TP-1610 16 E1/T1 cPCI VoIP Communication Blade

- High channel density
- Concurrent toll quality voice and fax support
- Carrier grade applications
- Reduced system cost and increased reliability
- Fast time-to-market
- Flexible and easy migration to VoIP Networks
- Extensive VoIP experience

AudioCodes
BLADE BUSINESS LINE

The **TP-1610** cPCI VoIP communication blade, based on dual AudioCodes' TPM-1100 PMC Modules, is an ideal building block for deploying high-density, high-availability Voice over IP (VoIP) systems. The TP-1610 is suitable for VoIP gateways, 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.

Deliver Feature-Rich Solutions

The TP-1610 supports a broad selection of voice processing related algorithms, including G.711, G.723.1 and G.729AB Vocoders, G.168-compliant echo cancellation, T.38 real-time Fax over IP, and a wide selection of In-band and Out-band tone detection and generation. In addition, signaling protocols supported include ISDN PRI, SIGTRAN (M2UA, M3UA, IUA) and CAS.

Comply with Industry Standards

The TP-1610 blade complies with industry standard network control protocols including MGCP, MEGACO (H.248), SIP, H.323 as well as 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.

Protect Customer Investment

The TP-1610 is based on the VoIPerfect™ architecture, AudioCodes' underlying, best-of-breed, core media gateway technology for all of its products. The TP-1610 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 TP-1610 is an ideal building block for scalable, reliable VoIP solutions. With the TP-1610's comprehensive feature set, customers can quickly design a wide range of solutions for migration to VoIP networks.

TP-1610 Features

- Up to 480 voice/fax independent multiple LBR channels
- VoIP packet streaming (RTP/RTCP) per RFC 3550/3551
- Standard control: MGCP (RFC 2705), MEGACO (H.248), SIP/H.323
- Real-time Fax over IP/T.38
- On-board announcement support towards PSTN/TDM and IP
- Tone detection and generation (MF, DTMF, RFC 2833)
- PSTN Signaling: CAS, ISDN PRI, V5.2 (AN), and SS7
- SIGTRAN IUA, M2UA, M3UA over SCTP
- cPSB PICMG 2.16 compliant Ethernet on the backplane

TP-1610

Specifications

Software Specifications	
Capacity	60, 120, 240 or 480 independent digital voice, fax and data ports
Voice Compression	G.711, G.723.1, G.729A/B, G.726/G.727, NetCoder® MS-GSM, GSM-FR, iLBC Additional coders supported – contact AudioCodes for further information
Echo Cancellation	G.168 compliant 32, 64, 128[1] msec echo tail
Gain Control	Programmable
Fax Relay and Termination	Real-time fax over IP/T.38 compliant, automatic fallback to G.711 and VBD for up to super G-3 fax machines Support for Fax Termination (Available with AudioCodes S/W based stack)
ASR - 3rd party Recognition Engines	Host-based Architecture – Media Stream over PCI Distributed Architecture – Media Stream over VoIP RTP
In-band/Out-band Signaling	Packet side or PSTN side, DTMF and tone detection and generation
IVR Support	On-board announcement storage – 10 Mb Recorded prompts – 20 minutes of G.711, 200 minutes of G.723
VoIP Standards Compliance	RTP/RTCP per RFC 3550/3551 DTMF over RTP per RFC 2833
Control Protocols	Media Gateway on a blade mode: Controlled by either MGCP or MEGACO PCI used for power only SIP, H.323 TPNCP – AudioCodes' proprietary VoIP API Library
Management Interfaces	• SNMP V2c: Standard MIB-2: system, interfaces, if-MIB, entity-MIB, RTP-MIB, DS1-MIB, snmpV2-MIB and AudioCodes' proprietary MIB • On-board embedded secure Web Server
Operating System	• Windows™ 2000, XP, 2003 • Linux™[2] RH8, RH9, Debian, Enterprise • Solaris™[2] 8,9 on Intel™/Sparc™ 32/64
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 V5.2 AN (Contact AudioCodes) **SS7** MTP2 and MTP3 link termination ISUP and SCCP/TCAP termination (Available with AudioCodes S/W based stack)
SIGTRAN	M2UA, M3UA, IUA and DUA over SCTP per RFC 2960
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 With reduced channel capacity
2 See release note for specific OS releases supported

Applications

- Next Generation Switches
- IP Services Platforms
- VoIP Access Gateways
- Trunking Gateways
- Cable Telephony Gateways
- IP Enabled Contact Centers

About AudioCodes

AudioCodes Ltd. (NASDAQ: AUDC), Your Gateway to VoIP, provides innovative, reliable and cost-effective Voice over Packet (VOP) technology and Voice Network products to OEMs, Network Equipment Providers, Service Providers and System Integrators worldwide. AudioCodes provides a diverse range of flexible, comprehensive media gateway and media processing technologies (based on VoIPerfect™ – AudioCodes' underlying, best-of-breed, core media gateway architecture) and Session Border Controllers (SBCs). The company is a market leader in product development, focused on VoIP Media Gateway, Media Server and SBC technologies and network products. AudioCodes has deployed tens of millions of media gateway and media server channels globally over the past few years and is a key originator of the ITU G.723.1 standard for the emerging Voice over IP market. The Company is a VoIP technology leader focused on quality, having recently received a number one ranking from ETSI for outstanding voice quality in its media gateways and media servers. AudioCodes voice network products feature media gateway and media server platforms for packet-based applications in the converged, wireline, wireless, broadband access, enhanced voice services and video markets. AudioCodes enabling technology products include VoIP and CTI communication blades, VoIP media gateway processors and modules, and CPE devices. AudioCodes' headquarters and R&D facilities are located in Israel with an R&D extension in the U.S. Other AudioCodes' offices are located in Europe, the Far East, and Latin America.

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