HARDWARE ECHO CANCELLATION

HFI10?



digium Asterisk

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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:

- AT&T certified Toll-Quality G.168 compliant algorithm
- Dynamic Nonlinear Processor
- · Comfort Noise Generator
- · Automatic Tail Search
- · Cancel Multiple Reflections
- Double-talk Detection
- · 128ms of Echo Cancellation across all channels

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 three hardware echo cancellation modules available to you: the VPMOCT128, the VPMOCT64, and the VPMADT032. The modules support Digium cards currently available, as well as future offerings.

VPMOCT128	•••	VPMOCT64		VPMADT032
 · Up to 128 channels · 128ms (1024 taps) per channel 		 · Up to 64 channels · 128ms (1024 taps) per channel 		 · Up to 32 channels · 128ms (1024 taps) per channel
This module works with the following cards:		This module works with the following cards:		This module works with the following cards:
 • TE410P (bundled as TE412P) • TE405P (bundled as TE407P) • TE420 (bundled as TE420B) 		 • TE210P (bundled as TE212P) • TE205P (bundled as TE207P) • TE220 (bundled as TE220B) 		• 8-port TDM800P • 24-port TDM2400P
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	 VPMOCT128 Up to 128 channels 128ms (1024 taps) per channel This module works with the following cards: TE410P (bundled as TE412P) TE405P (bundled as TE407P) TE420 (bundled as TE420B) 	 VPMOCT128 Up to 128 channels 128ms (1024 taps) per channel This module works with the following cards: 1E410P (bundled as TE412P) 1E405P (bundled as TE407P) TE420 (bundled as TE420B) 	VPMOCT128VPMOCT64• Up to 128 channels • 128ms (1024 taps) per channel• Up to 64 channels • 128ms (1024 taps) per channel• Tis module works with the following cards:This module works with the following cards:• TE410P (bundled as TE412P) • TE405P (bundled as TE407P) 	VPMOCT128VPMOCT64• Up to 128 channels • 128ms (1024 taps) per channel• Up to 64 channels • 128ms (1024 taps) per channel• Tis module works with the following cards:• Tis module works with the following cards:• TE410P (bundled as TE412P) • TE405P (bundled as TE407P) • TE205P (bundled as TE207P) • TE200 (bundled as TE208)• TE410P (bundled as TE408)



DIGIUM[®]

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

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; and the Asterisk Appliance™, a hardware-based telephony solution, to enterprises and telecommunications providers worldwide, and Switchvox™ turnkey communication solutions for business. Digium also offers a full range of professional services, including consulting, technical support and custom software development.

Used in combination with Digium's telephony interface cards, Asterisk offers a strategic, highly costeffective approach to voice and data transport over IP, TDM, switched and Ethernet architectures. Digium's offerings include VoIP, conferencing, voicemail, legacy PBX, IVR, auto attendant, media servers and gateways, and application servers and gateways.

ABOUT ASTERISK®

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