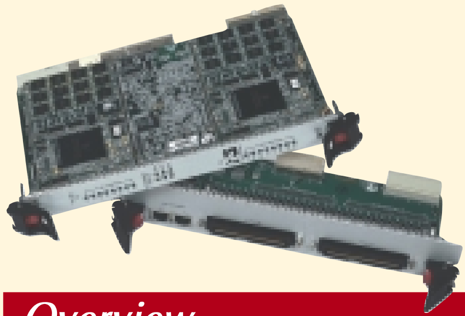
AudioCoded™ Carrier Solutions Data Sheet

AudioCoded™ TP-1610

cPCI Communication Board



The **TP-1610** cPCI communication board joins AudioCodes award-winning product line of enabling technologies for the transmission of voice and fax over IP.

Overview

The **TP-1610** cPCI communication board, based on AudioCodes' TPM-1100 PMC Modules, is an ideal building block for deploying high-density, high availability Voice over IP (VoIP) gateways. **The TP-1610** is suitable for VoIP gateways, carrier grade platforms, IP-enabled call centers, large telcos and next generation DLCs. Offering integrated voice gateway functionality capable of delivering up to 480 simultaneous calls, the **TP-1610** supports all necessary functions for voice and fax streaming over IP networks.

The **TP-1610** is powered by two AudioCodes' TPM-1100 Media Gateway Modules and supports voice related algorithms, including G.168-compliant echo cancellation, G.726 (40,32,24,16 kbps), G.711, G.723.1 and G.729A codecs, as well as MF-R1 (DTMF and MF) and MFC-R2 CAS tone detection and generation. An integrated TDM switch provides connection to an H.110 backplane. Direct connection to PSTN facilities is provided through optional T1/E1 interfaces.

The **TP-1610** board complies with industry standard control protocols including MGCP, MEGACO (H.248) or AudioCodes' proprietary TPNCP. These allow for the implementation of a distributed gateway architecture that separates call-processing functions from media streaming functions, resulting in better redundancy, scalability and higher system availability. Like all AudioCodes products, the **TP-1610** supports the AudioCodes API Library, which enables software download, provisioning and control. By maintaining API backward compatibility, AudioCodes protects its customers' investments in software development.

Enabling accelerated design cycles with high density and reduced costs, the **TP-1610** is an ideal building block for scalable, reliable VoP solutions. With the **TP-1610's** comprehensive feature set, customers can quickly design a wide range of solutions for migration to VoP networks.

Benefits

- High Channel Density
- Reduced System Cost and Increased Reliability
- Fast Time-to-Market
- Flexible and Easy Migration to VoP Networks

Features

- 480 voice/fax channels on a single slot cPCI board
- Integrated 16 E1/16 T1 telephony interfaces
- VoIP packet streaming (RTP/ RTCP) per RFC 1889/1890
- MGCP (RFC 2705), Megaco (H.248) or TPNCP (AudioCodes proprietary)
- Simultaneous support for G.711, G.726, G.727, G.723.1, G.729A voice compression
- Independent vocoder selection per channel
- Support for silence suppression (VAD and CNG)
- Automatic fax bypass mode
- G.168 compliant echo cancellation
- TIA464B DTMF detection and generation
- MF-R1/MFC-R2 and call progress tones detection and generation
- PSTN Signaling : CAS, ISDN PRI, V5.2 (AN)
- TDM switching from H.110 bus or from trunk interfaces
- Hot swap and hot insertion capability
- Dual, Redundant 100Base-T interfaces, cPSB PICMG 2.16 compliant Ethernet on the backplane

Applications

- Next Generation Switches
- IP Services Platforms
- VoIP Access Gateways
- Carrier Grade Trunking Gateways
- Cable Telephony Gateways
- IP Enabled Call Centers

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TP-1610 Selected Specifications

Capacity	480 independent digital LBR/PCM voice / fax ports
Voice Coders	G.711, G.726, G.727, G.723.1, G.729A Independent dynamic vocoder selection per channel
Silence Suppression	• Voice Activity Detection (VAD) • Comfort Noise Generation (CNG)
Echo Cancelation	G.168 compliant 32, 64, 128 msec echo tail (64m and 128m with reduced number of channels)
Gain Control	Programmable
Fax and Modem	• Automatic fax bypass to G.711 • Automatic switch to PCM for modem signals
DTMF and Tone Signaling	DTMF detection and generation per TIA464B MF-R1, MFC-R2 tone detection and generation Call progress tone detection and generation
Control Protocols	MGCP, MEGACO, TPNCP
Management Interfaces	Embedded Web Server SNMP v2: MIB2, RTP MIB, TRUNK MIB, AudioCodes' Proprietary MIB
Ethernet Interface	Dual redundant 100Base-T ports, RJ-45 connectors off rear I/O, cPSB PICMG 2.16 on the backplane
Telephony Interfaces	16 E1 or 16 T1 spans rear panel, using two 50-pin Telco connectors
PSTN Signaling	CAS T1 Robbed bits, E1 MFC-R2, ISDN PRI, V5.2 (AN)
TDM Interfaces	H.110 CT Bus interface
Hot Swap	Full Hot Swap Supported
Power Consumption	38 Watts typical - Preliminary
Cooling Requirements	• 400 LFM at 50°C (amb. temp.) • 250 LFM at 30°C (amb. temp.)
Power Supply	5V; 3.3V
Physical	6U single cPCI slot
Host Interface	Via PCI bus or Ethernet
Mechanical	PICMG 2.0, R2.1, R2.16 compactPCI
PCI Bus Interface	33 MHz, 32 bit, master or slave mode
Operating Systems	• Windows NT 4.0 • Win-NT library • Linux • Solaris/INTEL • Solaris/SPARC

Ordering Information

TP-1610 - H.110 I/F, 100BT, 120ohms

Device Name	Channels (#)	Configuration
TP-1610-480XXHFU-R*	480	REAR I/O, NO TRUNK I/F
TP-1610-48016THFU-R*	480	REAR I/O, 16 T1/E1 TRUNKS I/F

* R=Vocoder options

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