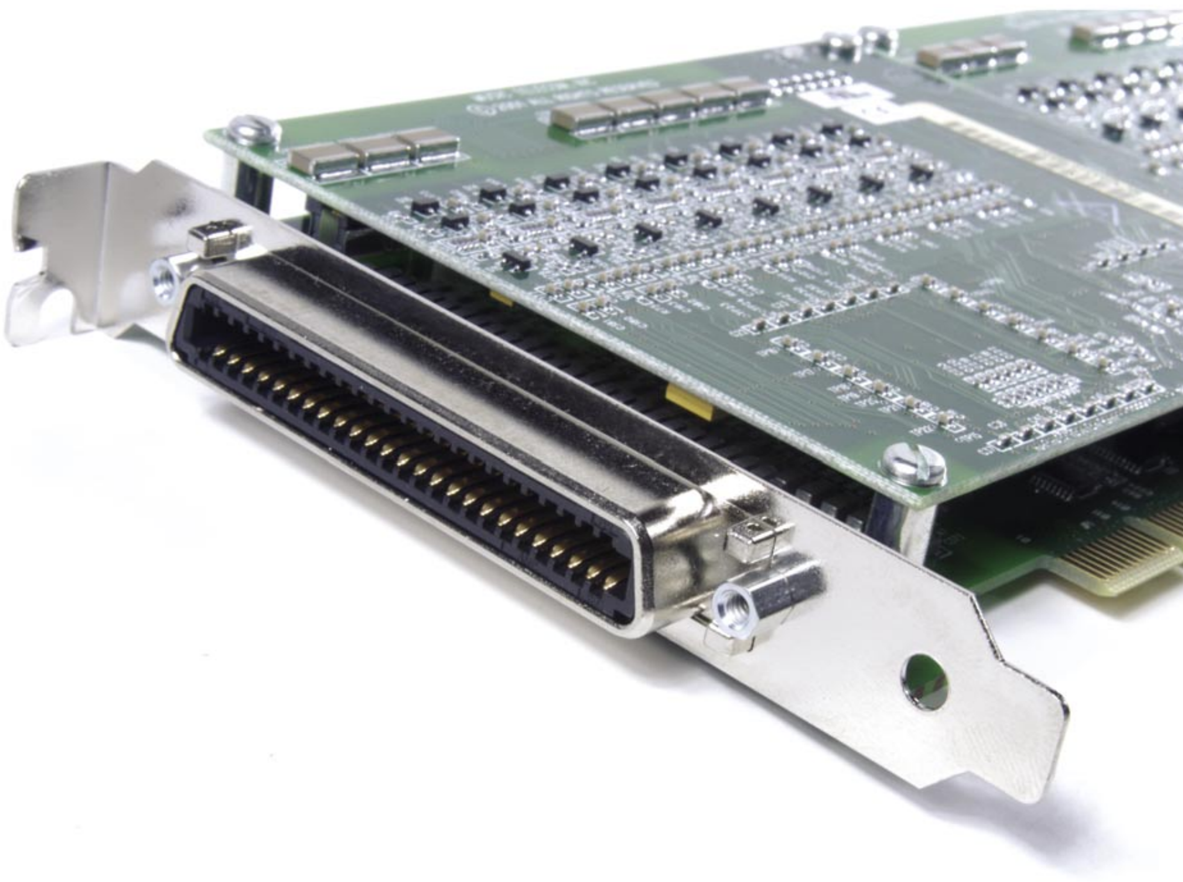
Ai-Logix

An AudioCodes Company



2005 Product Catalog

Ai-Logix Corporate Headquarters

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From the President's Desk:

It is my great pleasure to introduce the 2005 product catalog featuring our complete family of telephony components. New to this catalog is the VoIPerfect™ line of Voice over Packet (VOP) enabled telephony products. Together with our SmartWORKS™ product line, Ai-Logix now offers solutions to meet the needs of all major computer telephone segments.

At Ai-Logix, we are committed to building quality products, giving great service, and fostering long-lasting relationships. In fact, those are the guiding principles that have been an integral part of our success.

Working closely with our customers and listening to their needs has encouraged us to become a flexible company with versatile products. By partnering with Ai-Logix, you are doing more than just buying telephony cards; you are investing in the future of evolving technology in the CTI industry.

I am sure you will find everything you need to enhance your products in this catalog. We welcome your feedback and suggestions.

Thank you for your support of Ai-Logix and our products.

Sincerely,

Moshe Tal.

Moshe Tal
President & CEO, Ai-Logix, Inc.

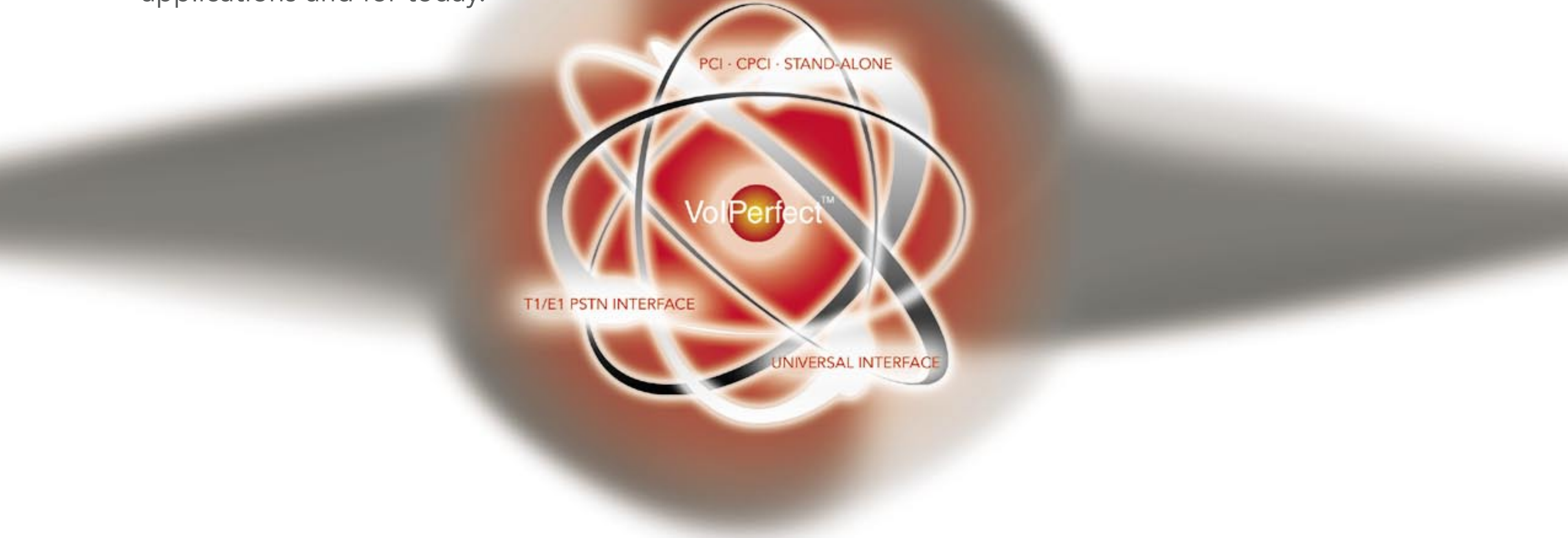
Table of Contents

VOIPERFECT™

Your Gateway to VoIP

What Is VoIPerfect™ ?

VoIPerfect™ is AudioCodes underlying, core media gateway architecture common to all of its products. It is designed to deliver the right mix of features to the right vertical market. VoIPerfect's unique mix of VoIP media processing capabilities combined with industry standard PSTN interfaces makes it an ideal choice for both next generation applications and for today.



Built from the AudioCodes VoP DSP and VoicePacketizer™ protocol stacks, the VoIPerfect™products share a common architecture and Application Programming Interface (API). This allows easy product migration from low-density analog solutions all the way to advanced high availability, high-density applications.

Features

· Scalable channel density

· Field-proven PSTN interface board

· Smooth migration to VoIP network, using software license only

· Concurrent toll

· 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 pacet streaming (RTP/RTCP) oer RFC 1889/1890

· MGCP, MEGACO and AudioCodes' proprietary TPNCP

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

· Automatic Speech Recognition (ASR)[1] and Text To Speech (TTS)[1]

A Complete VoIP Media Processing Solution

Combining many options on a single board, the IPM 260 can eliminate a number of separate special-function boards, resulting in reduced inventory, increased overall system density, reduced costs, and improved time-to-market.

Designed as a universal solution, all VoIPerfect™260 products are capable of delivering just the right mix of communication features for many markets. Like all members of the VoIPerfect™family, the 260 Series can be ordered with different software and interface options, enabling it to meet a wide range of market needs.

Universal solution

For traditional CTI applications, the 260 Series has the right mix of firmware-based media processing capabilities such as: record, playback, conferencing, voice coding, echo cancellation, fax processing, and call progress tones detection. In addition, the VoIPerfect™260 is capable of supporting advanced Voice Over Packet features like RTP packet streaming, voice activity detection, transcoding, and industry standard signaling formats. It is available with or without PSTN T1/E1 interfaces and can be configured with up to 8 T1/E1 spans.

Protect Customer Investment

The VoIPerfect™260 series is based on the VoIPerfect(TM) architecture, AudioCodes' underlying gateway technology for all of its products. As a member of the VoIPerfect™family, the 260 series shares the same API and feature set as the entire product line. This unique API 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 VoIPerfect™260 series is an ideal building block for scalable, reliable VoIP enabled media processing solutions. With the VoIPerfect™260 series' comprehensive features, customers can quickly design a wide range of solutions combining PSTN and VoIP networks.

Built on the heritage of the AudioCodes Voice over IP processing technology, the VoIPerfect™ 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.

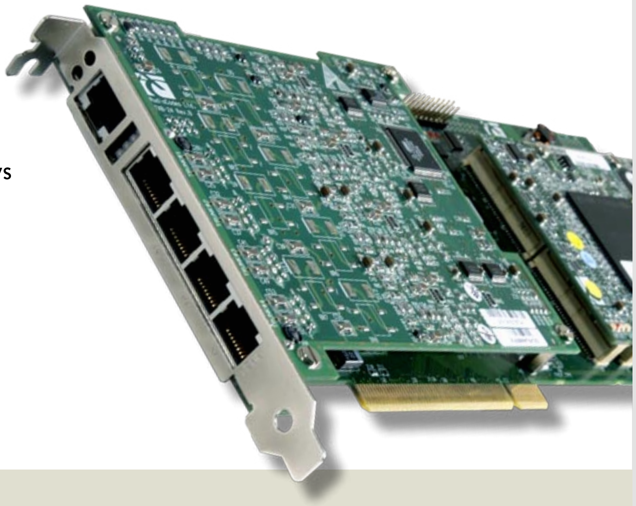
TP-260

E1/T1 VoIP Communication Gateway Board

Features

- Stand-alone SIP media gateway
- Low to high channel density (1, 2, 4, 8)
- Concurrent toll quality voice and fax support
- Wide range of PSTN signaling protocols
- Fast time-to-market
- Flexible and easy migration to VoIP networks
- Up to 256 independent voice/fax/data ports
- VoIP packet streaming (RTP/RTCP)
- Standard control: MGCP (RFC 2705), MEGACO (H.248)
- Real-time fax over IP/T.38
- On-board announcement memory
- Tone detection and generation (MF, DTMF, RFC 2833)
- PSTN Signaling: CAS, ISDN PRI, and SS7
- SIGTRAN IUA, M2UA, M3UA over SCTP
- Media Gateway on a blade mode
- MVIP SCbus and H.100 TDM interfaces
- Management Interfaces: SNMP, Web server
- On-board 10/100 Base-T Network interface
- Optional universal PCI version

The TP-260 PCI VoIP communication gateway board, based on AudioCodes' TPM-1100 PMC Modules, is an ideal solution for trunking gateways to the PSTN and integrated gateways for IP-PBXs and all-in-one communication servers. The TP-260 provides 256 ports for voice, fax or data implementing VoIP media gateway applications.



Deliver Feature-Rich Solutions

The TP-260 supports a broad selection of voice processing related algorithms, including G.711, G.723.1 and G.729A Vocoders, G.168-compliant echo cancellation, T.38 Real-time Fax over IP, as well as a wide selection of In-Band and Out-Band tone detection and generation. The TP-260 wide selection of TDM interfaces allows easy integration with other third party CTI boards. The E1/T1/J1 PSTN interface and wide range of supported telephony protocols provides TP-260 users a higher level of integration, saving backplane slot space, enabling higher density gateway platforms while reducing the costs per channel.

Universal solution

For traditional CTI applications, the 260 Series has the right mix of firmware-based media processing capabilities such as: record, playback, conferencing, voice coding, echo cancellation, fax processing, and call progress tones detection. In addition, the VoIPerfect™260 is capable of supporting advanced Voice Over Packet features like RTP packet streaming, voice activity detection, transcoding, and industry standard signaling formats. It is available with or without PSTN T1/E1 interfaces and can be configured with up to 8 T1/E1 spans.

Protect Customer Investment

The TP-260 is based on the VoIPerfect™ architecture, AudioCodes' underlying, best-of-breed core technology for all of its products. The TP-260 supports AudioCodes API, which enables software download, provisioning and control. It was designed to maintaining 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 low to high density and reduced costs, the TP-260 is an ideal building block for scalable, reliable VoIP solutions. With the TP-260's comprehensive feature set, customers can quickly design a wide range of solutions for smooth migration to VoIP networks.

