or J1 Mode
Half-Length, Full-height Card
TE405P for use only with a 5.0V PCI 2.2 compliant slot.
- Bundled with VPMOCT128 Echo Cancellation Module as TE407P

TE410P for use only with a 3.3V PCI 2.2 compliant slot.
- Bundled with VPMOCT128 Echo Cancellation Module as TE412P

TE420 for use only with a PCI-Express 1.0 compliant slot.
- Bundled with VPMOCT128 Echo Cancellation Module as TE420B

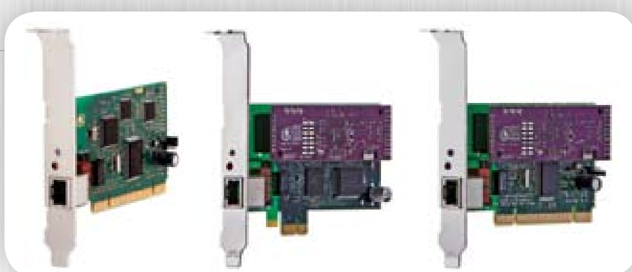


## 2 Port T1/E1 PRI

2 Digital Interface Ports Support both Voice and Data
Selectable T1, E1, or J1 Mode
Half-length Digital Card
TE205P for use only with a 5.0V PCI 2.2 compliant slot.
- Bundled with VPMOCT64 Echo Cancellation Module as TE207P

TE210P for use only with a 3.3V PCI 2.2 compliant slot.
- Bundled with VPMOCT64 Echo Cancellation Module as TE212P

TE220 for use only with a PCI-Express 1.0 compliant slot.
- Bundled with VPMOCT64 Echo Cancellation Module as TE220B



## 1 Port T1/E1 PRI

1 Digital Interface Port Supports both Voice and Data
Selectable T1, E1, or J1 Mode
Half-length, Low Profile, Digital Card
TE120P for use only with a 3.3V or 5.0V PCI 2.2 compliant slot.
TE122 for use only with a 3.3V or 5.0V PCI 2.2 compliant slot.
- Bundled with VPMOCT032 Echo Cancellation Module as TE122B

TE121 for use only with PCI-Express 1.0 compliant slot.
- Bundled with VPMOCT032 Echo Cancellation Module as TE121B

### Target Applications

Legacy PBX/IVR Services
Voice-over Internet Protocol (VoIP) services
Complex IVR Trees
"Meet-Me" Bridge Conferencing
VoIP Gateways (supports SIP, H.323 and IAX)
Calling Card Platforms
Voice/Data Router (replace expensive routers)
PRI Switch Compatibility – Network or CPE

### PRI Switch Compatibility

EuroISDN (PRI or PRA) – Q.931/Q.921
AT&T 4ESS
DMS 100
Lucent 5E
Network or CPE
National ISDN 2
CAS Voice Modes

### Data Modes

SyncPPP (both Fixed and Dialup)
Frame Relay
Cisco HDLC
Multi-link PPP

### CAS Voice Modes

Feature Group D
E&M Wink
a-Law, μ-Law, and Linear Modes Supported

Version 1.3 / 17 January 2008

# DIGIUM®

Digium® is the creator and primary developer of Asterisk®, the industry's first Open Source PBX.

## ABOUT DIGIUM®

Digium, Inc., the Asterisk company, is the original creator and primary developer of Asterisk, the industry's first open source telephony platform. Digium provides hardware and software products, including AsteriskNOW™, the complete open source software appliance; Asterisk Business Edition™, the professional-grade version of Asterisk; and the Asterisk Appliance™, a hardware-based telephony solution, to enterprises and telecommunications providers worldwide, and Switchvox™ turnkey communication solutions for business. Digium also offers a full range of professional services, including consulting, technical support and custom software development.

Used in combination with Digium's telephony interface cards, Asterisk offers a strategic, highly cost-effective approach to voice and data transport over IP, TDM, switched and Ethernet architectures. Digium's offerings include VoIP, conferencing, voicemail, legacy PBX, IVR, auto attendant, media servers and gateways, and application servers and gateways.

## ABOUT ASTERISK®

Code for Asterisk, originally written by Mark Spencer of Digium, Inc., has been contributed from open source software engineers around the world. Currently boasting over two million users, Asterisk supports a wide range of TDM protocols for the handling and transmission of voice over traditional telephony interfaces, featuring VoIP packet protocols such as SIP and IAX among others. It supports U.S. and European standard signaling types used in business phone systems, allowing it to bridge between next-generation voice-data integrated networks and existing infrastructure.

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