STRONG BASE. AUDIOSCIENCE Sound Engineering Sonic Excellence SMART FUTURE.

2022 CATALOG

LATEST PRODUCTS

AUDIOSCIENCE IS THE AUDIO I/O COMPANY. For over 25 years AudioScience continues to master the design and manufacturing of advanced DSP-based digital audio peripherals for the original equipment manufacturer (OEM), broadcast, installed sound, and entertainment industries. Our easy-to-integrate products provide outstanding digital audio functionality and compatibility. Instead of using proprietary designs that limit system hardware and software flexibility, we develop standards-based technology. With this approach, our innovations are fully compatible with other products and systems. What's more, in an industry that constantly evolves, we leverage our design and market expertise to deliver the most relevant, up-to-date products.

Our primary product lines are:

- **Network audio.** Our digital audio network products use Dante/AES67, AVB, and Livewire/AES67 network protocols.
- **Broadcast sound cards.** From the beginning, we've been on the leading edge of audio card technology. We offer products and technical partnerships to digital automation system designers.
- **Tuner cards.** Our tuner cards are used worldwide to monitor AM, FM with RDS, HD Radio™ and DAB/DAB+ broadcast transmissions.
- **OEM products.** We offer co-venture and custom-design services to facilitate the creation of new digital audio applications for OEMs and systems integrators, paving the way to greater market opportunities.

Keeping in mind that system integrators and developers require new technologies to create leading-edge applications, we focus on enhancing functionality. Our mission is to deliver powerful digital audio solutions at a fair price to businesses with distributed audio systems.

To learn more about our innovative, digital audio products, please visit www.audioscience.com, call (302) 235-7109, or email salesasi@audioscience.com. We look forward to speaking with you!

CONTENTS

SOUND ENGINEERING.
SONIC EXCELLENCE.

ASI 2816 STREAMING TUNER

THE ASI2816 IS A MULTI-CHANNEL STREAMING RADIO TUNER PACKAGED IN A 1U FORM FACTOR. It can be configured with up to four modules. Each module contains four radio tuners for a total of up to 16 tuners per ASI2816. The ASI1471 module contains four AM/FM/WeatherBand (WB) tuners. The ASI1472 module contains four AM/AM-HD/FM/FM-HD tuners. Up to sixteen ICY encoded audio streams can be sent from the ASI2816. The format of the streams is mono 44.1kHz AAC-LC.

ENVOY AVB SERIES

AUDIOSCIENCE INTRODUCES THE NEW ENVOY AVB intended for use in courtrooms, conference rooms, boardrooms, lecture halls or in business centers. All three units have a unique wedge-shaped anodized aluminum form factor.

Each unit is designed to allow tabletop or wall mounting via a bracket for convenience and ease of use.

All units are Power over Ethernet plus (PoE+) enabled with the Ethernet RJ45 jack being hidden behind a tamper proof panel. Identify LEDs on each side of the chassis allow remote identification over the network.

THE ENVOY AVB 1.1MS is an AVB device intended for intercom applications. It contains a full range mono speaker, a microphone preamp with phantom power and dual 3.5mm headphone outputs. The control panel contains 5 touch sensitive buttons.

THE ENVOY AVB 0.2AMP is a 2 channel AVB audio amplifier with a total power of 16W RMS. Outputs are on locking SpeakerCON connectors.

THE ENVOY AVB 4.4XLR is an AVB device that contains four balanced mic/line inputs and four balanced line outputs on XLR connectors. +48V phantom power is software selectable.

ENVOY AVB

ENVOY AVB SERIES

The Envoy AVB have a unique wedge-shaped anodized aluminum form factor. Each unit is designed to allow tabletop or wall mounting via a bracket for convenience and ease of use. All units are Power over Ethernet plus (PoE+) enabled with the Ethernet RJ45 jack being hidden behind a tamper proof panel. Identify LEDs on each side of the chassis allow remote identification over the network.

THE ENVOY AVB 1.1MS is an AVB device intended for intercom applications. It contains a full range mono speaker, a microphone preamp with phantom power and dual 3.5mm headphone outputs. The control panel contains 5 touch sensitive buttons.

