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.

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

4 Port T1/E1 PRI

4 Digital Interface Ports Support both Voice and Data
Selectable T1, E1 or J1 Mode / Half-Length, Full-height, Digital 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, Full-height, 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
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

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Under the Risk-Free Guarantee, Digium® will refund the purchase price of any qualifying Digium product(s) for any customer that is not 100% satisfied with the performance of the Digium product(s) they purchased. For more detailed information about Digium's ESP Program, please visit www.digium.com/esp.

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Digium® is the creator, sponsor, and innovative force behind Asterisk®, the industry's first and world's most popular open source telephony software. Additionally, Digium provides a variety of VoIP communication solutions that fit the needs of small, medium, and large businesses. Digium's product lines include commercial business phone systems, as well as software, hardware, and other components needed to create powerful custom telephony solutions.

BUSINESS PHONE SYSTEMS

Digium's line of award winning Switchvox IP PBX phone systems are built on a strong foundation of our open source Asterisk software. Switchvox solutions are designed to be extraordinarily easy to use and provide features that most small and medium businesses had previously considered out of their reach.

Switchvox's web-aware unified communications capabilities are unique in an industry cluttered with old technology. Integration with web and back office applications turns your phone system into a powerful platform for employees' productivity and efficiency. Its web-based Switchboard provides an intuitive control panel to assist with call management in real time while unifying phone calls, faxes, e-mails, instant messaging, Google Maps, CRM systems and other web tools from an easy-to-use, centralized control panel.

We're able to offer these PBX systems with superior capabilities for a fraction of the cost of traditional vendors' products because the shift to an open source software foundation represents a dramatic leap forward in telephony technology.

CUSTOM TELEPHONY SOLUTIONS

Digium empowers users, developers and integrators to build custom telephony solutions by offering a variety of software, hardware, and third-party components. From a simple phone system, to a sophisticated telephony application, Digium makes it possible for the world to communicate in an infinite number of ways at a fraction of the cost of proprietary solutions.

At the heart of these offerings is Asterisk, the powerful open source telephony development toolkit. Asterisk is free software that turns an ordinary computer into a feature-rich voice communications platform. Its flexible architecture lets you configure it as an IP PBX, a voicemail server, IVR server, VoIP gateway, call recorder, automatic call distributor and virtually any other voice-enabled application you can imagine.

To support Asterisk-based solutions, Digium offers a full line of high quality analog and digital interface cards to connect your solution to the public telephone network. In addition, Digium offers add-on software components like Skype® For Asterisk, Fax For Asterisk, G.729 codec, and high performance echo cancellation (HPEC) to enhance your solution.