DIGITAL PCI EXPRESS TELEPHONY CARDS

Digium's TE220 and TE420 PCI Express cards are high-performance, cost effective, digital telephony interface cards which support both T1 and E1 environments, making the cards ideal for interfacing with your Asterisk® based system. The environments are selectable on a per-card or per-port basis. This feature enables signalling translation between T1 and E1 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 telphony systems with emerging Voice-over-IP (VoIP) technologies.

The TE420 may be combined with Digium's VPMOCT128 Octasic DSP-based echo cancellation module. The VPMOCT128 provides the G.168 algorithm, which has been labeled a benchmark for echo cancellation and performs 128ms (1024 taps) of echo cancellation across all 120 channels in E1 mode or all 96 channels in T1/J1 modes. The TE220 may be combined with Digium's VPMOCT64 Octasic DSP-based echo cancellation module, which performs 128ms (1024 taps) of hardware-based echo cancellation across 60 channels in E1 mode or 48 channels in T1/J1 modes.

The TE220 and TE420 PCI Express 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 interfraces are supported, as well as advanced call features.

TE420

4 Digital Interface Ports Support both Voice and Data
Selectable T1, E1, or J1 Mode
PCI-Express 1.0 Compliant
Can be used in PCI-e x1, x4, x8, or x16 slot
Bundled with VPMOCT128 Hardware Echo Cancellation Module as TE420B

TE220

2 Digital Interface Ports Support both Voice and Data
Selectable T1, E1, or J1 Mode
PCI-Express 1.0 Compliant
Can be used in PCI-e x1, x4, x8, or x16 slot
Bundled with VPMOCT64 Hardware Echo Cancellation Module as TE220B

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

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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. 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.

www.asterisk.org

www.asterisknow.org