Digium® Communications Hardware

The Gold Standard

NO RISK GUARANTEE 100% SATISFACTION

Exceptional Satisfaction Program

Under the Risk-Free Guarantee, Digium will refund the purchase price of any qualifying Digium product(s) for any customer that is not 100% satisfied with the performance of the Digium product(s) they purchased. If Digium can not resolve the trouble to the customer's complete satisfaction, the customer may return the product(s) for a full refund of the purchase price up to MSRP. The refund will be paid in US Dollars only, to the same entity listed on the invoice. There is no predetermined end date for this offer, but Digium reserves the right to discontinue this offer at any time. Digium is committed to complete customer satisfaction.

All Digium analog and digital interface cards:

- are RoHS compliant
- are manufactured in an ISO 9001:2001 certified facility in the United States
- maintain an MTBF greater than one million hours.
- are backed by a five-year hardware warranty
- are backed by Digium's Exceptional Satisfaction Program (ESP)

Digium's Asterisk® software has, since its first release in December of 1999, become the global standard for companies and individuals who need to decrease their cost of ownership of, and improve their control over, telecommunications. With Asterisk, users build anything from simple home-based business solutions to complex multi-location enterprise telephone systems. Since 2001, Digium has designed, manufactured, and sold PC-based interface cards for extending the functionality of Asterisk to the Public Switched Telephone Network (PSTN).

Not only was Digium the first vendor of telephony interface cards built specifically for Asterisk, but it has always been the market leader, with over 50% of the world's board business.

So, what makes Digium's cards the best?

- Digium telephony voice cards are developed by the creators of Asterisk
- The only telephony board products certified to work with Asterisk
- Digium cards have been heavily tested in all the major server hardware platforms, such as Dell, HP, and IBM. Digium cards also work in a variety of hardware platforms with chipsets including AMD, Intel, VIA, Nvidia, etc.
- Digium cards are based on state-of-the-art technology developed by Digium, which ensures that Digium cards will work on the vast majority of hardware platforms. When used with Asterisk, the digium telephony cards provide superior compatibility, quality and performance.
- Digium cards have been tested with multiple Linux-based operating systems, including OpenSUSE, Ubuntu, Fedora, Redhat Enterprise Linux, and CentOS, Gentoo, Linux from Scratch, and Slackware.
- Digium built the first line of telephony cards for Asterisk and continues to be the market leader, selling more telephony cards than any other vendor
- Digium developed the analog signaling and digital line-signaling library used by the Digium cards (as well as competitive cards). If there is ever a question about connectivity between the cards and the protocol stack or signaling, then Digium is uniquely capable to resolve it.

Digium Analog Boards and Modules

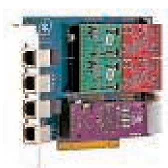
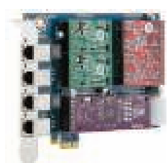
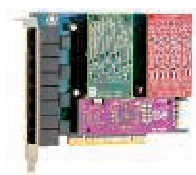
Digium analog cards were created for connecting analog telephones and analog POTS lines through a PC. Using one of our analog cards in concert with Digium's Asterisk software, standard PC platforms, and the Linux® OS, one can create telephony environments capable of satisfying the needs of business applications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

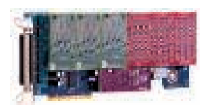
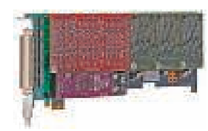
The analog cards, with their interchangeable single and quad FXS and FXO modules, can eliminate the requirement for separate channel banks or access gateways. Digium's commercial, toll-quality High Performance Echo Cancellation (HPEC) software is available to our analog customers at no additional cost. The optional VPMOCT032 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.