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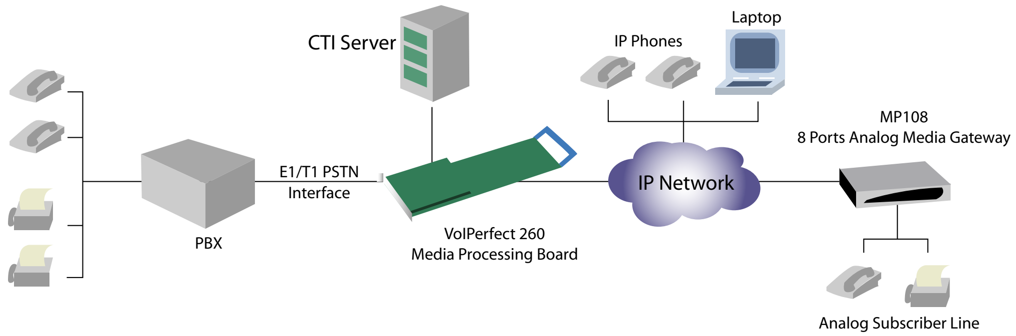
Application Diagram

Integration of VoIPerfect Products in NextGen Enviroment



SOFTWARE SPECIFICATIONS

CAPACITY

32, 64, 128 256 independent digital voice, fax and data ports

VOICE COMPRESSION

G.711, G.723.1, G.729A, G.726/G.727, Net Coder ®

Additional coders supported - contact AudioCodes for further information

ECHO CANCELLATION

G.168 compliant 32, 64 echo tail

128 msec tail available with reduced channel capacity

FAX RELAY

Real-time fax over IP/T.38 compliant, automatic fallback to G.711

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 RFC1889/1890

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

AudioCodes' proprietary VoIP API Library over IP (TPNCP) or PCI

MANAGEMENT INTERFACES

SNMP V2: Standard MIB-2, RTP MIB, Trunk MIB, AudioCodes proprietary MIB

On-Board Embedded Web Server

OPERATING SYSTEM

. WindowsT NT, 2000, XP . LinuxT . Solaris on SparcT/ IntelT

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 - MVOP, 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

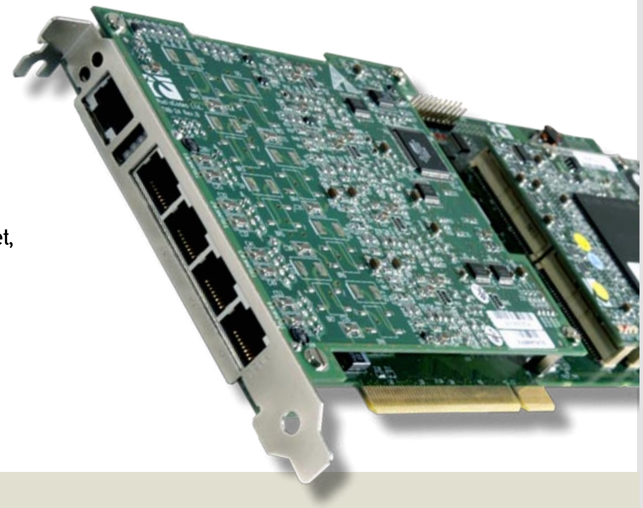
ATP-260

E1/T1 PCI PSTN Media Processing Board

Features

- Scalable channel density
- Field-proven PSTN interface board
- Smooth migration to VoIP using software license
- Concurrent toll quality voice and fax support
- All-in-one integrated board
- Based on AudioCodes' leading VoIP technology
- Up to 240 universal media processing ports
- Comprehensive IVR control
- Voice Record/Playback
- Real-time, multi-party conferencing
- Speech enabled application support*
- Interchangeable PSTN or TDM endpoints
- MVIP, SCbus and H100CT bus interface support
- Integrated software controlled E1/T1 interfaces
- Voice compression on a per channel base
- VoIP packet streaming: (RTP/RTCP) per RFC 1889/1890
- Universal PCI version

Built on the heritage of the AudioCodes Voice over IP processing technology, the Ardito™ 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 ATP-260 including: message record/playback, conferencing, voice coding, echo cancellation, fax processing, call progress tone detection and generation, as well as signaling protocols support including ISDN PRI, SS7 layer 3 termination (MTP2, MTP 3), CAS, and optional voice coding, VoIP functionality and SIGTRAN (M3UA, M2UA, IUA). All media processing, signaling and control protocols can be applied independently and simultaneously on all of the ATP-260 universal channels.

Universal solution

The Ardito™ 260 (ATP-260) is a complete media processing solution providing PSTN and optional IP interfaces required to build next generation applications for both today's and tomorrow's networks. By combining these capabilities on a single board, the ATP-260 enables developers to design a complete converged telephony solution that can be adapted according to exact customer requirements. The ATP-260 will enable customers to upgrade their current installations in the future by using software licenses only.

Protect Customer Investment

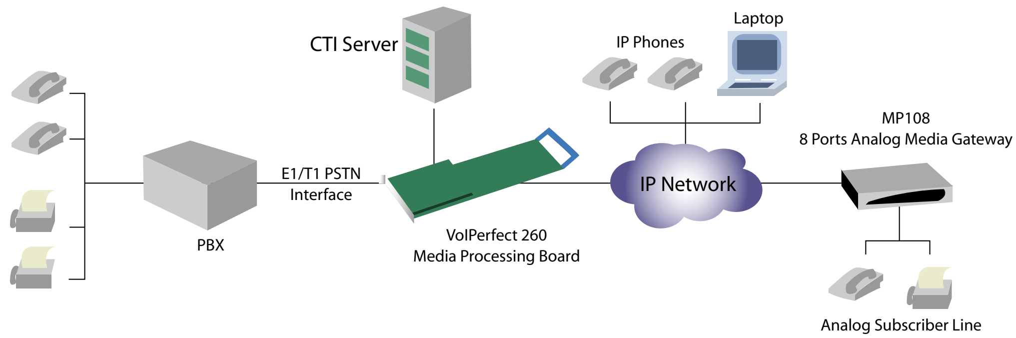
The ATP-260 is based on the VolPerfectTM architecture, AudioCodes' underlying, best-of-breed, core media gateway technology for all of its products. The ATP-260 supports AudioCodes' proprietary API, which enables handling PSTN, media processing, optional VoIP tasks and software download and provisioning. It was designed to maintain essential API backward compatibility of future releases 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, full flexibility and reduced costs, the ATP-260 is an ideal building block for scalable, reliable PSTN media processing solutions that can be upgraded to support VoIP networks. With the ATP-260's comprehensive feature set, customers can quickly design a wide range of solutions combining PSTN and VoIP networks that can be deployed gradually.

Application Diagram

Integration of VoIPerfect Products in NextGen Enviroment



SOFTWARE SPECIFICATIONS

CONFIGURATION

30, 60, 120, 240 universal ports (voice, fax), software licensed

SCALABILITY

Board capacity may be upgraded using software license, based on purchase of scalable hardware

VOICE/MESSAGING, RECORDING

Host-based record/play, WAV format (G.711, G.726, MS-GSM) LBR voice recording - by software license Playback speed control with pitch correction

ON-BOARD ANNOUNCEMENT STORAGE

Up to 1000 different prompts, 10 Mbyte RAM storage space Recorded prompts - 20 minutes of G.711, 200 minutes of G.723

CONFERENCING

Supports up to 240 ports of mixed IP, PSTN and TDM participants Maximum full-duplex parties per conference bridge: 64 endpoints Maximum simultaneous 3-way conferences per board: 40

ASR SUPPORT

Host-based and distributed architecture

VOICE BAND FEATURES

Echo Cancellation G.168 compliant with 32, 64 echo tail; 128 msec tail available with reduced capacity Gain Control Automatic (AGC) or programmable In-band Signaling DTMF tone detection and generation

MANAGEMENT INTERFACES

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

CONTROL

AudioCodes' proprietary API over PCI bus

OPERATING SYSTEM SUPPORT

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

SIGNALING

PSTN

CAS T1 robbed bit, MFCR2 numerous country variants CCS ISDN PRI: numerous country variants including ETSI EURO ISDN, ANSI N12, DMS, 5ESS, Japan INS1500 SS7 MTP2 and MTP3 termination

VoIP-enabled using a software license key on specific hardware

LICENSING POLICY

Software license can be purchased at anytime

Voice Processing

G.711, G.723.1, G.729A, G.726/G.727, NetCoder® Additional coders supported - contact AudioCodes for further information Trans-coding of G.711 RTP to any Low Bit Rate Coder RTP stream

IN-BAND/OUT-BAND SIGNALING

DTMF Packet side or PSTN side, RFC 2833

FAX RELAY

Real-time fax over IP/T.38 compliant, automatic fallback to G.711

SIGTRAN

IUA over SCTP per RFC 3057/2960 SS7 MTP2 link termination M2UA and M3UA over SCTP

CONTROL

AudioCodes' proprietary API over the network (TPNCP)

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

3.6 A 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

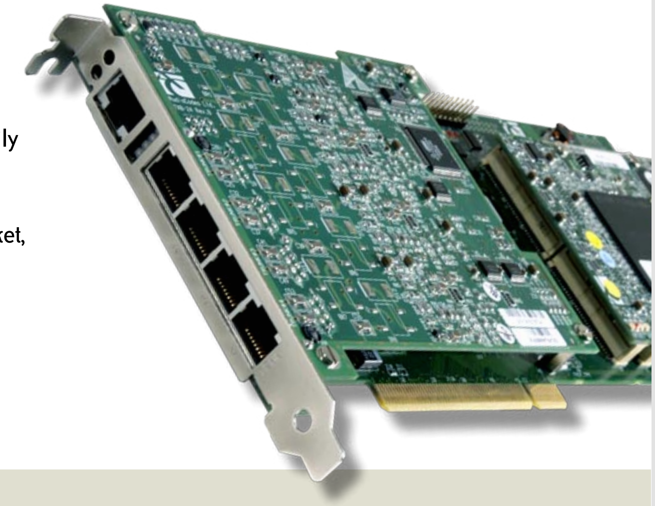
VOIPERFECT™
Your Gateway to VoIP

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

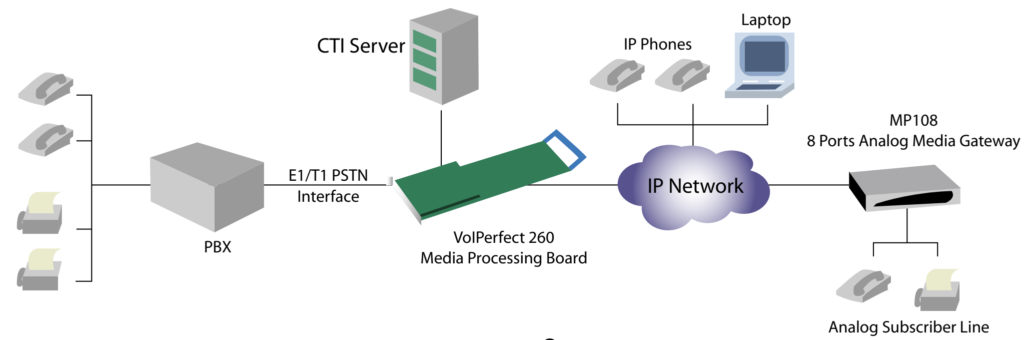
The IPM-260 is based on the VolPerfectTM architecture, AudioCodes' underlying, best-of-breed, core media gateway technology for all of its products. The IPM-260 supports AudioCodes' API, which enables software download, provisioning and control. It was designed to maintain essential API backward compatibility 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.

Application Diagram

Integration of VoIPerfect Products in NextGen Enviroment



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

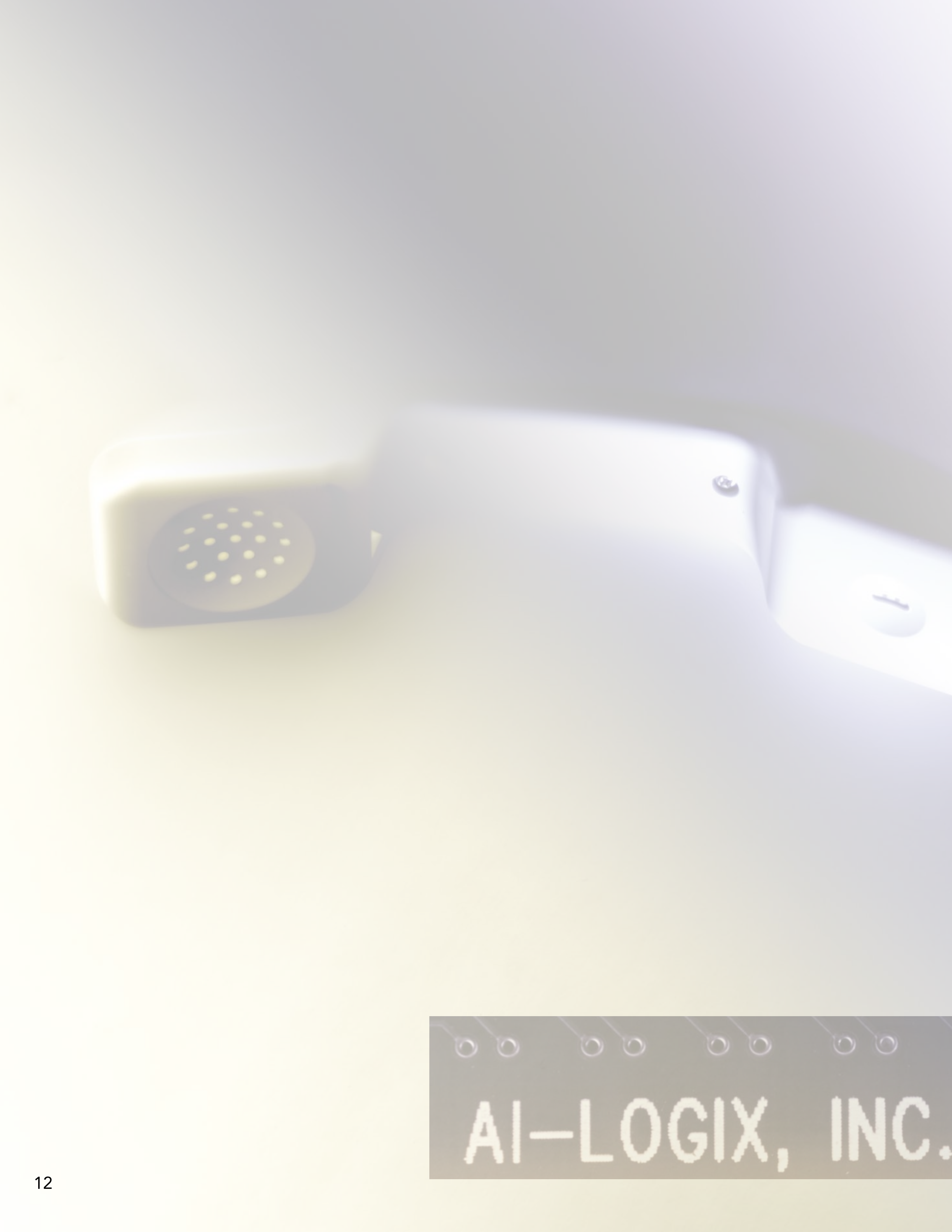
POWER

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


AI-LOGIX, INC.

