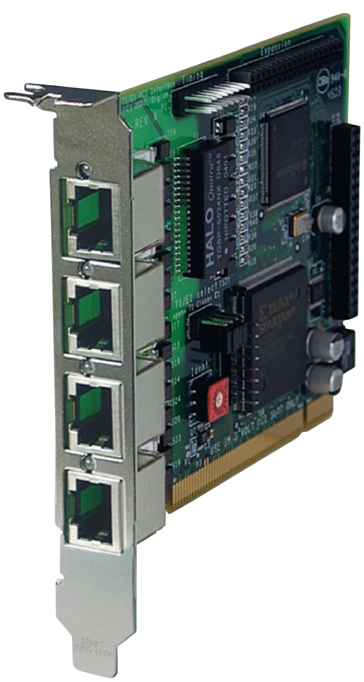
digium data sheet

TE405P

Ultimate Density and Performance in both T1 and E1

The TE410P is the next generation of Digium hardware that improves performance and scalability through bus mastering architecture. The TE410P supports both E1 and T1 environments and is 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. Because the TE405P improves I/O speed by up to 10 times, the result is reduced CPU usage and increased card density per server.

Digium has designed the TE405P 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 TE405P 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 TE405P is for use only with a 5.0 volt PCI slot. The TE410P is for use only with a 3.3 volt PCI slot.

The TE405P 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, Mu-Law, and Linear Modes Supported

Data Modes

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

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.