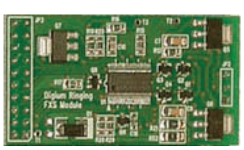
Target Applications

Channel Bank Replacement / Alternative
Small Office Home Office (SOHO) applications
Small and Medium Business (SMB) applications
Gateway Termination to analog telephones/lines
Analog PCI / PCI-Express
4-24 Modular Ports
32-bit 33MHz
Analog Trunk or Station
Loop Start or Kewl Start Signaling
Temperature: 0˚ to 50˚ C
Optional DSP Echo Cancellation

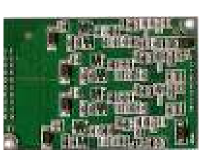




	TDM410	**AEX410**	**TDM800**	**AEX800**	**TDM2400**	**AEX2400**
Ports	4	4	8	8	24	24
	RJ11	RJ11	RJ11	RJ11	50-pin RJ21	50-pin RJ21
Bus Type	PCI 2.2+	PCI-E 1.0+	PCI 2.2+	PCI-E 1.0+	PCI 2.2+	PCI-E 1.0+
Connector	3.3/5.0V	X1	3.3/5.0V	X1	3.3/5.0V	X1
Dimensions	16.46 x 10.67 x 1.72 cm	16.46 x 10.67 x 1.72 cm	16.46 x 10.67 x 1.72 cm	16.46 x 10.67 x 1.72 cm	31.19 x 10.67 x 1.72 cm	31.19 x 10.67 x 1.72 cm
Half-Length	Yes	Yes	Yes	Yes	No	No
FXO Module	Single Trunk	Single Trunk	Single and/or Quad Trunk	Single and/or Quad Trunk	Quad Trunk	Quad Trunk
FXS Module	Single Station	Single Station	Single and/or Quad Trunk	Single and/or Quad Trunk	Quad Station	Quad Station
Echo Cancellation	VPMOCT032	VPMOCT032	VPMOCT032	VPMOCT032	VPMOCT032	VPMOCT032



S100M

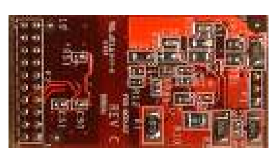


S400M

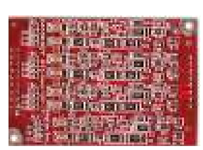
Station-FXS

(Foreign Exchange Station)

FXS is an interface that connects to a station, such as an analog telephone or the FXO interface of another PBX. It provides ringing voltage and battery to the FXO devices. FXS interfaces are used on the inside of your PBX, they do not connect directly to the PSTN. One FXS channel is required for each telephone that you wish to connect to your Asterisk system.



X100M



X400M

Trunk-FXO

(Foreign Exchange Office)

FXO is an interface that connects to a trunk line, like the one from your service provider. It receives ringing voltage and battery from FXS devices. FXO interfaces are used to connect your PBX to the PSTN. One FXO channel is required for each line your receive from your telco.

Digium Digital Boards

Digium cards in the TE series are high-performance, cost effective, digital telephony interfaces which support T1 and E1 environments. The environments are selectable on a per-card or per-port basis. This feature enables signaling translation between T1 and E1 equipment, and allows inexpensive T1 channel banks to connect with E1 circuits. The bus-mastering TE cards improve I/O speed over slave-only architectures, resulting in reduced CPU usage and increased card density per server. The cards provide the power to interconnect traditional telephony systems with Voice-over IP (VoIP) technologies.

The TE cards support industry standard telephony protocols, including Primary Rate ISDN (both N. American and Euro Standard) protocol families. Both line-side and trunk-side interfaces are supported, as well as advanced call features.
The TE cards have been designed to be fully compatible with existing software applications. They are fully integrated with Digium's Asterisk software. The open source drivers for these cards support an API for custom application development. With the combination of Digium hardware and Asterisk software, numerous telephony configurations are possible. From the traditional PBX to VoIP Gateways, Digium solutions are paving the way for a new generation of worldwide communications.