SMARTWORKS™

INTELLIGENT CALL RECORDING

Features

Common native API across entire product line

All SmartWORKS Products have a 1:1 ratio for DSP voice resource channels to external ports

Selectable CODECS per channel

Caller ID/FSK/DTMF/MF

Advanced SDK

Operating Systems Support

- Windows 2000
- Windows XP
- Windows Server 2003
- Linux

Common DSP Architecture accross SmartWORKS product line

- Full duplex for simultaneous record and play
- Stereo recording – record far and near end of conversations separately
- Supports industry leading CODECS, including G.729a

Many Products, One Powerful API

SmartWORKS is a complete line of CTI components for the call recording market.

The SmartWORKS API provides full access to a large variety of features on the SmartWORKS hardware platform.

SmartWORKS uses a single C++ API, which gives customers the flexibility to develop their applications with one board, or with any combination of boards in the family. With multiple programming models supported, developers can tailor their applications to suit their specific needs.

The SmartWORKS™ Family

Passive Tap Recording Cards:
Analog POTS · Digital Extension · Digital Trunk · Analog Trunk/POTS

Trunk Interface:
Analog POTS · Digital Trunk

Voice Resource:
Voice and DSP Resource

SmartWORKS Product Family

Analog POTs *Universal*	Digital Extension *Passive*	Digital Trunk *Passive*	Digital Trunk *Terminate*	NetTAP *IP Recording*
LD101	NGX800	DP3209	DT3209	IPx128
LD409	NGX1600	DP6409	DT6409	IPr256
LD809	NGX2400			
LD1609				
LD2409				

Legend

Currently Available · Coming Soon

SmartWORKS™ LD Series

Next Generation Analog Passive/Active Telephony Card

Standard Features for SmartWORKS™ Family of Call Recording Products

The SmartWORKS™ API provides a common interface that controls the following call recording features:

· Media Control - CODECS

· Tone Detection / Generation

· CallerID/FSK/DTMF/MF Detection

· Activity / Silence Detectors

· Switching (H.100 and MVIP)

· Automatic Gain Control (AGC)

· Automatic Volume Control (AVC)

· Stereo Recording

· Echo Cancellation

· Call Progress Monitoring (CPM)

· Full-duplex Channels

· Media Streaming

· Live Monitoring

· Start/Stop Call Recording Triggers

· Beep tone generation for passive mode

Since 1991, Ai-Logix has designed boards used in interactive and passive telephony applications. With global support for all types of telephone and radio systems - analog, digital, and enterprise PBXs, Ai-Logix products have set a new world standard in telephony communications. A single API, combined with event driven reporting simplifies application development by providing one standard for all types of networks.

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Designed for analog networks, the SmartWORKS™ LD has both passive and terminate network interface capabilities. Featuring programmable voltage thresholds and loop reversal detection, the SmartWORKS™ LD is easily configured to accommodate variations across analog networks. This series offers This product is offered in 4, 8, 16 and 24 port versions, suitable for small to large offices and call centers.

Key Features and Benefits

4-24 Port Telephony Cards

Offers low to high density boards that are ideal for any analog environment.

On Demand Voltage Detection

Voltage values are reported with standard SmartWORKS™ API events to simplify application development.

Programmable Voltage Thresholds

Control voltage detection event reporting with ease by adjusting the board to the local analog environment.

Detects Polarity Reversal

Adapts to environments where Tip and Ring are reversed.

Minimum 18k Ohm Impedance

High impedance receivers record both sides of a call without interrupting service.

CODEC Support

SmartWORKS™ offers a large selection of voice CODECS (including G.723.1, G.729A and MS GSM)

Tap Environment

The LD series accomodates low to high density environments with 4, 8, 16, or 24 port boards. The SmartWORKS™ API supports a total of 512 channels per system. The tapping point can be anywhere on an analog line: between Central Office and PBX, Central Office and phones, or PBX and phones.

Terminate Environment

The LD series can be used to initiate as well as terminate calls. When configured as an interactive resource, phone lines can directly connect to and terminate on the LD boards. Standard ring detection is available.

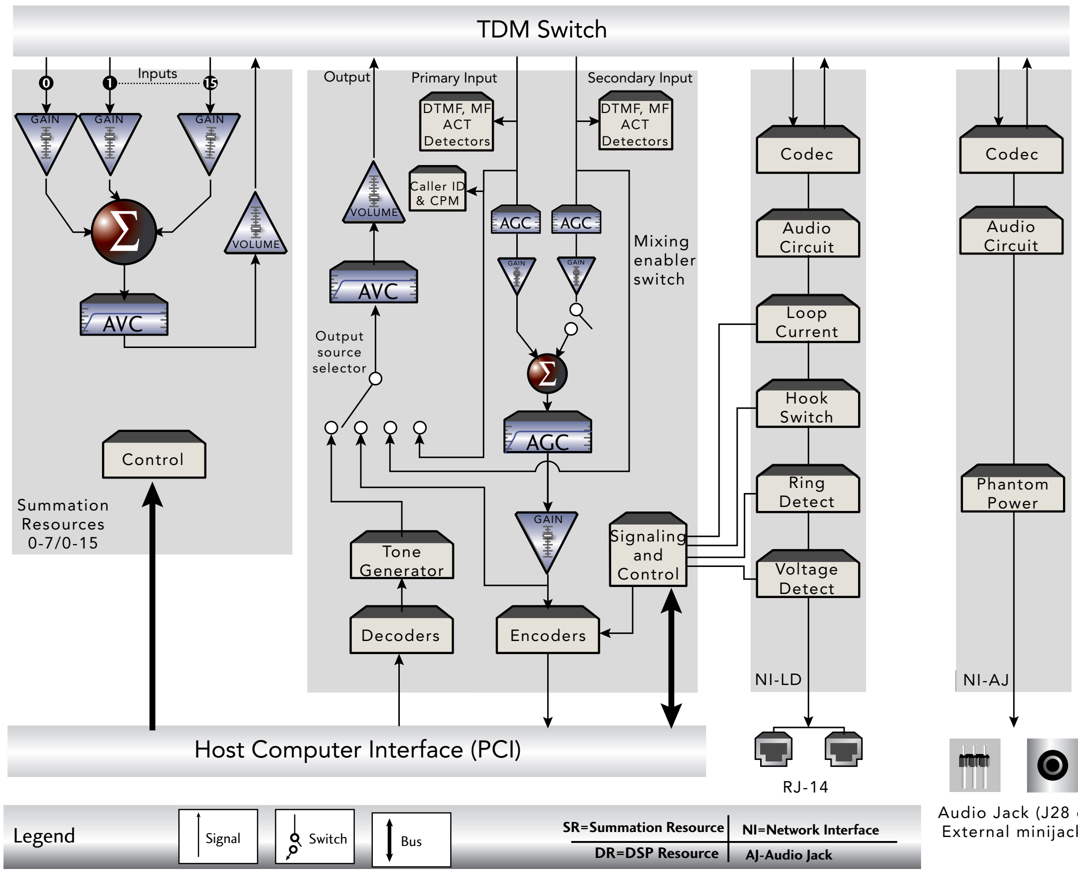
World-Wide Analog Support

The SmartWORKS™ LD supports passive call recording on ground start and loop start analog networks. It has line terminating capabilities for loop start environments. Features such as programmable voltage thresholds, voltage detection, and polarity reversal are managed through the common SmartWORKS™ API. As a result, the SmartWORKS™ LD easily adapts to variations found on analog systems throughout the world.

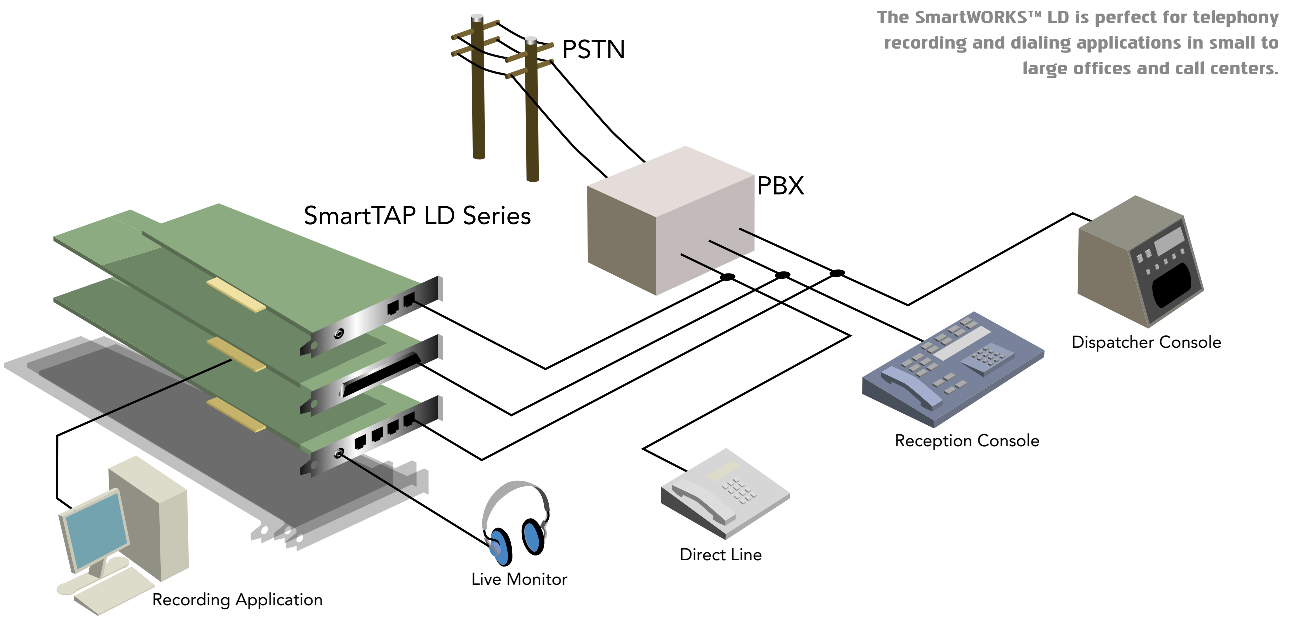
Built in Performance Monitoring

Built in voltage detection allows SmartWORKS™ LD to distinguish a disruption of service if a cable is damaged or disconnected. This feature is unique in the industry and only available on the LD series.

LD Logical Card Model



LD Application Model



The SmartWORKS™ LD is perfect for telephony recording and dialing applications in small to large offices and call centers.

SMARTWORKS™
INTELLIGENT CALL RECORDING

Technical Specs

LD SERIES

Product Specifications · SmartWORKS™ LD

Hardware System Requirements

Pentium 4 or equivalent · 2 GHz or better

PCI motherboard or passive backplane with 3.3V power supply, PCI 2.2 bus

Operating Systems

Windows 2000 · Windows XP

Windows 2003 32 bit· Linux (Call for variant details)

Technical Specifications

Max boards per system:	Any combination up to 512 ports
Max ports per system:	Up to 512
Resource Sharing Bus:	H.100 (409H, 809, 1609, and 2409 only)

Environmental Conditions

Operating Temperature:	0C to +60C
Storage Temperature:	-20C to +85C
Humidity:	8% to 80% non-condensing
Storage humidity:	8% to 80% non-condensing

Physical Characteristics

Form Factor:	Full or Half-size PCI card

Telephony Interface

Signal/Noise ratio:	35dB referenced to -15dBm
Idle channel noise:	Less then 20dBrnc
Crosstalk coupling:	Less then -70 dB (0dBm, 1004Hz)
Frequency response:	300Hz to 3400Hz +/-3dB
Ring detection:	30Vrms (min), 16 to 68Hz
Ringer Equivalence Number:	< 0.5
Echo return loss:	28 dB +/- 3dB @1400Hz
External Connector:	RJ-14 (LD409, 409H, 809) OR RJ-21 (LD1609, 2409)

Telephony Interface (Passive Mode)

Trunk Type:	Loop Start/Ground Start
Trunk Interface:	High Impedance (Z)
AC Impedance:	18 kOhms

Voltage Detection: Two software programmable thresholds

Range:	-61V to + 61V
Accuracy	+/- 2V

Telephony Interface (Terminate Mode)

Trunk Type:	Loop Start
AC Impedance:	Software Selectible FCC, EU, China, Australia
Loop Detection:	Off Hook: 8mA (max) *LD409, LD409H, LD809* On Hook: 6mA (min) *LD409, LD409H, LD809* OFF Hook: 11mA (max) *LD1609, LD2409* On Hook: 9mA (min) *LD1609, LD2409*

Telephony Connectors

LD409:	RJ-14
LD409H:	RJ-14
LD809:	RJ-14
LD1609:	RJ-21x
LD2409:	RJ-21x

SDK

Ai-Logix Native SmartWORKS™ API

SmartControl (Control Panel)

SmartVIEW (Card functionality test application)

Host Interface

Bus Compatibility:	Complies with PCISIG Bus Specifications, Rev. 2.2
Bus Speed:	33 MHz
Bus Mode:	32 bit bus master/target

Analog Jack

Audio Connector:	3-pin 0.1" ctr header (LD409, 409H, 809) -OR- 3.5mm (LD1609, 2409) Male Stereo Plug (1609 & 2409 only)
Output impedance:	300Ohms
Input impedance:	33KOhms
Mic bias:	+5VDC @ 4.7KOhms
Input gain:	+9dB
Output gain:	0 db @ 300Ohms
Full scale input:	370 mVRMS
Full scale output:	1.1 mVRMS open circuit

