

ANALOG TELEPHONY CARDS

Digium® analog cards were created for connecting analog telephones and analog POTS lines through a PC. Using one of our analog cards in concert with Digium's Asterisk® software, standard PC platforms, and the Linux® OS, one can create telephony environments capable of satisfying the needs of business applications with industry-leading quality.

The analog cards, with their interchangeable single and quad FXS and FXO modules, can eliminate the requirement for separate channel banks or access gateways. Digium's commercial, toll-quality High Performance Echo Cancellation (HPEC) software is available to our analog customers at no additional cost. The optional VPMADT032 hardware echo cancellation module provides the same toll-quality as HPEC, but without the performance impact of a software based solution. Scaling of an analog card solution is accomplished by adding additional cards.

Digium's analog cards utilize patent pending VoiceBus™ technology. VoiceBus technology allows these cards to use an industry standard, bus-mastering interface as found in millions of PCs worldwide. VoiceBus maximizes system compatibility and prevents system conflicts.



4 Port Analog

4 Ports for connecting analog telephones or POTS lines
Half-length Analog Card
Up to 4 FXS or FXO Modules
High Performance Echo Cancellation (HPEC) Software (Optional)
TDM410 for use with a PCI 2.2 compliant slot - Bundled with VPMADT032 as TDM410PE

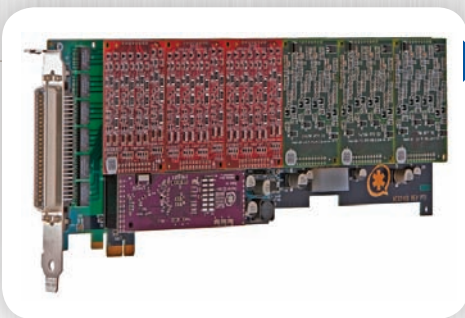
8 Port Analog

8 RJ-11 interfaces on a single PCI bracket
8 Ports for connecting analog telephones or POTS lines
Half-length Analog Card
Up to 4 Single FXS or FXO Modules, or 2 Quad FXS or FXO Modules
High Performance Echo Cancellation (HPEC) Software (Optional)
VoiceBus™ technology
TDM800 for use with a PCI 2.2 compliant slot - Bundled with VPMADT032 as TDM800E
AEX800 for use with a PCI-express 1.0 compliant slot - Bundled with VPMADT032 as AEX800E



24 Port Analog

Up to 24 Ports through a combination of FXS and FXO modules
Full-length Analog Card
Up to 6 Quad FXS or FXO Modules
RJ21X Connector
High Performance Echo Cancellation (HPEC) Software (Optional)
VoiceBus™ technology
TDM2400 for use with a PCI 2.2 compliant slot - Bundled with VPMADT032 as TDM2400E
AEX2400 for use with a PCI-express 1.0 compliant slot - Bundled with VPMADT032 as AEX2400E



Target Applications

Channel Bank Replacement / Alternative
Small Office Home Office (SOHO) applications
Small and Medium Business (SMB) applications
Gateway Termination to analog telephones and lines

Services and Features

Caller ID and Call Waiting Caller ID
ADSI Telephones
Loopstart Signaling Support



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Digium reserves the right to change,
without notice, product offerings
and/or specifications.

DIGIUM®

Digium® is the creator and primary developer of Asterisk®, the industry's first Open Source telephony platform.

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; the Asterisk Appliance™, a hardware-based telephony solution; and Switchvox™, turnkey communication solutions for business. Digium also offers a full range of professional services, including consulting, training, 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, and circuit switched architectures. Digium's suite of products include VoIP, conferencing, voicemail, IP PBX, IVR, automated attendant, media servers, gateways, and application servers.

ABOUT ASTERISK®

Code for Asterisk, originally written by Mark Spencer of Digium, Inc., has been contributed to by open source software engineers around the world. Currently boasting over four million downloads, Asterisk supports a wide range of protocols for the handling and transmission of voice including packet protocols such as SIP and IAX among others. It supports U.S. and international standard signaling formats used in business phone systems, allowing it to bridge between next-generation voice-data integrated networks and existing infrastructure.