	TE133	TE134	TE205	TE210	TE220	TE405	TE410	TE420	TE820
	1 Port E1/T1/J1		2 Port E1/T1/J1			4 Port E1/T1/J1			8 Port E1/T1/J1
Ports	1		2			4			8
Bus Type	PCI-E 1.0+	PCI 2.2+	PCI 2.2+	PCI 2.2+	PCI-E 1.0+	PCI 2.2+	PCI 2.2+	PCI-E 1.0+	PCI-E 1.0+
Connector	X1	3.3/5.0V	5.0V	3.3V	X1	5.0V	3.3V	X1	X1
Dimensions (cm)	12.7 x 5.4 x 1.6		12.7 x 9.53 x 1.6			12.7 x 9.53 x 1.6			16.8 x 9.5 x 2.2
Low-Profile	Yes		No			No			No
Half-Length	Yes		Yes			Yes			Yes
Echo Cancellation	Built-in		VPMOCT064			VPMOCT128			VPMOCT256

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 – Network or CPE

Digium Digital PCI/PCI-Express

1-8 Ports E1/T1/PRI
32-bit 33MHz
RJ48C Ports
Built-in CSU/DSU
RoHS Compliant
5 Year Warranty
Temperature: 0 to 50° C
Built in Octasic DSP Echo Cancellation for the TE133 and TE134
Optional Octasic DSP Echo Cancellation (2, 4, and 8 port models)

Framing Types

Superframe (D4)
Extended Superframe (ESF)
Channel Associated Signaling (CAS)
Common Channel Signaling (CCS)

Coding Types

Alternative Mark Inversion (AMI)
Bipolar with Eight (8) Zeros Substitution (B8ZS)
High Density Bipolar of Order Three (3) Code (HDB3)*
*Optional Cyclic Redundancy Check 4 (CRC4)
Primary Rate Interface (PRI) Switch Types
National ISDN 1 (NI1)
National ISDN 2 (NI2)
Nortel DMS100
AT&T 4ESS
Lucent 5ESS
EuroISDN Q.931
Q.SIG (Limited Support)
*Both PRI NET and PRI CPE are Supported

Signaling Types

E&M, E&M E1, E&M Wink
Feature Group D (DTMF)
Feature Group D (MF)
Feature Group D (Tandem Access)
Feature Group B
Feature Group C-CAMA
Feature Group C-CAMA (MF)
Foreign Exchange Station (FXS) Loop Start, Ground Start, Kewl Start
Foreign Exchange Office (FXO) Loop Start, Ground Start, Kewl Start
PRI Network and CPE

TC400 Series - Voice Processing

Asterisk, in software and with Digium G.729a licensing, is capable of transforming the G.729a codec into other codecs for the purposes of call origination or termination, bridging disparate calls, or VoIP to TDM connectivity. These transformations in software are very expensive, in terms of MIPS, and require a substantial amount of CPU time to accomplish. The TCE400B and TC400B not only relieve the CPU of this duty, freeing it up to handle other tasks or complete additional call processing, but also provide Asterisk with the capability of bridging G.723.1 compressed audio into other formats, a capability not otherwise possible.

The TCE400B and TC400B decompress G.729a (8.0kbit) or G.723.1 (5.3kbit/6.0kbit) into G.711 u-law or a-law and compress G.711 u-law or a-law into G.729a (8.0kbit) or G.723.1 (5.3kbit). The TCE400B and TC400B are rated to handle up to 120 bi-directional G.729a-only transformations or 92 bi-directional mixed-mode G.729a/G.723.1 transformations. The TCE400B and TC400B do not require additional licensing fees for the use of these codecs nor does it require the registration process association with Digium's software-based G.729a codec licensing.

The TCE400B is a bundle of the half-length, low-profile PCI-Express x1 TCE400P base card and the TC400M voice processing module. The TC400B is a bundle of the half-length, low-profile PCI 3.3/5.0V TC400P base card and the TC400M voice processing module. The TCE400B and TC400B are designed to handle, in dedicated DSP resources, the complex codec translations for highly compressed audio as would otherwise be processed by Asterisk in software.

Features

TC400M Voice Processing Module
TC400P - Half-Length Low-Profile PCI 2.2+ 3.3/5.0V Card
TCE400P - Half-Length Low-Profile PCI-Express x1 Card
Includes Codec Licensing and Indemnification
120 G.729a Transformations
92 G.723.1 Transformations

Requirements

DAHDI 2.2.0 or greater and
Asterisk 1.6.0.10 or greater
Linux Kernel 2.6
Available PCI-Express or PCI Slot

Target Applications

Media Gateway
Conferencing Server
IVR Server
Distributed Office PBX
Call Centers

Codec Support

G.729a - 8.0kbit/s
G.723.1 - 6.3kbit/s (decode-only)
5.3kbit/s (decode/encode)
G.711 μ-law
G.711 a-law

Hardware Echo Cancellation

In the early days of telephone systems, echo during a call was not much of a problem. It was more likely that your grandma may a have gotten a tad of reverb, or heard what her ears perceived merely as side-tone. As telephone systems have become more modern, they have also become prone to more frequent and bothersome echo.