Audio Signal

Receive range:	-68 dBm to + 3 dBm
Input gain control:	+24 to -50 dB
Silence Detection:	Programmable from API
Transmit volume control:	+24 to -50 dB to H.100
Automatic Gain Control (AGC):	Programmable from API
Automatic Volume Control (AVC):	Programmable from API
Activity Detection:	Programmable from API
Frequency Response:	300 - 3400 Hz (+/- 3dB)

DTMF Tone Detection

DTMF digits:	0 - 9, *, #, A, B, C, D
Dynamic range:	-38 dBm to 0 dBm
Minimum tone detection:	40 ms / programmable
Interdigit timing:	40 ms min.
Acceptable twist:	Per LSSGR sec. 6, 8 dB forward, 4 dB reverse
Frequency variation:	Accept all +/- 1.5%, reject all +/-2.5%
Noise tolerance:	Per LSSGR sec. 6
Talk off:	Bellcore TR-TSY-000762

MF Detection

MF Detection	R1 & R2
R1 digits:	Per Q.151

Call Progress Monitoring (Terminate Mode)

Number of programmable tones	20
Number of bandpass filters	10
Number of filters per tone	1,2 or 3
Number of cycles	0 to 255
SIT tones	Yes, programmable frequencies and duration
Answering Machine Detection	Yes

Voice Processing

Caller ID	V.23 & Bell 202
DTMF Detector	Primary & Secondary channel

Echo Cancellation (Terminate Mode)

Input Dynamic Range	G.165 compliant
Double-talk detection	G.165 compliant
End path delay	8ms

Tone Dialing (Terminate Mode)

DTMF digits	0 – 9, *, #, A, B, C, D
Frequency variation	Less then 1 Hz
Rate	API Programmable
Duration	API Programmable

Safety and Certifications (Pending)

Telecom:	DOC
Emissions:	FCC Part 15 class A · EN 55022
Immunity:	EN 55024
Safety:	EN 60950
Estimated MTBF:	250,000 hours per Bellcore Method I

Ports

LD409	4 ports, no H.100
LD409H	4 ports
LD809	8 ports
LD1609	16 ports
LD2409	24 ports

Audio Digitizing (Encoding & Decoding)

5.3 Kb/s	G.723.1
6.3 Kb/s	G.723.1
8 Kb/s:	G.729A
13 Kb/s:	GSM 6.10, Microsoft GSM
16 Kb/s:	G.726
24 Kb/s:	G.726, OKI
32 Kb/s:	G.726, OKI
40 Kb/s:	G.726
64 Kb/s:	μ-law or A-law per G.711, 8 bit linear PCM (signed & unsigned)
96 Kb/s	6 Khz 16 bit linear PCM(signed)
128 Kb/s:	16 bit linear PCM (signed & unsigned)
Wave file formats:	Microsoft GSM, Linear signed 8 & 16-bit PCM
Digitization selection:	Programmable per channel, independent for encode and decode

Power Requirements (4 or 8 channel)

+ 3.3 VDC:	1.0 A
+5 VDC:	n/a
-12 VDC:	n/a
+12 VDC:	100 mA
Watts (Max)	4.5W

Power Requirements (16 channel)

+ 3.3 VDC:	1.3 A
+5 VDC:	n/a
-12 VDC:	n/a
+12 VDC:	200 mA
Watts (Max)	6.7W

Power Requirements (24 channel)

+ 3.3 VDC:	1.5 A
+5 VDC:	n/a
-12 VDC:	n/a
+12 VDC:	220 mA
Watts (Max)	7.6W

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SmartWORKS™ LD 101 - Station Port

Next Generation Analog Passive/Active Telephony Card

Standard Features for SmartWORKS™ Family of Call Recording Products

The SmartWORKS™ API provides a common interface that controls the following call recording features:

· Media Control - CODECS

· Tone Detection / Generation

· CallerID/FSK/DTMF/MF Detection

· Activity / Silence Detectors

· Switching (H.100 and MVIP)

· Automatic Gain Control (AGC)

· Automatic Volume Control (AVC)

· Stereo Recording

· Echo Cancellation

· Call Progress Monitoring (CPM)

· Full-duplex Channels

· Media Streaming

· Live Monitoring

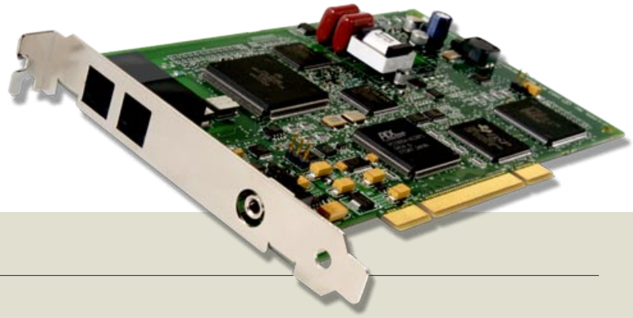
· Start/Stop Call Recording Triggers

· Beep tone generation for passive mode

· Beep tone generation for passive mode

Since 1991, Ai-Logix has designed boards used in interactive and passive telephony applications. With global support for all types of telephone and radio systems - analog, digital, and enterprise PBXs, Ai-Logix products have set a new world standard in telephony communications. A single API, combined with event driven reporting simplifies application development by providing one standard for all types of networks.

The SmartWORKS™ LD 101 is a 2-line card designed for analog environments requiring one trunk line and one station port. Typical applications include IVR, and outbound notification. The SmartWORKS™ LD 101 is cost efficient, which makes it an ideal choice for application prototyping.



Key Features and Benefits

Glare Protection

Automatically tests the line's voltage before the channel is opened, protecting the station port from damage.

On Demand Voltage Detection

Voltage values are reported on both lines with standard SmartWORKS™ API events to ease application development.

Programmable Voltage Thresholds

Control voltage detection event reporting with ease by adjusting the board to the local analog environment.

Detects Polarity Reversal

Adapts to environments where Tip and Ring are reversed.

CODEC Support

SmartWORKS™ offers a large selection of voice CODECS.(including G.723.1, G.729A and MS GSM)

Analog Station Port

The LD101's station port provides real-time direct access to the board for the purpose of storing and retrieving voice information from a local POTS telephone.t The station port and trunk line also maintain separate DSP resources allowing simultaneous play/record control on both lines. The FXS (foreign exchange-station) feature is used to connect to local phone sets.

World-Wide Analog Support

The SmartWORKS™ LD 101 supports passive call recording on Ground Start and Loop Start analog networks. It has line terminating capabilities for loop start environments. Features such as programmable voltage thresholds, voltage detection, and polarity reversal are managed through our common SmartWORKS™ API. As a result, the SmartWORKS™ LD 101 easily adapts to variations found on analog systems throughout the world.

Built in Performance Monitoring

With built in voltage detection, service disruptions are quickly detected if a cable is damaged or disconnected. This feature is currenlty not available on other SmartWORKS™ analog cards.

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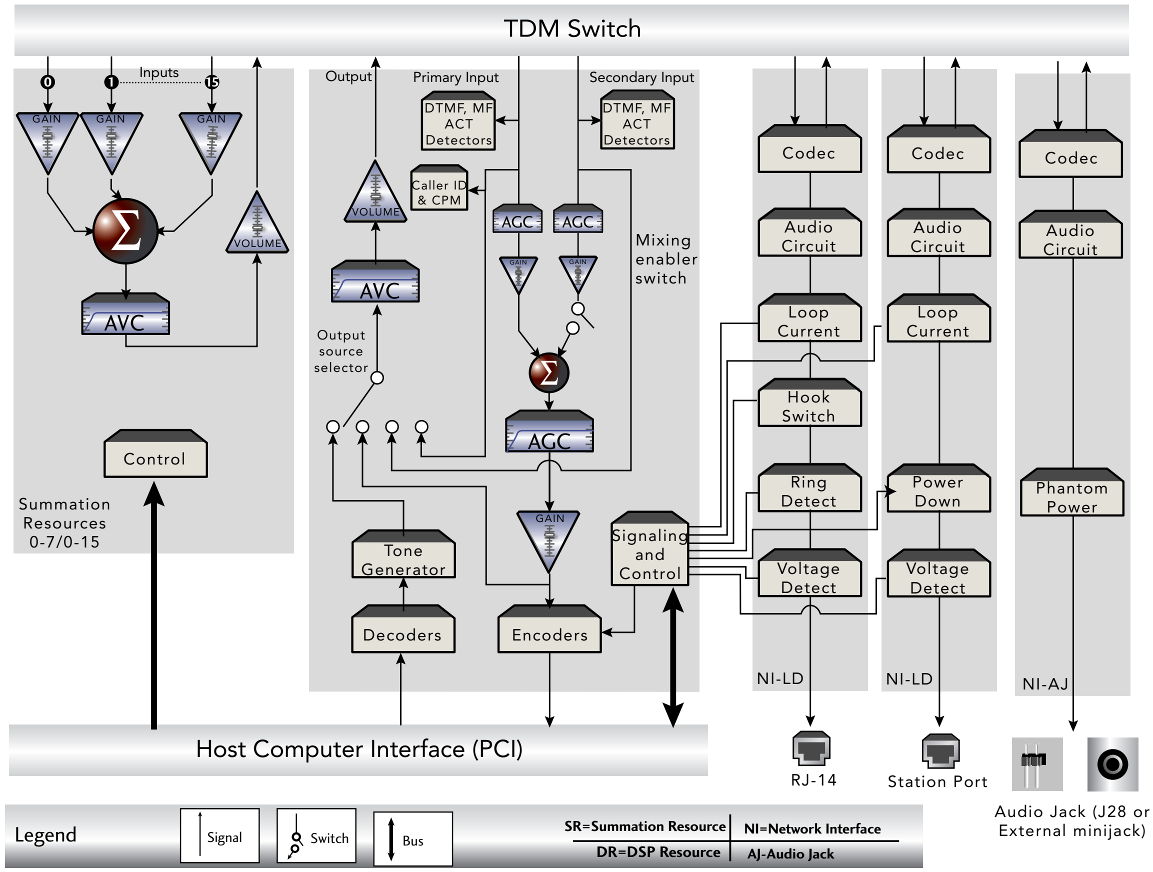
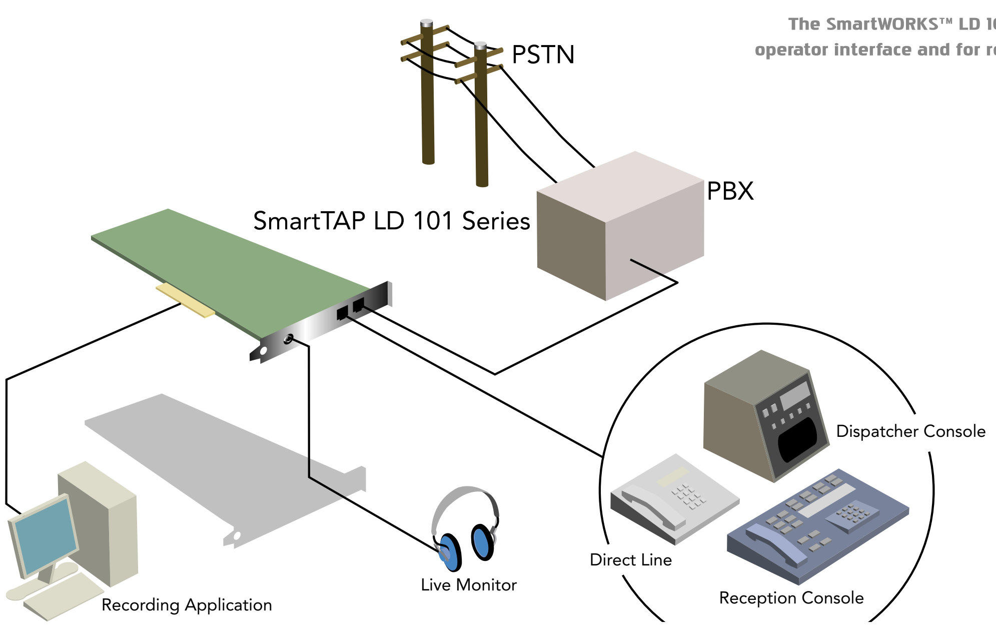
LD 101 Logical Card Model



LD 101 Application Model



The SmartWORKS™ LD 101 is perfect as an operator interface and for recording prompts.

