Digium® Communications Hardware

The Gold Standard

Digium®
The Asterisk Company

Exceptional Satisfaction Program

Under the Risk-Free Guarantee, Digium will refund the purchase price of any qualifying Digium product for any customer who is not 100% satisfied with the performance of the Digium product they purchased. If Digium can not resolve the trouble to the customer's complete satisfaction, the customer may return the product 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.

NO RISK GUARANTEE 100% 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 it's 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 more than 50% of the world's card business.

What makes Digium the best?

- Digium telephony voice cards are developed by the creators of Asterisk

- The only telephony card 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 Cards 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 Linux®, 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. The optional hardware echo cancellation modules provide a G.168-compliant solution to eliminate echo, but without the performance impact of a software-based solution. Scaling of an analog card solution is accomplished by adding additional cards.
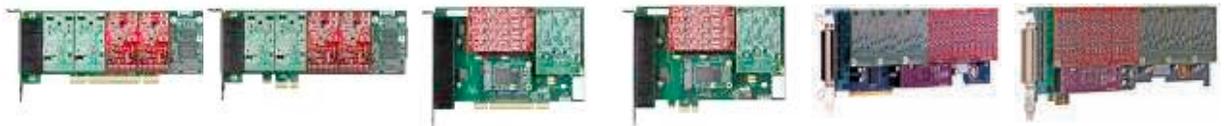
|	A4A	A4B	A8A	A8B	TDM2400	AEX2400
Ports	4	4	8	8	24	24
	RJ11	RJ11	RJ11	RJ11	50-pin RJ21	50-pin RJ21
Bus Type	PCI 2.2+	PCI-E 1.0+	PCI 2.2+	PCI-E 1.0+	PCI 2.2+	PCI-E 1.0+
Connector	3.3/5.0V	X1	3.3/5.0V	X1	3.3/5.0V	X1
Dimensions (cm)	16 x 6.3	16 x 6.3	15.7 x 10.6	15.7 x 10.6	31.19 x 10.67	31.19 x 10.67
Low-Profile	Yes	Yes	No	No	No	No
Half-Length	Yes	Yes	Yes	Yes	No	No
FXO Module	Single Trunk	Single Trunk	Single and/or Quad Trunk		Quad Trunk	Quad Trunk
FXS Module	Single Station	Single Station	Single and/or Quad Station		Quad Station	Quad Station
Echo Cancellation Module	VPM032	VPM032	VPM032	VPM032	VPMOCT032	VPMOCT032

Station: FXS
(Foreign Exchange Station)

S100M

S400M

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.
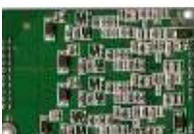
Station: FXO
(Foreign Exchange Office)

X100M

X400M

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 Cards

Digium's digital cards are high-performance, cost- effective, digital telephony interfaces that 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 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.

Digium's digital cards support industry standard telephony protocols, including Primary Rate ISDN (both North American and Euro Standard) protocol families. Both line-side and trunk-side interfaces are supported, as well as advanced call features.

Digium's digital cards have been designed to be fully compatible with existing software applications. They are fully integrated with Digium's Asterisk software. 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.

	TE131/TE133	TE132/TE134	TE235	TE236	TE435	TE436	TE820
	E1/T1/J1		E1/T1/J1		E1/T1/J1		E1/T1/J1
Ports	1		2		4		8
Bus Type	PCI-E 1.0+	PCI 2.2+	PCI-E 1.0+	PCI 2.2+	PCI-E 1.0+	PCI 2.2+	PCI-E 1.0+
Connector	X1	3.3V/5.0V	X1	3.3V/5.0V	X1	3.3V/5.0V	X1
Dimensions (cm)	12.7 x 5.4		12.7 x 6.7	12.7 x 6.7	12.7 x 6.7	12.7 x 6.7	16.8 x 9.5
Low-Profile	Yes						No
Half-Length	Yes						
Echo Cancellation Module	TE131 & TE132 - N/A TE133 & TE134 - Built-In		VPM064	VPM064	VPM128	VPM128	VPMOCT256

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)

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

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

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 callers experienced side-tone, or heard what their ears perceived as reverb. 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 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 echo heard by callers on a VoIP system whenever they call someone on the PSTN.

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.

Fortunately, Digium's 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?

- Octasic DSP-based (all modules)
- 128ms (1024 taps) of echo cancellation across all channels
- G.168-2004-compliant algorithm
- Dynamic Nonlinear Processor
- 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.

	VPMOCT256	VPMOCT032	VPM032	VPM064	VPM128
Max. Channels	256	32	32	64	128
Card Compatibility	TE820F (TE820BF)*	TDM2400P AEX2400	A4A A4B A8A A8B	TE235 TE236	TE435 TE436

* TE820BF is bundled with the echo cancellation module

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 to 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 do they ᵒrequire 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

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

TCE400B and TC400B
The TCE400B (left) and TC400B (right) transform the G.729a codec into other codecs, freeing the CPU to handle other tasks. They also provide Asterisk with the capability of bridging G.723.1 compressed audio into other formats.

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, number of FXS ports, number 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**

digium®
The Asterisk Company

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

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