digium data sheet

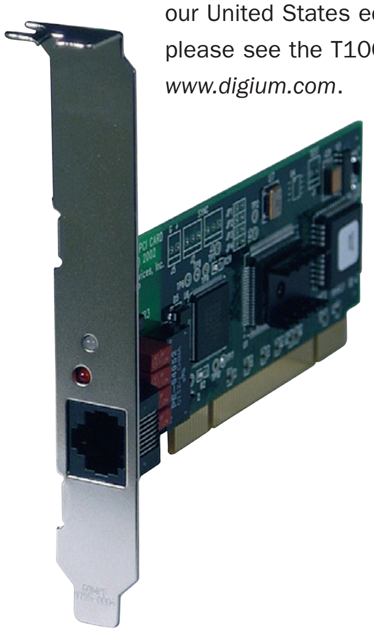
Wildcard E100P

High-performance and Cost-effective E1

The Wildcard E100P brings a high-performance, cost-effective, and flexible single span E1/PRI interface to the Digium line-up of telephony interface devices. The E100P is a compact and powerful interface supporting voice and data transmission over E1 and Primary Rate ISDN (PRI) connections.

The E100P supports industry-standard telephony and data protocols, including PRI protocol for voice and Cisco HDLC, PPP and Frame Relay for data transmission. A low profile, half-length PCI form factor allows this device to fit within a 2U rackmount case or equivalent chassis, offer excellent density for call center, service provider, and other space-sensitive applications.

Of course, the E100P is fully supported by Digium's Open Source Asterisk PBX software. Used in conjunction with Asterisk, the E100P offers the power to create a seamless network, interconnecting traditional telephony systems with the emerging Voice-over IP technologies. The E100P can be used to deliver a wide range of PBX and IVR services to the network or handset including Voicemail, Call Conferencing, Three-Way Calling, and VoIP Gateways. If you would like information concerning our United States equivalent of the E100P, the T100P, please see the T100P product data sheet available at *www.digium.com*.



Target Applications

- Packet Voice Gateways and Switches
- Calling Card Services
- One Number Services
- Message Services
- Conferencing
- Customized and Web Telephony
- Voice/Data Integration
- Future-Proof PBX
- ISDN Remote Access Servers

PRI Switch Capability

- EuroISDN
- Network or CPE

RBS Voice Modes

- A-Law, Mu-Law, and Linear Modes Supported
- E&M
- E&M Wink
- Feature Group D
- Groundstart (FXO and FXS)
- Loopstart (FXO and FXS) with Optional Disconnect Supervision

Data Modes

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

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.