Hardware System Requirements

Pentium 4 or equivalent · 2 GHz or better

PCI motherboard or passive backplane with 3.3V power supply, PCI 2.2 bus

Operating Systems

Windows 2000 · Windows XP

Windows 2003 32 bit· Linux (Call for variant details)

Technical Specifications

Max boards per system:	Any combination up to 512 ports
Max ports per system:	Up to 512
Boards Status:	On-board LEDs
Clocking:	Master/Slave

Environmental Conditions

Operating Temperature:	0C to +60C
Storage Temperature:	-20C to +85C
Humidity:	8% to 80% non-condensing
Storage humidity:	8% to 80% non-condensing

Physical Characteristics

Form Factor:	Full or Half-size PCI card

Telephony Interface

Signal/Noise ratio:	35dB referenced to -15dBm
Idle channel noise:	Less then 20dBrnc
Crosstalk coupling:	Less then -70 dB (0dBm, 1004Hz)
Frequency response:	300Hz to 3400Hz +/-3dB
Ring detection:	30Vrms (min), 16 to 68Hz
Ringer Equivalence Number:	< 0.5
Echo return loss:	28 dB +/- 3dB @1400Hz
External Connector:	RJ-14

Telephony Interface (Passive Mode)

Trunk Type:	Loop Start/Ground Start
Trunk Interface:	High Impedance (Z)
AC Impedance:	18 kOhms
Voltage Detection: Thresholds	Two software programmable
Range:	-61V to + 61V
Accuracy	+/- 2V

Telephony Interface (Terminate Mode)

Trunk Type:	Loop Start
AC Impedance:	Software Selectible FCC, EU, China, Australia
Loop Detection:	Off Hook RJ-14: 8mA (max) On Hook RJ-14: 6mA (min)

Station Port

Loop Current:	25 mA
Tip / Ring Voltage:	-24 Vdc @ on_hook
Input Gain:	0 dB +/- 0.5 dB
Output Gain:	-3 dB +/- 0.5 dB
Loop Detection:	off_hook 7mA(max) on_hook 5 mA(min)
Noise tolerance:	Per LSSGR sec. 6
Talk off:	Bellcore TR-TSY-000762
Ring Voltage	NONE*

** The LD101 does not provide FXS ring down feature*

MF Detection

MF Detection.	R1 & R2
R1 digits:	Per Q.151

SDK

Ai-Logix Native SmartWORKS™ API
SmartControl (Control Panel)
SmartVIEW (Card functionality test application)

Host Interface

Bus Compatibility:	Complies with PCISIG Bus Specifications, Rev. 2.2
Bus Speed:	33 MHz
Bus Mode:	32 bit bus master/target

Power Requirements (8 channel)

+ 3.3 VDC:	1.0 A
+5 VDC:	n/a
-12 VDC:	n/a
+12 VDC:	100 mA

Analog Jack

Audio Connector:	3.5mm Male Stereo Plug
Output impedance:	300Ohms
Input impedance:	33KOhms
Mic bias:	+5VDC @ 4.7KOhms
Input gain:	+9dB
Output gain:	0 db @ 300Ohms
Full scale input:	370 mVRMS
Full scale output:	1.1 mVRMS open circuit

Audio Signal

Receive range:	-68 dBm to + 3 dBm
Input gain control:	+24 to -50 dB
Silence Detection:	Programmable from API
Transmit volume control:	+24 to -50 dB to H.100
Automatic Gain Control (AGC):	Programmable from API
Automatic Volume Control (AVC):	Programmable from API
Activity Detection:	Programmable from API
Frequency Response:	300 - 3400 Hz (+/- 3dB)

DTMF Tone Detection

DTMF digits:	0 - 9, *, #, A, B, C, D
Dynamic range:	-38 dBm to 0 dBm
Minimum tone detection:	40 ms / programmable
Interdigit timing:	40 ms min.
Acceptable twist:	Per LSSGR sec. 6, 8 dB forward, 4 dB reverse
Frequency variation:	Accept all +/- 1.5%, reject all +/-2.5%
Noise tolerance:	Per LSSGR sec. 6
Talk off:	Bellcore TR-TSY-000762

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Audio Digitizing (Encoding & Decoding)

5.3 Kb/s	G.723.1
6.3 Kb/s	G.723.1
8 Kb/s:	G.729A
13 Kb/s:	GSM 6.10, Microsoft GSM
16 Kb/s:	G.726
24 Kb/s:	G.726, OKI
32 Kb/s:	G.726, OKI
40 Kb/s:	G.726
64 Kb/s:	μ-law or A-law per G.711, 8 bit linear PCM (signed & unsigned)
96 Kb/s	6 Khz 16 bit linear PCM(signed)
128 Kb/s:	16 bit linear PCM (signed & unsigned)
Wave file formats:	Microsoft GSM, Linear signed 8 & 16-bit PCM
Digitization selection:	Programmable per channel, independent for encode and decode

Call Progress Monitoring(Terminate Mode)

Number of programmable tones	20
Number of bandpass filters	10
Number of filters per tone	1,2 or 3
Number of cycles	0 to 255
SIT tones	Yes, programmable frequencies and duration
Answering Machine Detection	Yes

Voice Processing

Caller ID	V.23 & Bell 202
DTMF Detector	Primary & Secondary channel

Echo Cancellation(Terminate Mode)

Input Dynamic Range	G.165 compliant
Double-talk detection	G.165 compliant
End path delay	8ms

Tone Dialing(Terminate Mode)

DTMF digits	0 – 9, *, #, A, B, C, D
Frequency variation	Less then 1 Hz
Rate	API Programmable
Duration	API Programmable

Safety and Certifications(Pending)

Telecom:	DOC
Emissions:	FCC Part 15 class A · EN 55022
Immunity:	EN 55024
Safety:	EN 60950
Estimated MTBF:	250,000 hours per Bellcore Method I

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Standard Features for SmartWORKS™ Family of Call Recording Products

The SmartWORKS™ API provides a common interface that controls the following call recording features:

· Media Control - CODECS

· Tone Detection / Generation

· CallerID/FSK/DTMF/MF Detection

· Activity / Silence Detectors

· Switching (H.100 and MVIP)

· Automatic Gain Control (AGC)

· Automatic Volume Control (AVC)

· Stereo Recording

· Echo Cancellation

· Call Progress Monitoring (CPM)

· Full-duplex Channels

· Media Streaming

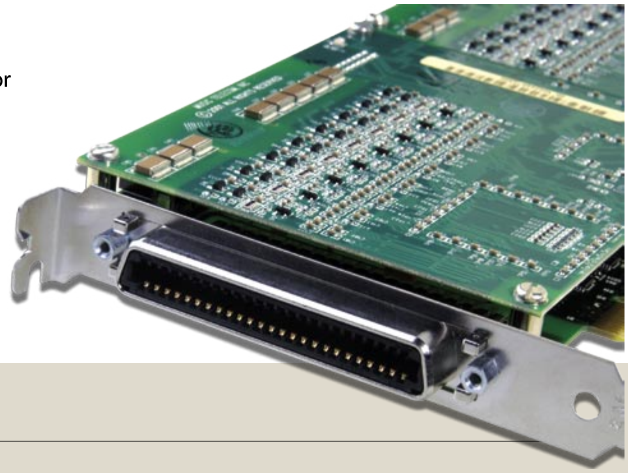
· Live Monitoring

· Start/Stop Call Recording Triggers

· Beep tone generation for passive mode

Since 1991, Ai-Logix has designed boards used in interactive and passive telephony applications. With global support for all types of telephone and radio systems - analog, digital, and enterprise PBXs, Ai-Logix products have set a new world standard in telephony communications. A single API, combined with event driven reporting simplifies application development by providing one standard for all types of networks.

The SmartWORKS™ NGX is an all-in-one resource for logging behind a PBX. Every key pressed, call taken, and telephone action performed by an agent is automatically decoded and sent to the recording application. A powerful set of features, combined with PBX integration, makes the NGX a true single slot solution for call logging application providers.



Key Features and Benefits

Multiple PBX support

A single board interfaces with a majority of industry leading PBXs to simplify the design of global call recording applications.

Firmware Upgraded

A simple firmware upgrade allows the NGX to adapt to different PBX environments.

Wide Spectrum of Trigger Events

Initiate and terminate recordings based on voice activity, raw D-channel, or Call Progress Monitoring (CPM) events.

Summation

Monitors up to 24 channels in real-time with on-board audio jack resources.

CODEC Support

SmartWORKS™ offers a large selection of voice CODECS.(including G.723.1, G.729A and MS GSM)

Tap Environment

The NGX is designed for tapping behind a proprietary PBX. Residing between the PBX and agent phones, the SmartWORKS™ NGX's high impedance receivers record both sides of a call without interrupting service. The NGX is available in 8,16,and 24 port configurations. The SmartWORKS™ API supports a total of 512 channels per system. As a result, the SmartWORKS™ NGX is ideal for low to high-density environments.

Extensive PBX Support

Designed with international deployment in mind, the SmartWORKS™ NGX taps 2-wire, 4-wire, BRI and full duplex PBX's. The list of PBXs the NGX supports is constantly growing. Contact your Ai-Logix sales representative for more information.

Built in Performance Monitoring

The SmartWORKS™ API provides framer alarms and network statistics to pass easily into performance monitoring applications. Event driven framer alarms are generated with a loss of signal condition. Network statistics are available for both sides of the conversation, incoming and outgoing. Statistics such as synchronization errors, line amplitude, noise or clipping are available via a simple API function call.

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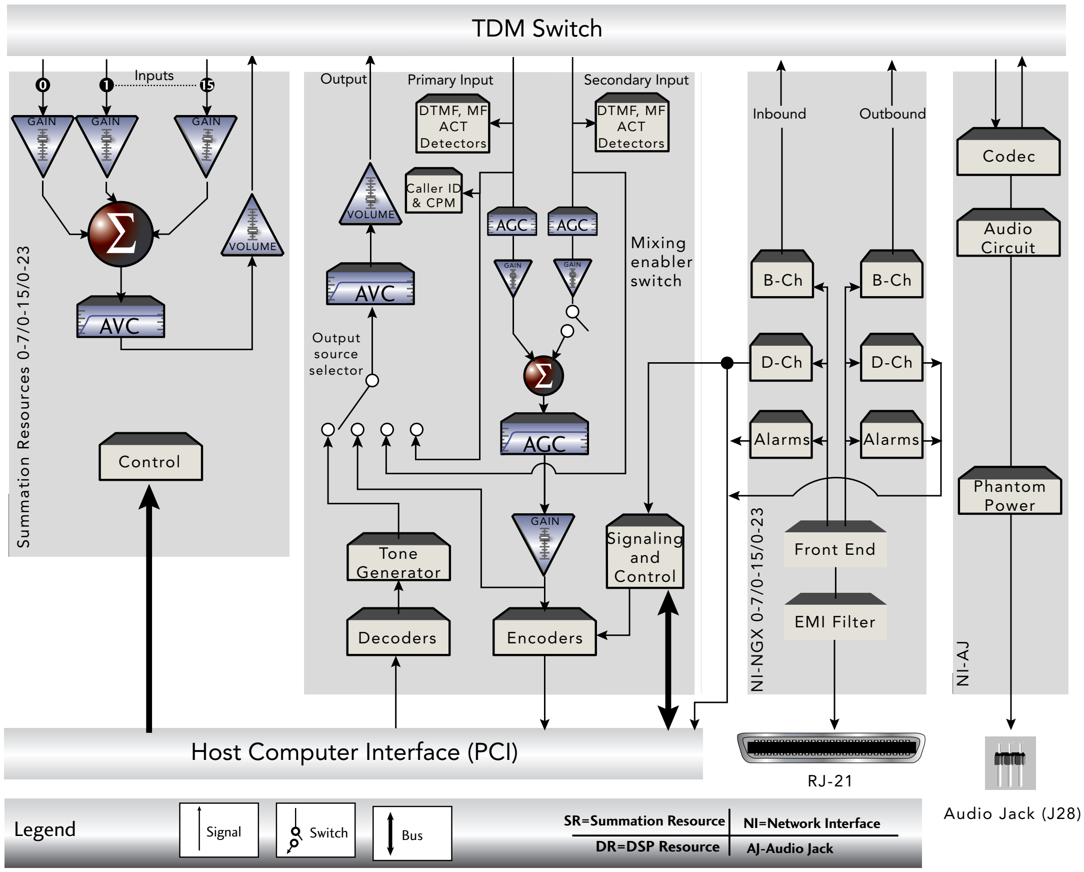
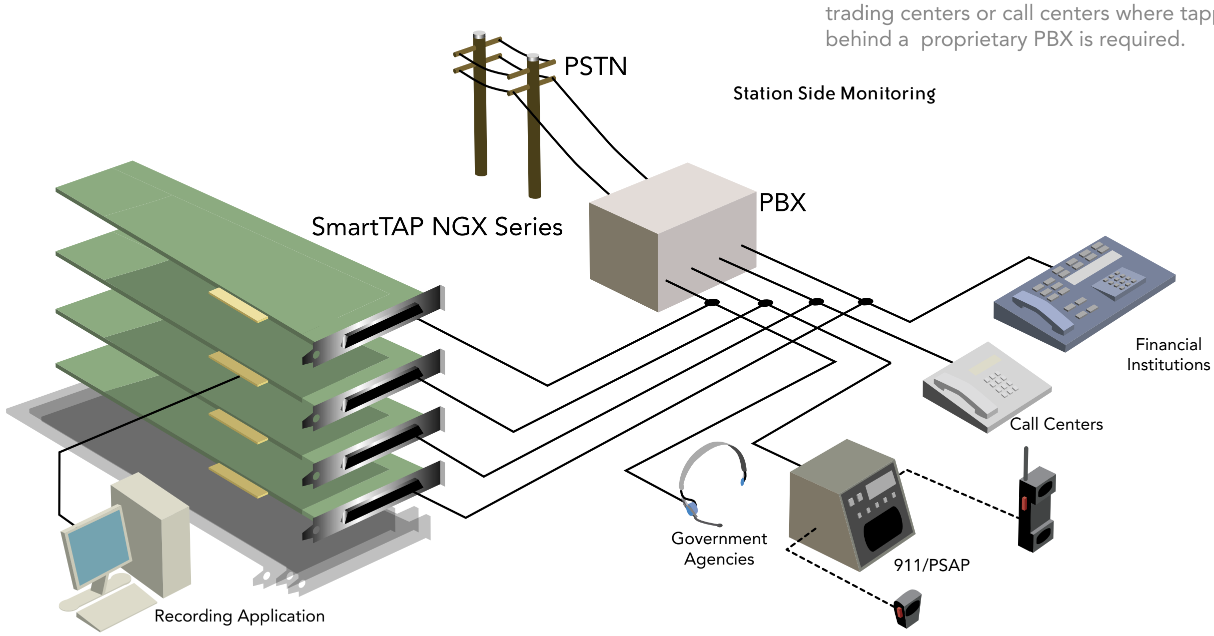
NGX Logical Card Model



NGX Application Model

Applied Use: The SmartWORKS™ NGX is perfectly suited for information centers, financial trading centers or call centers where tapping behind a proprietary PBX is required.



