digium data sheet

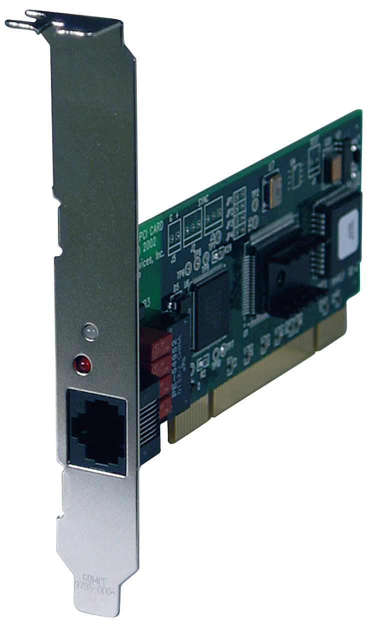
Wildcard T100P

High-performance and Cost-effective T1

The Wildcard T100P brings a high-performance, cost-effective, and flexible single span T1/PRI interface to the Digium line-up of telephony interface devices. The T100P is a compact and powerful interface supporting voice and data transmission over T1 and Primary Rate ISDN (PRI) connections.

The T100P supports industry-standard telephony and data protocols, including Robbed Bit Signaling (RBS), GR-303, and PRI protocols including NFAS (Non-Facility Association Signaling) 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 T100P is fully supported by Digium's Open Source Asterisk PBX software. Used in conjunction with Asterisk, the T100P offers the power to create a seamless network, interconnecting traditional telephony systems with the emerging Voice-over IP technologies. The T100P 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 European equivalent of the T100P, the E100P, please see the E100P 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

- AT&T 4ESS
- DMS 100
- Lucent 5E
- National ISDN 2
- Network or CPE
- NFAS

RBS Voice Modes

- GR-303
- 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

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.

Digium, Inc. www.digium.com · 877.linux.me! · (877.546.8963)