

## AudioCodes PCI TP-260/SIP Media Gateway

**Full function SIP Media Gateway in a unique PCI form factor**

**A**s SIP and VoIP continue their dramatic growth, application developers are challenged with delivering solutions that can be used in both SIP-enabled infrastructure and in existing legacy TDM equipment.

The TP-260/SIP is a full-function SIP Media Gateway in a unique PCI form-factor. Acting as a stand-alone gateway, the TP-260/SIP uses the Ethernet interface for all communications to/from the host computer and eliminates the PCI interface, avoiding the complexity of PCI drivers, operating system compatibility and H.100 cabling.



### **Typical Applications**

- IP-PBX
- Softswitches
- Contact Centers
- IVR Systems
- Messaging Systems
- Speech Applications
- Prepaid Calling
- Conferencing Systems
- ACDs/Predictive Dialers

### **Standard SIP Control**

By leveraging SIP, the TP-260/SIP is controlled using one of the fastest growing control protocols in the industry. Using IETF standard RFC 3261, developers can work in a standard environment that is widely accepted across the industry.

### **Operating System Independent**

The TP-260/SIP is configured and is controlled via the Ethernet interface. There is no dependency on the host computer operating system. This allows your developers to choose the appropriate operating system for your application without waiting or worrying about O/S driver compatibility. The TP-260/SIP is an excellent choice to provide PSTN interfaces, whether the application runs on a Windows, Linux, Solaris or virtually any other legacy operating system.

### **Proven SIP Interoperability**

With demonstrated SIP compatibility, the TP-260/SIP is interoperable with a wide range of existing SIP applications and SIP proxy server software. As SIP continues to evolve, AudioCodes' continued interoperability testing and software updates will enable smooth integration of our products in your SIP networks .

### **Reduce Integration Costs**

By eliminating the PCI driver interface, the TP-260/SIP avoids many of the lengthy and costly integration issues that traditional CTI boards have suffered.

### **Eliminate Engineering Bottlenecks**

By providing a quick "ready to plug-in" solution that can SIP enable legacy PSTN applications, your product launch isn't dependent on the exhaustive porting efforts by your engineering development staff. With the TP-260/SIP you can launch the SIP version of your product and begin to build momentum now.

### **Benefit from an Internal Gateway over an External Gateway**

There are a number of benefits to using a blade-level internal media gateway over using an external stand-alone gateway:

### One Box Solution

By integrating the TP-260/SIP into the application server, resellers, installers and end customers perceive the system as “one box”, making installation easier with less confusion. Packaging the gateway into the application server also eliminates “multi vendor confusion”, simplifying support.

### Increase Up-Sell Opportunities

By offering an integrated gateway as part of the application proposal, you give your salespeople an opportunity to up-sell the customer and capture more of the final system sale. If your sales people are currently referring their customers to other gateway vendors, they may be missing a good part of the overall opportunity.

### Reduce Cost

A blade level gateway reduces the final system cost because it shares space in the application server, eliminating its own dedicated power supply, chassis and cooling systems. Blade level gateways also are less expensive to maintain, avoiding a separate chassis-level support contract.

### Partnership Management

By building a one-box solution using OEM components, you have better control over the project partnerships.

## TP-260/SIP Advantages

- Isolates the PSTN connectivity, configuration and alarms away from the application (simplifying application development)
- Eliminates CTI boards (application developer can concentrate on 100% IP solutions going forward)
- Standard control interface (SIP) avoids working with proprietary or legacy APIs
- Independent of host computer operating system
- Quick time to market
- Proven SIP interoperability
- Reduced Integration Costs:
  - No drivers
  - No APIs
  - No H.100 cabling
- Eliminates engineering bottlenecks
- One box solution
- Eliminates confusing external gateways
- Eliminates competing gateway vendors from the end customer proposals
- Increases “up-sell” opportunities

**With a wide range of applications and deployment models, the TP-260/SIP offers a common solution to the challenge of connecting the legacy TDM world to the new SIP-based IP-PBXs and carriers. Whether you are just starting to port an application to SIP or have a sophisticated SIP solution ready for market, the TP-260/SIP is a valuable addition to your strategy.**

# Instant PSTN Interface for SIP Applications

Developers of SIP-based software applications face a challenge in leveraging their existing solutions for deployment in PSTN or PBX environments. Using the TP-260/SIP enables a nearly instant TDM interface by adding the card to existing SIP-based systems, as shown in the configuration below.