SMARTWORKS™
INTELLIGENT CALL RECORDING

Technical Specs

NGX Series

Product Specifications · SmartWORKS™ NGX

Hardware System Requirements

Pentium 4 or equivalent · 2 GHz or better

PCI motherboard or passive backplane with 3.3V power supply, PCI 2.2 bus

Operating Systems

Windows 2000 · Windows XP

Windows 2003 32 bit· Linux (Call for variant details)

Technical Specifications

Max boards per system:	Any combination up to 512 ports
Max ports per system:	Up to 512
Resource Sharing Bus:	MVIP or H.100
Boards Status:	On-board LEDs
Clocking:	Master/Slave

Environmental Conditions

Operating Temperature:	0C to +60C
Storage Temperature:	-20C to +85C
Humidity:	8% to 80% non-condensing
Storage humidity:	8% to 80% non-condensing

Physical Characteristics

Form Factor:	Full-size PCI card

Power Requirements

SmartWORKS™ NGX (base)

+ 3.3 VDC:	0.9 A
+5 VDC:	15 mA
-12 VDC:	25 mA
+12 VDC:	25 mA

SmartWORKS™ NGX (expanded 24 channels)

+ 3.3 VDC:	1.6 A
+5 VDC:	15 mA
-12 VDC:	35 mA
+12 VDC:	35 mA

Tap Interface

Insertion loss:	<1dB
Isolation:	Galvanic 500VDC +/-10%, 100VRMS 1 sec
Impedance:	Soft-Switchable 1KOhms/100Ohms
External connector:	RJ-21X 25 Pair female

SDK

Ai-Logix Native SmartWORKS™ API

SmartControl (Control Panel)

SmartVIEW (Card functionality test application)

Host Interface

Bus Compatibility:	Complies with PCISIG Bus Specifications, Rev. 2.2
Bus Speed:	33 MHz
Bus Mode:	32 bit bus master/target

Analog Jack

Audio Connector:	3-pin 0.1" ctr header
Output impedance:	300Ohms
Input impedance:	33KOhms
Return loss:	>25dB
Mic bias:	+5VDC @ 4.7KOhms
Input gain:	+9dB
Output gain:	2.6dBm @ 300Ohms
Full scale input:	370 mVRMS
Full scale output:	1.5 VRMS open circuit

PBX Interface

PBX Support:	Software Configurable

see www.ai-logix.com for a complete list

Audio Signal

Receive range:	-68 dBm to + 3 dBm
Input gain control:	+24 to -50 dB
Silence Detection:	Programmable from API
Transmit volume control:	+24 to -50 dB to MVIP/H.100
Automatic Gain Control (AGC):	Programmable from API
Automatic Volume Control (AVC):	Programmable from API
Activity Detection:	Programmable from API
Frequency Response:	300 - 3400 Hz (+/- 3dB)

Audio Digitizing (Encoding & Decoding)

5.3 Kb/s	G.723.1
6.3 Kb/s	G.723.1
8 Kb/s:	G.729A
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16 Kb/s:	G.726
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64 Kb/s:	μ-law or A-law per G.711, 8 bit linear PCM (signed & unsigned)
96 Kb/s	6 Khz 16 bit linear PCM(signed)
128 Kb/s:	16 bit linear PCM (signed & unsigned)
Wave file formats:	Microsoft GSM, Linear signed 8 & 16-bit PCM
Digitization selection:	Programmable per channel, independent for encode and decode

DTMF Tone Detection

DTMF digits:	0 - 9, *, #, A, B, C, D
Dynamic range:	-38 dBm to 0 dBm
Minimum tone detection:	40 ms /programmable
Interdigit timing:	40 ms min.
Acceptable twist:	Per LSSGR sec. 6, 8 dB forward, 4 dB reverse
Frequency variation:	Accept all +/- 1.5%, reject all +/-2.5%
Noise tolerance:	Per LSSGR sec. 6
Talk off:	Bellcore TR-TSY-000762

D Channel Events

The following types of D-channel events are decoded:

PBX Event (Command Events):

Generated by the PBX and passed to the phone as a command to perform some type of action.

Signaling - these events indicate a call progress tone (dial tone, ring tones), or audio changes

LEDs - these events correspond to light changes on the phone

Display - these events indicate that the LCD on the phone has been updated. These are usually related to the clock display, or messages displayed on the LCD.

Phone Events

Generated by the phone indicating an action has been taken (i.e. button pressed).

Hook State - off hook and on hook changes occur when the handset is removed or replaced

Button events - indicate that a button on the phone was used. For example: digits pressed, speaker buttons etc.

Safety and Certifications

Telecom:	DOC
Emissions:	FCC Part 15 class A · EN 55022
Immunity:	EN 55024
Safety:	EN 60950
Estimated MTBF:	250,000 hours per Bellcore Method I

Models Available

NGX800	8 port
NGX1600	16 port
NGX2400	24 port
MX80	8 port daughtercard

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· Media Control - CODECS

· Tone Detection / Generation

· CallerID/FSK/DTMF/MF Detection

· Activity / Silence Detectors

· Switching (H.100 and MVIP)

· Automatic Gain Control (AGC)

· Automatic Volume Control (AVC)

· Stereo Recording with AGC

· Echo Cancellation

· Call Progress Monitoring (CPM)

· Full-duplex Channels

· Media Streaming

· Live Monitoring

· Start/Stop Call Recording Triggers

· Beep tone generation for passive mode

Since 1991, Ai-Logix has designed boards used in interactive and passive telephony applications. With global support for all types of telephone and radio systems - analog, digital, and enterprise PBXs, Ai-Logix products have set a new world standard in telephony communications. A single API, combined with event driven reporting simplifies application development by providing one standard for all types of networks.

The SmartWORKS™ DP sets the standard for passive tapping of T1/E1 trunks in high-density environments. The SmartWORKS™ DP is a reliable tool used globally by many of the world's largest call logging application providers.

Key Features and Benefits

Software Switchable T1/E1 Interface

Supports T1 and E1 using the same board. Automatically configures for all supported ISDN variants.

ISDN Call State Monitoring

Interprets the ISDN signaling protocol and reports the call states and call parameters via comprehensible API events.

True Dual Span Capabilities

A single RJ-45 interface is capable of recording both sides of a conversation, which maximizes the usefulness of each individual port.

On-board DSP to Complete Voice Processing

Robust call recording features combined with ISDN call control, eliminates the need for other resources on the system.

CODEC Support

SmartWORKS™ call recording products offer a large selection of voice CODECS.(including G.723.1, G.729A and MS GSM)

High Density Passive Tap Capabilities

Operating between a central office and PBX, the SmartWORKS™ DP's high impedance receivers records both sides of a call without interrupting service. Each board can process up to 60 channels, with a maximum of 512 channels per host. Service is never interrupted even if the SmartWORKS™ DP-equipped PC is shut down.

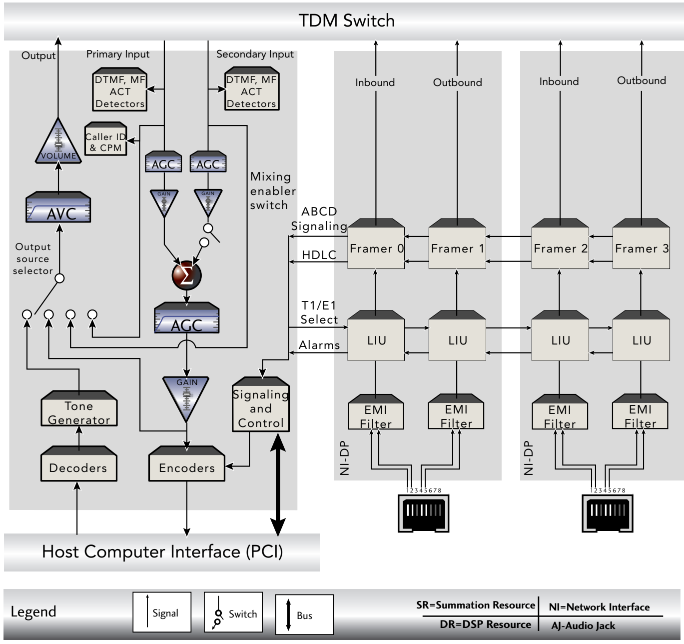
International Protocol Support

The SmartWORKS™ DP supports Channel Associated Signaling (CAS), Non-Facility Associated Signaling (NFAS) , DASS2 and any Q.931 based ISDN variant. Trunk coding and framing is selected on a per framer basis. This allows a single board to monitor two trunks, each with different settings.

Built in Performance Monitoring

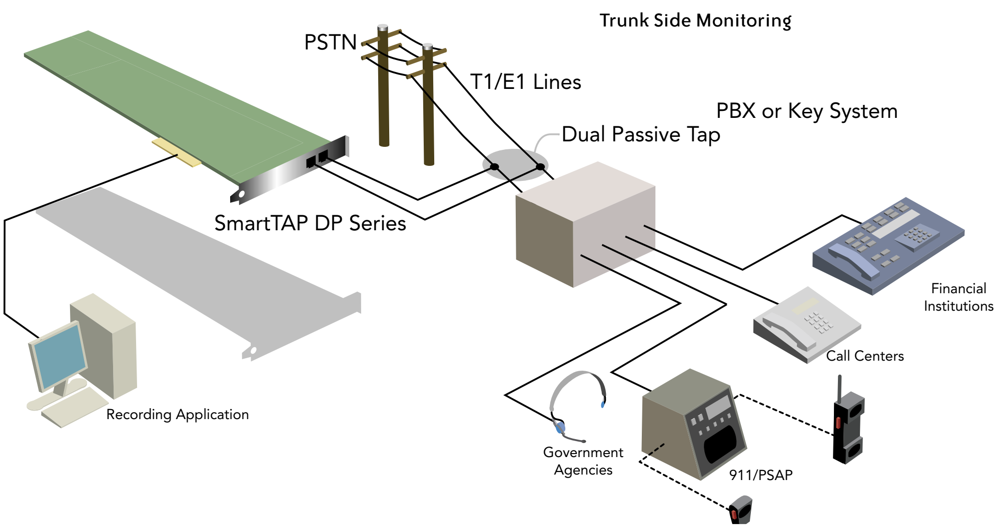
Network conditions and call statistics are easily accessed via the SmartWORKS™ API. Event driven alarms are reported for loss of signal conditions or synchronization errors. Framer and call statistics are available through standard API function calls.

DP Logical Card Model



DP Application Model

Applied Use: With a proven field record, the SmartWORKS™ DP has been successfully deployed in various international agencies such as banking, law enforcement, trading and customer support centers.



Hardware System Requirements

Pentium 4 or equivalent · 2 GHz or better

PCI motherboard or passive backplane with 3.3V power supply, PCI 2.2 bus

Operating Systems

Windows 2000 · Windows XP

Windows 2003 32 bit· Linux (Call for variant details)

Technical Specifications

Max boards per system:	Any combination up to 512 ports
Max ports per system:	Up to 512
Resource Sharing Bus:	MVIP or H.100
Boards Status:	On-board LEDs
Clocking:	Master/Slave

Environmental Conditions

Operating Temperature:	0C to +60C
Storage Temperature:	-20C to +85C
Humidity:	8% to 80% non-condensing
Storage humidity:	8% to 80% non-condensing

Physical Characteristics

Form Factor:	Full-size PCI card

Power Requirements (6409)

+ 3.3 VDC:	2.8 A
+5 VDC:	5 mA
-12 VDC:	n/a
+12 VDC:	20 mA

SDK

Ai-Logix Native SmartWORKS™ API

SmartControl (Control Panel)

SmartVIEW (Card functionality test application)

Host Interface

Bus Compatibility:	Complies with PCISIG Bus Specifications, Rev. 2.2
Bus Speed:	33 MHz
Bus Mode:	32 bit bus master/target

DTMF Tone Detection

DTMF digits:	0 - 9, *, #, A, B, C, D
Dynamic range:	-38 dBm to 0 dBm
Minimum tone detection:	40 ms / programmable
Interdigit timing:	40 ms min.
Acceptable twist:	Per LSSGR sec. 6, 8 dB forward, 4 dB reverse
Frequency variation:	Accept all +/- 1.5%, reject all +/-2.5%
Noise tolerance:	Per LSSGR sec. 6
Talk off:	Bellcore TR-TSY-000762

Telephony Interface

Trunk type:	T1/E1
Trunk Interface	Digital High Impedance (Z)
AC Impedance	1k Ohms
Input Impedance	1000 Ohm +/- 5%
Maximum Tap Length	30m (100 feed) of CAT 3 cable
Connectors	Two RJ-45 connectors

T1 Interface

Receive Clock Rate	1.544 MHz +/-200ppm
Transmit Clock	Recovered RX clock or 50 ppm
Input Level	LBO 0dB to -22dB
Framing	SF (D4), ESF
Line Coding	AMI, B8ZS
Signaling Protocol	ISDN, NFAS, CAS
Clock and Data Recovery	Complies with AT&T TR62411 and Bellcore TA-TSY-000170
Loss of Signal Detection	ANSI T1.231
Alarm Detection and Integration	LOS, LOF, Yellow, and AIS per ANSI T1.231
Binary Sequence Detector	Per ITU-T 0.151

E1 Interface

Receive Clock Rate	2.048 +/- 175ppm
Transmit Clock	Recovered RX clock or 50 ppm
Input Level	3.2V down to 0.45 V
Framing	Basic G.704, CRC-4
Line Coding	AMI, HDB3
Signaling Protocol	ISDN, DASS2, CAS
Loss of Signal Detection	per ITU-T G.775
Alarm Detection and Integration	LOS, LOSMF, TS16, CRC
Binary Sequence Detector	Per ITU-T 0.151

Audio Signal

Receive range:	-68 dBm to + 3 dBm
Input gain control:	+24 to -50 dB
Silence Detection:	Programmable from API
Transmit volume control:	+24 to -50 dB to MVIP/H.100
Automatic Gain Control (AGC):	Programmable from API
Automatic Volume Control (AVC):	Programmable from API
Activity Detection:	Programmable from API
Frequency Response:	300 - 3400 Hz (+/- 3dB)

Audio Digitizing (Encoding & Decoding)

5.3 Kb/s	G.723.1
6.3 Kb/s.	G.723.1
8 Kb/s:	G.729A
13 Kb/s:	GSM 6.10, Microsoft GSM
16 Kb/s:	G.726
24 Kb/s:	G.726, OKI
32 Kb/s:	G.726, OKI
40 Kb/s:	G.726
64 Kb/s:	μ-law or A-law per G.711, 8 bit linear PCM (signed & unsigned)
96 Kb/s	6 Khz 16 bit linear PCM(signed)
128 Kb/s:	16 bit linear PCM (signed & unsigned)
Wave file formats:	Microsoft GSM, Linear signed 8 & 16-bit PCM
Digitization selection:	Programmable per channel, independent for encode and decode

Safety and Certifications

Telecom:	DOC
Emissions:	FCC Part 15 class A · EN 55022
Immunity:	EN 55024
Safety:	EN 60950
Estimated MTBF:	250,000 hours per Bellcore Method I

Models Avaiable

DP3209:	Single E1/T1
DP6409:	Dual E1/T1

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Standard Features for SmartWORKS™ Family of Call Recording Products

The SmartWORKS™ API provides a common interface that controls the following call recording features:

· Media Control - CODECS
· Tone Detection / Generation
· CallerID/FSK/DTMF/MF Detection
· Activity / Silence Detectors
· Switching (H.100 and MVIP)
· Automatic Gain Control (AGC)
· Automatic Volume Control (AVC)
· Stereo Recording with AGC
· Echo Cancellation
· Call Progress Monitoring (CPM)
· Full-duplex Channels
· Media Streaming
· Live Monitoring
· Start/Stop Call Recording Triggers
· Beep tone generation for passive mode

Since 1991, Ai-Logix has designed boards used in interactive and passive telephony applications. With global support for all types of telephone and radio systems - analog, digital, and enterprise PBXs, Ai-Logix products have set a new world standard in telephony communications. A single API, combined with event driven reporting simplifies application development by providing one standard for all types of networks.

