AudioCoded™ Enabling Technology Products

IPM-1610 16 E1/T1 cPCI VoIP Media Processing Board

- IP-enabled, cost-effective technology
- Field-proven PSTN interface board
- High-density, high performance board
- Independent call-by-call basis LBR ports
- Carrier grade applications
- Concurrent toll quality voice and fax support
- Enables scalable distributed architecture
- Shorter development cycle

The **IPM-1610** is a complete VoIP media processing solution providing IP and PSTN interfaces as needed to build next generation applications for both today's and tomorrow's networks. Moreover, introducing a packet interface for voice streaming the board enables a smooth evolution from legacy PCI bus architecture to distributed, scalable, packet-based architecture.

Deliver Feature-Rich Solution

A broad selection of firmware based media processing capabilities is available with the IPM-1610 including: message record/playback, conferencing, on-board announcements storage, voice coding, echo cancellation, fax processing and call progress tone detection. Each channel resource on the IPM-1610 is universal and can perform media processing functions while utilizing full flexibility in endpoints.

Comply with Industry Standards

The IPM-1610 board complies with industry standard network control protocols including MGCP, MEGACO (H.248), SIP or AudioCodes' proprietary TPNCP. These allow for the implementation of a distributed media server architecture that separates call-processing functions from media processing functions, resulting in better redundancy, scalability and higher system availability.

Protect Customer Investment

The IPM-1610 is a member of AudioCodes' 4th generation of the widely deployed TrunkPack® Media Gateway on a blade family. Like the other members of the TrunkPack® family the IPM-1610 supports AudioCodes API, which enables software download, provisioning and control. Maintaining essential API backward compatibility of future releases in order to protect customers' investment, is an important feature of AudioCodes' software update/upgrade program.

Enable Fast & Easy Integration

Enabling accelerated design cycles with high-density and reduced costs, the IPM-1610 is an ideal building block for scalable, reliable VoIP enabled media processing solutions. With the IPM-1610's comprehensive feature set, customers can quickly design a wide range of solutions combining PSTN and VoIP networks.

IPM-1610 Features

- Up to 480 IVR streaming ports
- Up to 240 Universal media processing ports
- Voice Record/Playback
- Fax termination and generation (T.37)
- Interchangeable RTP or PSTN or H.110 endpoints
- Real-time, multi-party conferencing
- Comprehensive IVR control
- VoIP packet streaming (RTP/RTCP) per RFC 1889/1890
- MGCP, MEGACO, SIP and AudioCodes' proprietary TPNCP
- cPSB PICMG 2.16 compliant Ethernet on the backplane
- Automatic Speech Recognition (ASR) [1]
- Text To Speech (TTS) [1]

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IPM-1610

Specifications

Software Specifications	
Configuration	"Streaming"- 480 Voice/Fax Messaging ports "Universal" - 120, 240 universal ports
Voice Messaging, Recording	Host-based record/play, WAV format (G.711, MS-GSM) Up to 240 ports of record/play - PCI bus based Up to 480 ports of record/play - packet streaming based Playback speed control with pitch correction Time Slot summation - Record RX+TX of the call On-board announcement storage - 10 Mb Recorded prompts - 20 minutes of G.711, 200 minutes of G.723
Conferencing	Supports up to 240 ports of Mixed IP, PSTN and TDM (H.110) participants Maximum half-duplex parties per conference bridge: 120 Maximum simultaneous 3-way conferences per board: 80 Maximum full-duplex parties per conference bridge: 64 endpoints[2] Supports various conference control modes
Fax Messaging	Termination/Generation • Supports up to 96 ports - PCI bus or packet streaming based • T.37 store and forward
Fax Relay	Real-time fax over IP/T.38 compliant, automatic fallback to G.711
ASR - 3rd party Recognition Engines	Host-based Architecture - Media Stream over PCI Distributed Architecture - Media Stream over VoIP RTP
Voice Processing	Supports up to 480 ports of: • G.711, G.723.1, G.729A, G.726/G.727, NetCoder® • Additional coders supported - contact AudioCodes for further information Voice Activity Detection (VAD) and CNG Echo Cancellation: G.168 with tail of 30 msec, 64 msec[3] and 128 msec[3] Trans-coding of G.711 RTP to any Low Bit Rate Coder RTP stream Gain Control: Automatic (AGC) or Programmable
In-band/Out-band Signaling	Packet side or PSTN side, DTMF and tone detection and generation, RFC 2833
Control	AudioCodes' proprietary TPNCP, MGCP (RFC 2705), MEGACO (H.248), SIP
Management Interfaces	SNMP V2: Standard MIB-2, RTP MIB, Trunk MIB, AudioCodes' proprietary MIB Embedded Web Server
Operating System	• Windows NT, 2000, XP • Linux • Solaris on Intel/Sparc
Signaling	
PSTN	**CAS** T1 robbed bit, MFC/R2 numerous country variants **CCS** ISDN PRI: numerous country variants including ETSI EURO ISDN, ANSI NI2, DMS, 5ESS, Japan INS1500
SIGTRAN	IUA over SCTP per RFC 3057/2960 SS7 MTP2 link termination M2UA and M3UA over SCTP
Hardware Specifications	
Ethernet	Dual redundant 100 Base-T ports
Hot Swap	Full hot swap supported per PICMG 2.1
Physical Interfaces	Form factor - 6U PICMG 2.0 single cPCI slot TDM Interface - H.110 CT Bus Telephony - two 50-pin Telco connectors on rear panel Ethernet - cPSB PICMG 2.16 on the backplane, Dual RJ-45 on rear panel
Power	40.7W, 3A at 5V, 7.8A at 3.3V

1 Integrated Partner Technologies
2 Available in Q1/03
3 Future release, may affect density

Applications

- Call Centers
- Conference Servers
- IVR Servers
- Unified Communications/Messaging
- Voice Portals
- CTI Applications
- Voice Recording

About AudioCodes

AudioCodes Ltd. (NASDAQ: AUDC) designs, develops and markets Voice over Packet media gateway technologies and systems for converged networks. The company is a market leader in voice compression technology and the key originator of the ITU G.723.1 standard for the emerging Voice over IP market. AudioCodes' product line includes enabling technology products such as Voice over Packet chip processors, VoIP communication boards, VoIP media gateway modules and CPE devices. In addition, AudioCodes provides OEMs with media gateway system solutions for packet networks in the wireline, wireless, broadband access and media server markets.

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