IPM-260

E1/T1 PCI VoIP Media Processing Board

Features

· IP-enabled, cost-effective technology

· Field-proven PSTN interface board

· Low to high channel density

· Independent call-by-call basis LBR ports

· All-in-one integrated board

· Shorter development cycle

· 240 universal ports supporting voice, fax and data

· Various voice compression includes G.711, G.723.1, G.729A

· Voice Record/Playback

· Real-time, multi-party conferencing

· Interchangeable RTP or PSTN or TDM endpoints

· Comprehensive IVR control

· VoIP packet streaming: (RTP/RTCP) per RFC 1889/1890

· MVIP, SCbus and H.100 CT bus interface support

· Automatic Speech Recognition (ASR)

· Text To Speech (TTS)

· Optional Universal PCI Version

Built on the heritage of the AudioCodes Voice over IP processing technology, the IPM 260 family of products are capable of serving the most demanding applications. It has every feature needed to meet all the demands of the CTI market, both today and tomorrow.

Deliver Feature-Rich Solutions

A broad selection of firmware-based media processing capabilities is available with the IPM-260 including: message record/playback, conferencing, voice coding, echo cancellation, fax processing and call progress tones detection. Each channel resource on the IPM-260 is universal and can perform media processing functions while utilizing full flexibility in endpoints.

Universal solution

The IPM-260 is a complete VoIP media processing solution providing IP and PSTN interfaces to build next generation applications for both today's and tomorrow's networks. By combining these capabilities on a single board, the IPM-260 can eliminate a number of separate special-function boards, resulting in reduced inventory, increased over-all system density, reduced costs and improved time-to-market.

Protect Customer Investment

The IPM-260 is based on the VoIPerfectTM architecture, AudioCodes' underlying, best-of-breed, core media gateway technology for all of its products. The IPM-260 supports AudioCodes' API, which enables software download, provisioning and control. It was designed to maintain essential API backward compatibility in order to protect customers' investment in the development of products based on former generations.
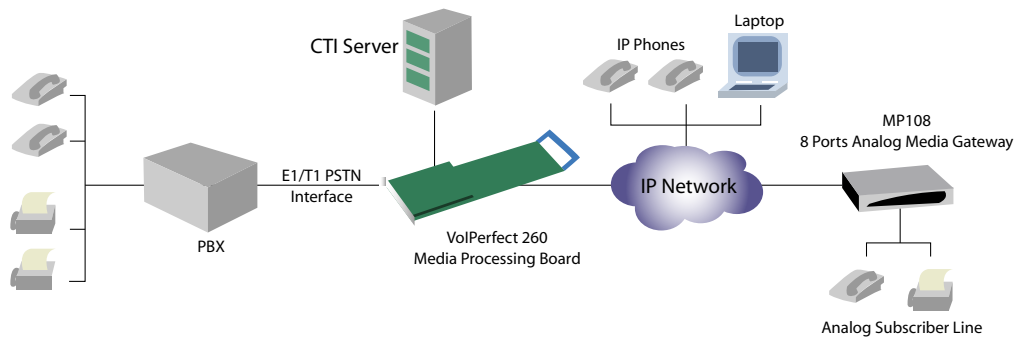
Enable Fast & Easy Integration

Enabling accelerated design cycles with high density and reduced costs, the IPM-260 is an ideal building block for scalable, reliable VoIP enabled media processing solutions. With the IPM-260's comprehensive feature set, customers can quickly design a wide range of solutions combining PSTN and VoIP networks.

Ai-Logix «
An **AudioCodes** Company

Application Diagram

Integration of VoIPerfect Products in NextGen Enviroment

Diagram labels: CTI Server · IP Phones · Laptop · MP108 8 Ports Analog Media Gateway · PBX · E1/T1 PSTN Interface · VoIPerfect 260 Media Processing Board · IP Network · Analog Subscriber Line

SOFTWARE SPECIFICATIONS

CONFIGURATION

30, 60, 120, 240 universal ports

VOICE MESSAGING, RECORDING

Host-based record/play, WAV format (G.711, G.726, MS-GSM)

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

Record/play via standard HTTP Web interface

CONFERENCING

Supports up to 240 ports of mixed IP, PSTN and TDM participants Maximum simultaneous 3-way conferences per board: 40

Maximum full-duplex parties per conference bridge: 64 endpoints

Supports various conference control modes

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

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 compliant 32, 64 msec echo tail;

128 msec tail available with reduced channel capacity

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 VoIP API Library over IP (TPNCP) or PCI

MANAGEMENT INTERFACES

SNMP V2: Standard MIB-2, RTP MIB, Trunk MIB, AudioCodes' proprietary MIB Embedded Web Server

OPERATING SYSTEM SUPPORT

· Windows(TM) NT, 2000, XP · Linux(TM) Solaris(TM) on Sparc(TM)/Intel(TM) 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

10/100 Base-T

PHYSICAL INTERFACES

Form factor - Full length PCI board

TDM Interfaces - MVIP, SCbus, H.100

Telephony - 120 Ohm - RJ48C connectors

Ethernet - RJ-45

POWER

3.6A at 5 V with quad E1/T1 interface

OPTIONAL

Universal PCI 5 V/3.3 V signaling

PCI bus - 32/64 bit, 33/66 MHz

VOIPERFECT™
Your Gateway to VoIP