The SmartWORKS™ DT provides trunk termination and call control on digital T1/E1 networks. Call Progress Monitoring (CPM), DTMF detection, voice play / record, and barge-in features makes this board an invaluable resource for interactive telephony applications.

Key Features and Benefits

Software Switchable T1/E1 Interface

Supports T1 and E1 using the same board. Uses software to switch the telephony interfaces to T1 or E1 on a trunk basis.

Auto-configures for all ISDN variants

Configure to any supported ISDN variant. Save time and reduce operator error when installing and configuring the board in the field.

ANI and DNIS

Calling and called numbers are collected from ISDN signaling packets and passed to the user application via the SmartWORKS™ API.

On-board DSP to complete voice processing

Encoding capabilities, with a rich set of CODECS, reduces the need to purchase other hardware components.

CODEC Support

SmartWORKS™ products offer a large selection of voice CODECS.(including G.723.1, G.729A and MS GSM)

Terminate Environment

The SmartWORKS™ DT connects directly to a Central Office or PBX providing line supervision to answer and generate inbound and outbound calls. Each board processes up to 60 channels, with a maximum of 512 channels per system. Each channel has programmable volume control, tone generation, echo cancellation, and Call Progress Monitoring. Outbound dialing and call control is managed through the SmartWORKS™ API.

International ISDN Support

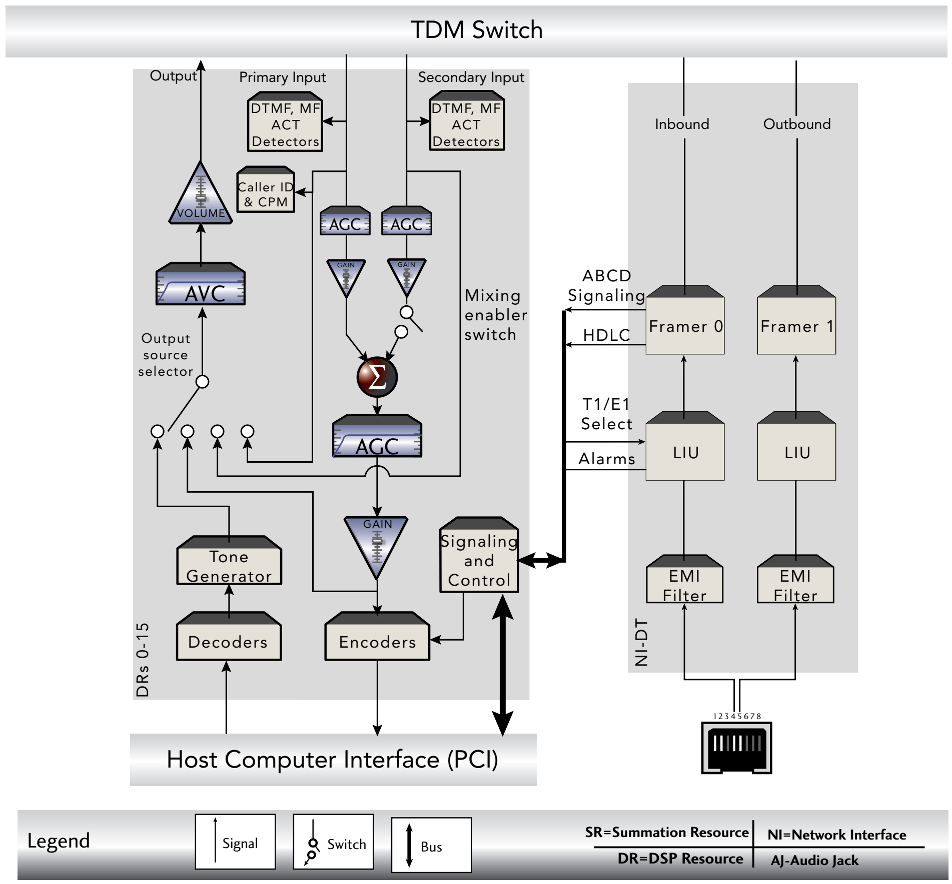
The SmartWORKS™ DT supports Channel Associated Signaling (CAS), and any Q.931 based ISDN variant. Trunk coding and framing is selected on a per framer basis. This allows a single board to control two trunks, each with different settings.

Built in Performance Monitoring

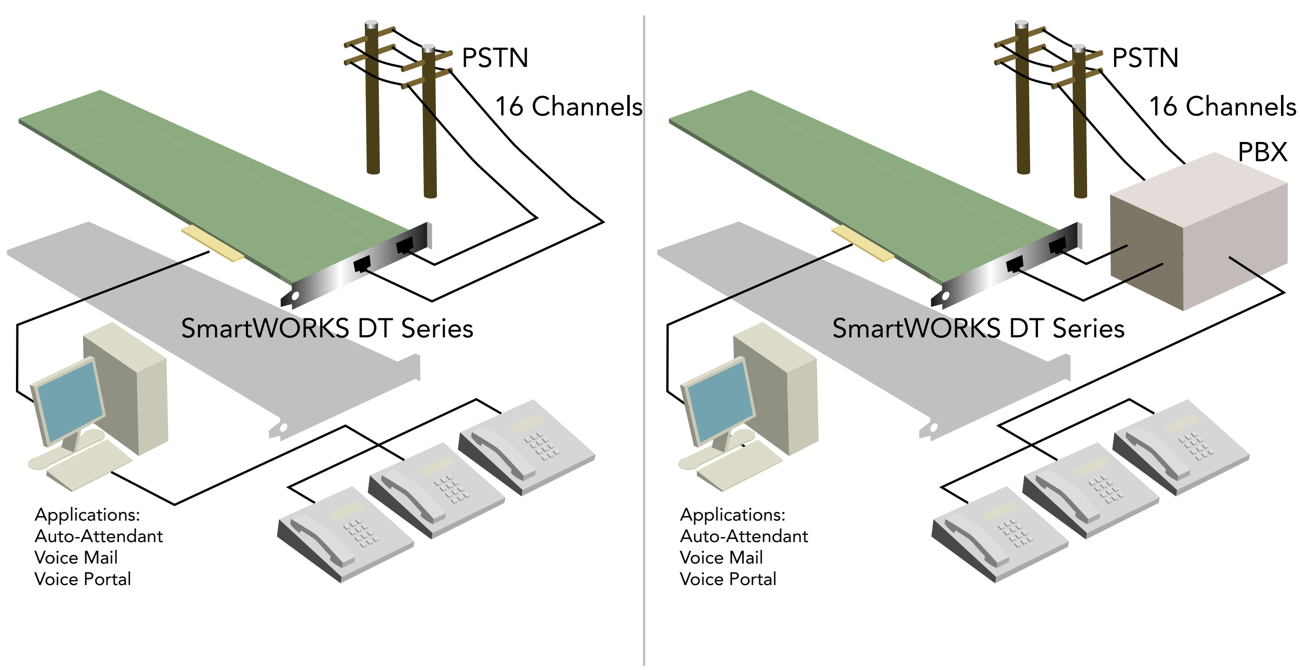
Network conditions and call statistics are available via the SmartWORKS™ API. Event driven alarms are reported for loss of signal conditions or synchronization errors. Framer and call statistics are available through standard API function calls.

Ai-Logix An AudioCodes Company

DT Logical Card Model



DT Application Model



Product Specifications · SmartWORKS™ DT

Hardware System Requirements

Pentium 4 or equivalent · 2 GHz or better
PCI motherboard or passive backplane with 3.3V power supply, PCI 2.2 bus

Operating Systems

Windows 2000 · Windows XP
Windows 2003 32 bit· Linux (Call for variant details)

Technical Specifications

Max boards per system:	Any combination up to 512 ports
Max ports per system:	Up to 512
Control Microprocessor	Motorola Coldfire™ RISC (50 MHz)
DSP	Multiple Texas Instruments TMS320C5409 A
Boards errors	On-board LEDs
Clocking	Master/Slave
DRAM	16 MB per board
SRAM	128 Kword/DSP

Environmental Conditions

Operating Temperature:	0C to +50C
Storage Temperature:	-20C to +85C
Humidity:	8% to 80% non-condensing
Storage humidity:	8% to 80% non-condensing

Physical Characteristics

Form Factor:	Full-size PCI card

Host Interface

Bus Compatibility:	Complies with PCISIG Bus Specifications, Rev. 2.2
Bus Speed:	33 MHz
Bus Mode:	32 bit bus master/target
Shared Memory:	16 MB Global shared RAM

SDK

Ai-Logix Native SmartWORKS™ API
SmartControl (Control Panel)
SmartVIEW (card functionality test application)
SmartWF (firmware flash update utility)

Power Requirements (6409)

+3.3 VDC:	2.8 Amp
+5 VDC:	5mA
-12 VDC:	Not Required
+12 VDC:	20 mA

Telephony Interface

Trunk Type	T1/E1
Trunk Interface	Digital network interface
Connectors	RJ-45 connectors

T1 Interface

Receive Clock Rate	1.544 MHz +/-200ppm
Transmit Clock	Recovered RX clock or 50 ppm
Input Level	LBO 0dB to -22dB
Framing	SF (D4), ESF
Line Coding	AMI, B8ZS
Signaling Protocol	ISDN, NFAS, CAS
Robbed Bit Signaling	E&M Immediate, E&M wink, FXS, FXO
Clock and Data Recovery	Complies with AT&T TR62411 and Bellcore TA-TSY-000170
Loss of Signal Detection	ANSI T1.231
Alarm Detection and Integration	LOS, LOF, Yellow, and AIS per ANSI T1.231
Binary Sequence Detector	Per ITU-T 0.151

E1 Interface

Receive Clock Rate	2.048 +/- 175ppm
Transmit Clock	Recovered RX clock or 50 ppm
Input Level	3.2V down to 0.45 V
Framing	Basic G.704, CRC-4
Line Coding	AMI, HDB3
Signaling Protocol	ISDN, DASS2, CAS
Loss of Signal Detection	per ITU-T G.775
Alarm Detection and Integration	LOS, LOSMF, TS16, CRC, and Yellow
Binary Sequence Detector	Per ITU-T 0.151

Audio Signal

Receive range:	-68 dBm to + 3 dBm
Input gain control:	+24 to -50 dB
Silence Detection:	Programmable from API
Transmit volume control:	+24 to -50 dB
Automatic Gain Control	(AGC)Programmable from API
Automatic Volume Control (AVC)	Programmable from API
Activity Detection	Programmable from API
Alert Tone	Programmable
Frequency Response	300 - 3400 Hz (+/- 3dB)

Call Progress Monitoring

Number of programmable tones	20
Number of bandpass filters	10
Number of filters per tone	1,2 or 3
Number of cycles	0 to 255
SIT tones and duration	Yes, programmable frequencies
Answering Machine Detection	Yes

Tone Dialing

DTMF digits	0 – 9, *, #, A, B, C, D
Frequency variation	Less then 1 Hz
Rate	API Programmable

Audio Digitizing (Encoding & Decoding)

5.3 Kb/s	G.723.1
6.3 Kb/s	G.723.1
8 Kb/s:	G.729A
13 Kb/s:	GSM 6.10, Microsoft GSM
16 Kb/s:	G.726
24 Kb/s:	G.726, OKI
32 Kb/s:	G.726, OKI
40 Kb/s:	G.726
64 Kb/s:	μ-law or A-law per G.711, 8 bit linear PCM (signed & unsigned)
96 Kb/s	6 Khz 16 bit linear PCM(signed)
128 Kb/s:	16 bit linear PCM (signed & unsigned)
Wave file formats:	Microsoft GSM, Linear signed 8 & 16-bit PCM
Digitization selection:	Programmable per channel, independent for encode and decode

DTMF/MF Tone Detection

DTMF digits:	0 - 9, *, #, A, B, C, D
MF R2 Digits	15 Digits Forward & Reverse per Q.441
Dynamic range:	-38 dBm to 0 dBm
Minimum tone detection:	40 ms / programmable
Interdigit timing:	40 ms min.
Acceptable twist:	Per LSSGR sec. 6, 8 dB forward, 4 dB reverse
Frequency variation:	Accept all +/- 1.5%, reject all +/-2.5%
Noise tolerance:	Per LSSGR sec. 6
Talk off:	Bellcore TR-TSY 000762

Trigger Conditions

Event Driven	Caller ID, Min/Max silence · Min/Max activity

Global Tone Generation

Tone Type	Single or dual frequency
Frequency range	300 Hz – 3400 Hz
Frequency resolution	1 Hz
Duration	1 ms – 8191 ms programmable in 1 ms steps
Amplitude	+3 dBm to –68 dBm
Duration	API Programmable

Voice Processing

Echo cancellation	G.165
Caller ID	V.23 & Bell 202
DTMF Detector channel	Primary & Secondary
MF Detection	R1 & R2

Safety and Certifications

Telecom:	DOC
Emissions: EN 55022	FCC Part 15 class A ·
Immunity:	EN 55024
Safety:	EN 60950
Estimated MTBF: Bellcore Method I	150,000 hours per

Models Available

DT3209	Single E1/T1
DT6409	Dual E1/T1

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WordALERT™ Series

Realtime Word Spotting

Standard Features for WordALERT Bundles

· Discrete - Word Recognition

· Phrase – Word Recognition

· Echo cancellation

· Polling mode API

· SNMP support

· VXML and SALT support through ABNF

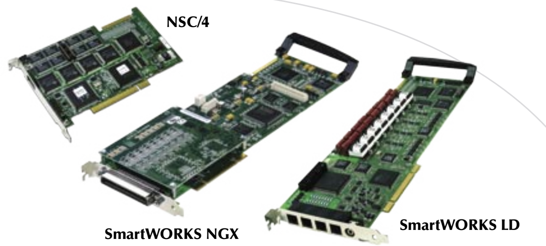
· Grammar Development Tools

· Multiple Language

· Board based licensing model

Since 1991, Ai-Logix has designed boards used in interactive and passive telephony applications. With global support for all types of telephone and radio systems - analog, digital, and enterprise PBXs, Ai-Logix products have set a new world standard in telephony communications. A single API, combined with event driven reporting simplifies application development by providing one standard for all types of networks.

