digium data sheet

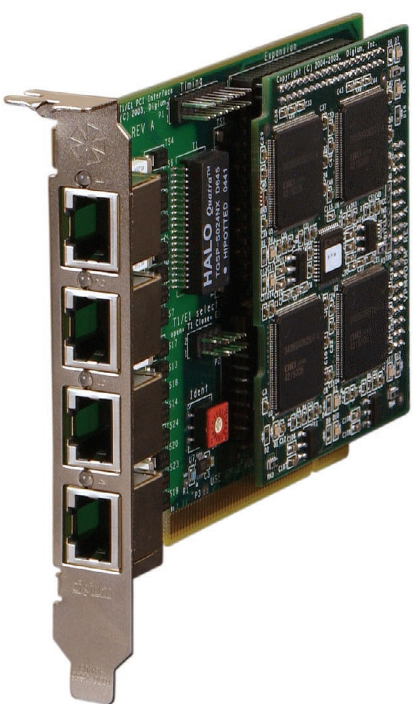
TE411P

Ultimate Density and Performance with on-board echo cancellation

The TE411P is the next generation of Digium hardware that offers an on-board echo cancellation module. It supports both E1 and T1 environments and is selectable on a per-card or per-port basis.

The echo cancellation module supports all T1 and E1 channels and improves voice quality in environments where software echo cancellation is not sufficient. The TE411P reduces CPU overhead required for software echo cancellation thereby, freeing resources for other processes, such as codec translation. By supporting 16ms with 128 channels or 64ms on 32 channels this card will perform in the most difficult of environments.

Digium has designed the TE411P to be fully compatible with existing software applications and it is fully integrated with the Asterisk Open Source PBX/IVR platform. Also, the open source driver supports an API interface for custom application development. With the combination of Digium Hardware and Asterisk software, numerous combinations of telephony configurations are possible. From the traditional PBX to VoIP Gateways, Digium solutions are paving the way for a new generation of worldwide communications.



The TE411P supports industry standard telephony and data protocols, including Primary Rate ISDN (both N. American and Standard Euro) protocol families for voice, PPP, Cisco, HDLC, and Frame Relay data modes. Both line-side and trunk-side interfaces are supported, also included are advanced call features.

The TE411P is for use only with a 3.3 volt PCI slot. The TE406P is for use only with a 5.0 volt PCI slot.

The TE411P is certified for Europe, North America, and Australia. Please visit both our web sites at *www.digium.com* and *www.asterisk.org*.

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

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

CAS Voice Modes

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

Data Modes

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

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About Digium

Based in high-tech Huntsville, Alabama, Digium is the creator and primary developer of Asterisk, the industry's first Open Source PBX. Used in combination with Digium's PCI telephony interface cards, Asterisk offers a strategic, highly cost-effective approach to voice and data transport over TDM, switched, IP, and Ethernet architectures.

Digium solutions reduce the costs of traditional TDM and VoIP implementations through Open Source, standards-based software and innovative hardware solutions, including legacy PBX, IVR, Auto-attendant, and next-generation gateways, media servers, and application servers. Digium hardware supports traditional voice protocols, including PRI, RBS, FXS, FXO, E&M, Feature Group D, Groundstart, and Loopstart. Data protocols include PPP, Cisco HDLC, and Frame Relay. For packet voice, Asterisk supports IAX (Inter-Asterisk eXchange), SIP, MGCP, Skinny, and H.323 VoIP protocols.

Digium provides a highly refined selection of quality hardware and software products, developed and implemented using innovative engineering techniques (primarily Open Source development). A full range of professional services complement these product lines, including consulting, technical support, and custom software development services.

The Open Source communications revolution is here, and Digium is leading the way.