The TP-260/SIP doesn't use host operating system drivers or an API. The card can be inserted in virtually any full-size PCI slot in a wide range of servers, utilizing the PCI for power supply only.

The TP-260/SIP Media Gateway provides the required translation between SIP commands issued by the application and the PSTN signaling protocols. By using SIP as a control protocol to the gateway, the legacy TDM interface is isolated, leaving configuration, maintenance and alarms to the TP-260/SIP.

Developers struggle every day with finding the resources to maintain two code-bases (one for the PSTN-based CTI cards and one for SIP interfaces). A solution many developers have adopted is to focus their development efforts 100% on the SIP code-base, leaving the interface of legacy TDM equipment to the TP-260/SIP, acting as a SIP Media Gateway.



## Selected TP-260/SIP Technical Specifications

### Software Specifications

Capacity	1, 2, 4, 8 E1/T1 spans
Voice Compression	G.711, G.723.1, G.729 AB, G.726/G.727, Net Coder®, MS-GSM, GSM-FR Additional coders supported – contact AudioCodes for further information
Echo Cancellation	G.168-2002 compliant 32, 64 echo tail 128 msec tail with reduced channel capacity
Fax Relay	Real-time fax over IP/T.38 compliant
In-band/Out-band Signaling and DTMF	Packet side or PSTN side, DTMF and tone detection and generation, RFC 2833 compliant
VoIP Standards Compliance	RTP/RTCP per IETF RFC 390, DTMF over RTP per RFC 2833
Operating System	No PCI driver is needed
PSTN Protocol Termination	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

### Hardware Specifications

Ethernet	10/100 Base-T
Power	3.6A at 5 V with quad E1/T1 interface

# Fast SIP Enabling for TDM Applications

Developers face a challenge in figuring out how their existing TDM applications and hardware can be deployed in new SIP-based IP-PBX and IP-carrier infrastructure. Using the TP-260/SIP enables a nearly instant SIP interface by adding the card to existing TDM systems, as shown in the configuration below.

The TP-260/SIP doesn't use host operating system drivers or an API, utilizing the PCI for power supply only. The card can be inserted in virtually any full-size PCI slot in a wide range of servers. Once connected using the existing T1/E1 interfaces via a cross-over cable to the TP-260/SIP, the application can immediately support standard call control functionality in a SIP environment.

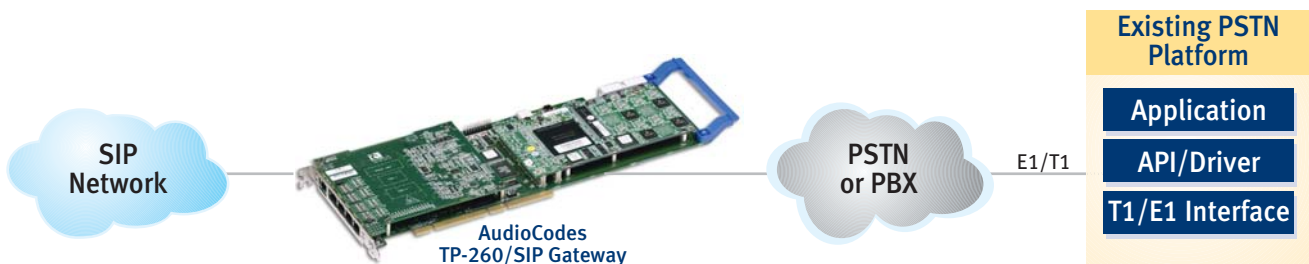
## Shortcut to the SIP World

This deployment model is important as many applications face significant porting or even rewriting efforts to achieve a native SIP capability. Using the TP-260/SIP as a bridge technology allows existing systems to quickly be viable in SIP installations, while the market is tested and a more permanent and cost optimized solution is developed.

## Making the impossible ... possible

In many cases, the original legacy equipment has to be reused in new SIP deployments and the original manufacturer has yet to introduce their SIP solution. The legacy equipment would require significant upgrades, or the original manufacturer would be no longer in business. In any case, by connecting the existing T1/E1 interfaces via a cross-over cable to the TP-260/SIP, the application can immediately be deployed in a wide range of SIP environments.

### SIP Enabling CTI Applications



## Selected TP-260/SIP Technical Specifications

### Software Specifications

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### Hardware Specifications

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### 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.

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