The WordALERT family of products is ideal for adding real-time key word spotting (KWS) or post real-time word mining to your application. WordALERT boards are speaker independent voice recognition products tailored for ASR in the call recording industry. Featuring a unique DSP based engine (designed by NSC Speech), WordALERT products provide flexibility and performance without taxing the host processor as well as board based language licensing.



Key Features and Benefits

Predictable Performance

Easily determine the number of channels and words supported per board.

Dynamic Grammar Allocation

Change the list of words to be recognized on the fly by using the provided grammar development tools.

Multiple Language Support

Each DSP can run a different standard or custom language.

Custom Language Support Available

The Phoneme based architecture makes it easy to add special language cases

Host Independent

The recognition runs on the board and not on your host processor. No host MIPS calculation needed.

Flexible Architecture

The DSP architecture supports from 1 to 4 sessions per DSP depending on the applications grammar requirements

Scalable

Up to 4 NSB boards can be used in a chassis

Tap Environment

Combine WordALERT products with any of the SmartWORKS Passive Tap products to add KWS. Up to 512 channels of passive tap can be serviced via H.100.

Word Mining Environment

Use the WordALERT products with any available SmartWORKS VR resource.

Files recorded using any Ai-Logix supported codec are converted via the H.100 bus for WordALERT analysis.

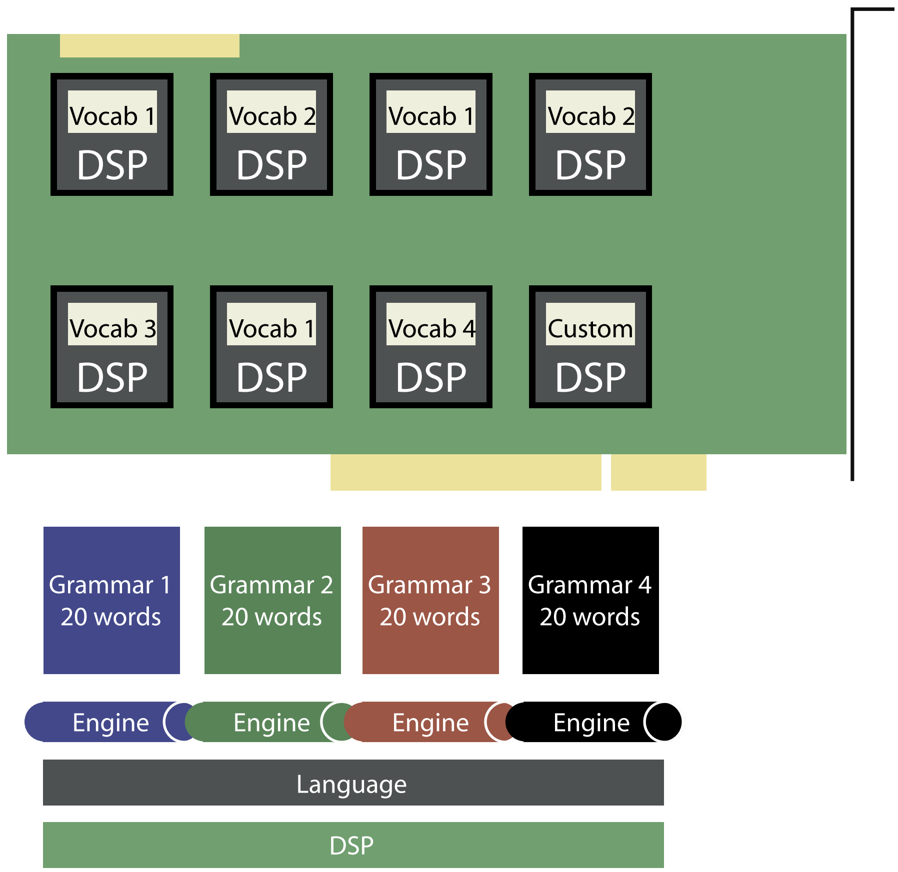
Grammar Development

NSCGrammar is a development tool for defining recognition tasks through the specification of a grammar syntax.

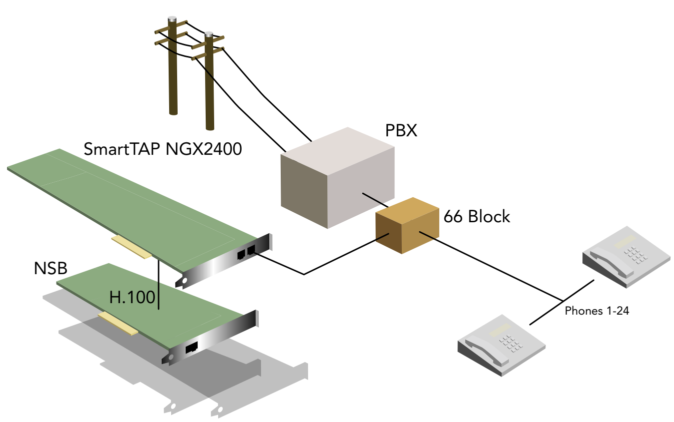
NSCConfig is a configuration tool that provides control of the ASR resource.

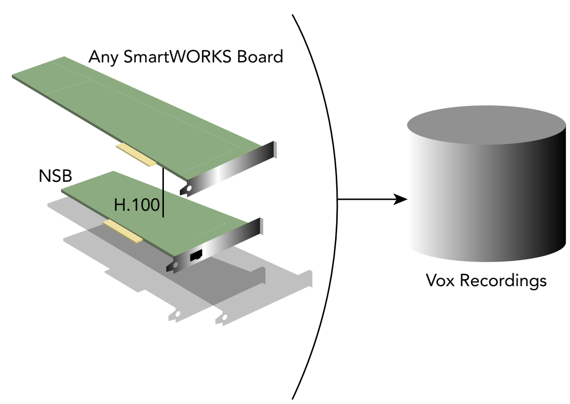
Ai-Logix An AudioCodes Company

WordALERT Architecture



WordALERT Application Models





Product Specifications · WordALERT

Hardware System Requirements

Pentium 4 or equivalent · 2 GHz or better
PCI motherboard or passive backplane with 3.3V power supply, PCI 2.2 bus

Operating Systems

Windows 2000
Windows XP (planned for Q1 2005)
Linux (please call for variant information)

Speech Format

G.711: .... DSR (ETSI ES 201 108)

Languages

US English
UK English
Arabic
Russian
Hebrew
Spanish
French
German

Physical Characteristics

Form Factor:. .... Full or Half-size PCI

Power Requirements

PCI (Complies with PCISG Bus Specification, Rev 2.2)

NSB2 (pci-u)	+3.3VDC, typical: 0.94A maximum:1.25A
NSB4 (pci-u)	+3.3VDC, typical:1.3A maximum:1.7A
NSB12	TBD
NSB30	TBD

Host Interface

Bus Speed .... 33 MHz
Bus Width. .... 32-bit bus target

Safety and Certifications (pending)

Emissions	FCC Part 15 Class A, EN55022
Immunity	EN55024
Safety	EN60950
*NSB2 & NSB4	planned for Q1 - 2005
*NSB12 & NSB30	planned for Q2 – 2005

Technical Specifications

Resource Sharing Bus...................................H.100 and SCSA

Products

Boards:

NSB2	Half size 2 DSP board, up to 8 concurrent channels
NSB4	Half size 4 DSP board, up to 16 concurrent channels
NSB12	Full size 12 DSP board, up to 48 concurrent channels
NSB30	Full size 30 DSP board, up to 120 concurrent channels

Packages:

WADK	(WordALERT Developers Kit): NSB2, LD409H, SDK, NA English Language
WT3209	(WordALERT Trunk Kit): DP3209, NSB12
WS2400	(WordALERT Station Kit): NGX2400, NSB12

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ACCESSORY: RTS BOX

RESISTIVE TAP SPLITTER

Features

Provides both voice and D-channel data for:

· Avaya INDeX

· Mitel SX200/SX2000

· Siemens Rolm 9751

Breaks out full-duplex signal into 2 half-duplex signals

Passive device - No power supply required

Key Features and Benefits

The RTS BOX is a tap-point product designed for use with the SmartWORKS™ NGX. Using the RTS BOX with two NGX2400 boards provides a 24-port solution for call recording applications.

The RTS BOX provides voice and D-channel data for the Avaya INDeX, Mitel SX200/ SX2000* and the Siemens Rolm 9751. The unique design supports all 3 PBXs with one assembly. The RJ21x Amp connectors provide reliable connections to the NGX, PBX, and phones by treating the PBX as a 4-wire supporting half the cahnnels.

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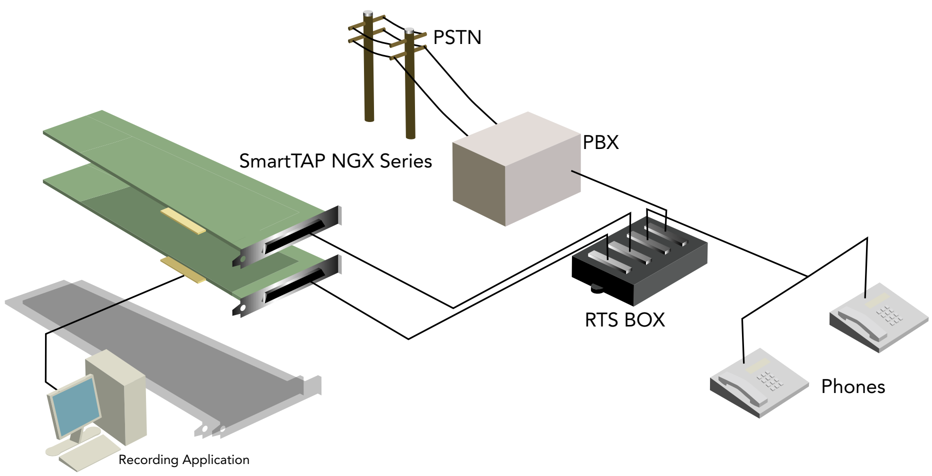
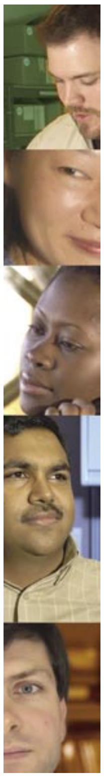
RTS BOX Application Diagram



*Mitel requires a per port license (sold separately)

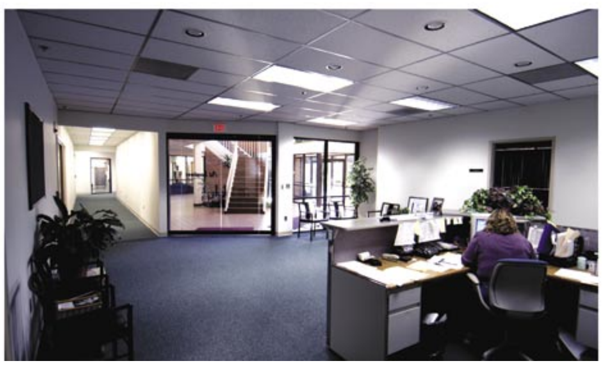
Inside Ai-Logix



Ai-Logix, Inc. designs and manufactures the finest hardware-based computer telephony integration (CTI) products for interactive and passive telephony applications. We are a subsidiary of AudioCodes, Ltd. (Nasdaq: AUDC, a leading provider of innovative, reliable and cost-effective Voice over Packet technology and Voice Network products.

Our headquarters are located in Somerset, NJ with local and international branches throughout the globe. We are committed to developing quality products, providing great service, and partnering with our customers to build long-term relationships.

Since 1991, we've provided complete analog, digital, and enterprise PBX support for various types of telephone and radio systems. We have raised the standards for telephony communications and take great pride in our ability to mix and match our technology to meet the evolving needs of the CTI industry. Our versatility arms us with a distinct competitive advantage unmatched by any other board supplier in the open market.



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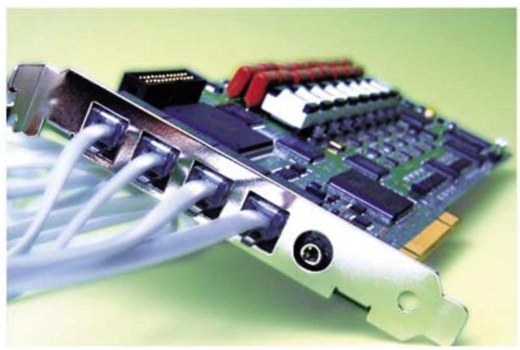

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