# TP-6310 STM-1/OC-3 cPCI VoP Communication Blade

- Very high channel density – up to 2016 LBR VoIP channels
- Packet to Packet mediation
- Concurrent toll quality voice and fax support on all channels
- STM-1/OC-3 Automatic Protection Switching on PSTN interface
- Designed for carrier-grade applications
- Built on 4 previous generations of proven and widely deployed VoP technology
- Flexible and easy migration to VoP networks

AudioCodes
BLADE BUSINESS LINE

The **TrunkPack®-6310** cPCI VoP communication platform, with a STM-1/OC-3 PSTN interface, is an ideal building block for deploying high-density, high availability Voice over Packet (VoP) systems. The TP-6310 is suitable for high-density VoIP gateways, Packet to Packet mediation, media servers and cable telephony gateways. Offering integrated voice and signaling gateway functionality, the TP-6310 supports all necessary functions for voice and fax streaming over IP networks.

## Deliver Feature-Rich Solutions

The TP-6310 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, a wide selection of In-band and Out-band tone detection and generation. In addition, the TP-6310 also supports signaling protocols including ISDN PRI, SIGTRAN (M2UA, M3UA, IUA) and CAS. All media processing, signaling and control protocols are applied independently and simultaneously on all of the 2016 LBR channels.

## Comply with Industry Standards

The TP-6310 blade complies with industry-standard network control protocols including MGCP, MEGACO (H.248) as well as AudioCodes' proprietary API (TPNCP). This allows for the implementation of a distributed gateway architecture that separates call-processing functions from media streaming functions. This results in better redundancy, scalability and higher system availability.

## Protect Customer Investment

The TP-6310 is based on the VoIPerfect™ architecture, AudioCodes' underlying, best-of-breed, core media gateway technology for all of its products. The TP-6310 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-6310 is an ideal building block for reliable VoP solutions. With the TP-6310's comprehensive feature set, customers can quickly design a wide range of solutions for migration to VoP networks.

## TP-6310 Features

- 2016 voice/fax independent multiple LBR channels
- Integrated Automatic Protection Switching (APS) OC-3/STM-1 for PSTN interface
- Standard control: MGCP, MEGACO
- Complete media gateway on a blade
- G.168 compliant echo cancellation
- Real-time fax over IP/T.38
- PSTN Signaling: CAS, ISDN PRI and SS7 layer 3 termination
- Tone detection and generation (MF, DTMF, RFC 2833)
- SIGTRAN IUA, M2UA, M3UA over SCTP
- Dual redundant 10/100/1000 Base-T interfaces, PICMG 2.16 compliant

# TP-6310

## Specifications

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|---|---|
| **Software Specifications** | |
| Capacity | 2016 independent digital voice, fax and data ports<br>Up to 1000 additional mediation channels without transcoding |
| Voice Compression | G.711, G.723.1, G.729A/B, G.726/G.727, AMR, MS-GSM, GSM-FR, iLBC<br>Wide Band coders including G.722 and AMR<br>Additional coders supported – contact AudioCodes for further information |
| Echo Cancellation | G.168 compliant 32, 64, 128 msec echo tail |
| Fax Relay and Termination | Real-time fax over IP/T.38 compliant, automatic fallback to G.711 and VBD for up to SG-3<br>Concurrent fax sessions on all the channels<br>Support for Fax Termination (Available with AudioCodes S/W based stack) |
| ASR - 3rd Party | Distributed Architecture - Media Stream over VoIP RTP |
| In-band/Out-band Signaling | Packet side or PSTN side, DTMF and tone detection and generation, CAS Relay, RFC 2833 |
| VoIP Standards Compliance | RTP/RTCP per RFC 3550/3551<br>DTMF over RTP per RFC 2833 |
| Control Protocols | AudioCodes' proprietary API - VoPLIB, SIP, MEGACO (H.248), MGCP (RFC 3435, RFC 3660) |
| Management Interfaces | • SNMP V2c: Standard MIB-2: system, interfaces, if-MIB, entity-MIB, RTP-MIB, DS1-MIB, snmpV2-MIB and AudioCodes' proprietary MIB<br>• On-board embedded secure Web Server<br>• SysLog |
| Operating System | • Windows™ 2000, XP, 2003 • Linux™[1] RH8, RH9, Debian, Enterprise<br>• Solaris™[1] 8,9 on Intel™/Sparc™ 32/64 |
| **Signaling** | |
| PSTN | **CAS** T1 robbed bit, MFC/R2 numerous country variants<br>**CCS** ISDN PRI: numerous country variants including ETSI EURO ISDN, ANSI NI2, DMS, 5ESS, Japan INS1500<br>**SS7** MTP2 and MTP3 link termination<br>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 10/100/1000 BASE-TX ports |
| Hot Swap | Full hot swap supported per PICMG 2.1 |
| Physical Interfaces | Form factor – 6U PICMG 2.0 single cPCI slot<br>TDM Interface – H.110 CT Bus<br>Telephony – Dual optical Replaceable LC Connector, APS protected<br>Ethernet – cPSB PICMG 2.16 on the backplane, Dual Optical Replaceable LC connectors on rear panel |
| Power | 75W |

*1 See Release note for specific OS releases supported*

## Applications

- Next Generation Switches
- IP Services Platforms
- Packet to Packet mediation (with or without transcoding)
- VoIP Access Gateways

## 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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