Echo is most common when you are utilizing a VoIP system. Why? Because a VoIP system often introduces latency, which analog systems do not have, and frequently attempts conversion between a 2-wire and 4-wire system. The result of this is an echo in your conversation so that when you talk on the phone it sounds as if you are throwing messages across the Grand Canyon. That may be mildly amusing to everyone inside the IT department, but is extremely frustrating to everyone else.

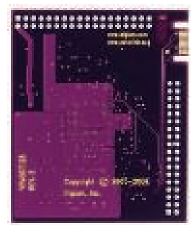
Even though echo may be present, you should never have to experience it when making a call. There are two primary ways with which you can combat this problem: software and hardware. Asterisk does the best job possible utilizing several free echo cancellation tools. While they can do a decent job eliminating minor echoes, they can also do a bad job when the echoes are anything but minor. The best software solution is provided by Digium's High Performance Echo Cancellation (HPEC) software, provided at no-cost to in-warranty Digium® analog hardware customers, and at $10 per channel for non-Digium customers. If your interface card is not equipped with the capability to use a hardware module, this is your best bet!

Fortunately, Digium's latest telephony card offerings have the ability to use hardware echo cancellation modules. Hardware echo cancellation can be more successful, because it removes the burden of echo cancellation from the PC. Hardware echo cancellation is also advantageous when handling large call volumes or a high number of channels that would otherwise stress the CPU and result in the potential for poor audio quality. What makes the hardware echo cancellation so great? Well, how about this:

- Octasic DSP-based (all modules)
- 128ms (1024 taps) of Echo Cancellation across all channels
- AT&T certified Toll-Quality G.168 compliant algorithm
- Dynamic Nonlinear Processor
- Comfort Noise Generator
- Automatic Tail Search
- Cancel Multiple Reflections
- Double-talk Detection

What all this means is that your call has less chance of sounding like you've stepped into a canyon, canyon, canyon or empty concert hall, hall, hall because the hardware echo cancellation module is standards compliant and certified to perform.

There are four hardware echo cancellation modules available to you: the VPMOCT256, VPMOCT128, the VPMOCT064, and the VPMOCT032. The modules support Digium cards currently available, as well as future offerings.

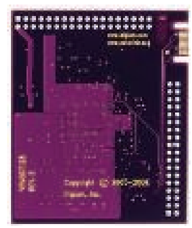


VPMOCT256

- Up to 256 channels

Compatible with the following Digital card:

- TE820F (bundled as TE820BF)

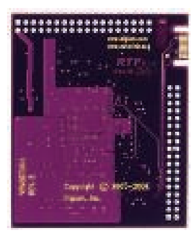


VPMOCT128

- Up to 128 channels

Compatible with the following Digital cards:

- TE405PF (bundled as TE407PF)
- TE410PF (bundled as TE412PF)
- TE420F (bundled as TE420BF)



VPMOCT064

- Up to 64 channels

Compatible with the following Digital cards:

- TE205PF (bundled as TE207PF)
- TE210PF (bundled as TE212PF)
- TE220F (bundled as TE220BF)

VPMOCT032

- Up to 32 channels

Compatible with the following Analog cards:

- 4-port TDM410
- 4-port AEX410
- 8-port TDM800P
- 8-port AEX800
- 24-port TDM2400P
- 24-port AEX2400

Compatible with the following Digital cards:

- 1-port TE121
- 1-port TE122

Telephony Card Selector

Digium now offers a Telephony Card Selector to help you easily determine your hardware needs. Simply select from the available filters (Digital or Analog, # of FXS ports, # of FXO ports, echo cancellation requirements) to find the telephony card(s) that match your requirements.

Visit the online Telephony Card Selector: ***store.digium.com/boards***

We're changing the way the world communications. ***Again.***

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