AUDIO INPUTS

- 1 balanced analog microphone/line input
- Software adjustable, non-volatile, input levels from –60 to +24dBu
- Software selectable 48V phantom power
- Shock mounted XLR connector
- Mute button (with disable option)

AUDIO OUTPUTS

- 1 mid range speaker
- 1 tweeter speaker
- 2 stereo headphone outputs on 3.5mm phone jacks
- Individual volume up/down buttons

THE ENVOY AVB 0.2AMP is a 2 channel AVB audio amplifier with a total power of 16W RMS. Outputs are on locking SpeakerCON connectors.

AUDIO OUTPUTS

- 2 amplified audio outputs.
- Neutrik 4-pole SpeakON connectors
- 8W RMS maximum into 4 or 8 ohm speakers when both outputs used
- 16W RMS maximum into 4 or 8 ohm speakers when one output used
- Software adjustable, non-volatile, output levels from -30 to 0dBFs
- THD+N is < 0.02% @ 0dBFs, 1kHz -1dB sinewave, A weighted
- Frequency response is 20-20kHz +/-3dB

THE ENVOY AVB 4.4XLR is an AVB device that contains four balanced mic/line inputs and four balanced line outputs on XLR connectors. +48V phantom power is software selectable.

AUDIO INPUTS

- 4 balanced analog microphone/line input
- Software adjustable, non-volatile, input levels from –60 to +24dBu
- Software selectable 48V phantom power
- THD+N is <-90dB @ +20dBu, 1kHz -1dBFs sinewave, A weighted
- Dynamic range is >100dB
- Frequency response is 20-20kHz +/-3dB

AUDIO OUTPUTS

- 4 balanced analog line outputs
- Software adjustable, non-volatile, input levels from –10 to +24dBu
- THD+N is <-90dB @ +20dBu, 1kHz -1dBFs sinewave, A weighted
- Dynamic range is >100dB
- Frequency response is 20-20kHz +/-3dB+48V phantom power is software selectable.

ASI2816 STREAMING TUNER

ASI2816 AM/FM/HD/WB 4-16 CHANNEL STREAMING TUNER

The ASI2816 is a multi-channel streaming radio tuner packaged in a 1U form factor. It can be configured with up to four modules. Each module contains four radio tuners for a total of up to 16 tuners per ASI2816. The ASI1471 module contains four AM/FM/WeatherBand (WB) tuners. The ASI1472 module contains four AM/AM-HD/FM/FM-HD tuners.

Up to sixteen ICY encoded audio streams can be sent from the ASI2816. The format of the streams is mono 44.1kHz AAC-LC. The ASI2816 contains an OLED front panel display showing current configuration and status of all tuners including band, frequency, signal strength and HD status.

Each tuner module also includes balanced stereo audio outputs on a StudioHub compatible RJ-45 connector with a software adjustable level of 0 to +24dBu

Modules	Description
ASI2816	Base streaming tuner unit
ASI1471	4 tuner AM/FM/WB module
ASI1472	4 tuner AM/FM/AMJD/FMHD module

FEATURES

- Up to 16 channels of AM/FM HD Radio audio capture
- Up to 16 channels of analog AM/FM audio capture
- Up to 16 channels of Weather Band capture
- Antenna connector on each module or chainable across multiple modules
- Audio monitoring of all tuners simultaneously using RJ-45 StudioHub analog outputs
- Gb Ethernet interface
- 100 to 240VAC universal power supply
- Internet streaming: One ICY encoded, mono, 48kbps, AAC-LC stream @ 44.1kHz per tuner
- SNMP for monitoring
- HTML based web UI for configuring tuners and streaming parameters
- Front panel display shows tuner status and audio

ASI2816 front

ASI2816 back

IYO DANTE

THE IYO DANTE® IS A COST-EFFECTIVE FAMILY OF MICROPHONE/LINE DANTE INTERFACES WITH AES67 IN A 1U RACK MOUNT FORMAT. Several models provide various configurations of balanced analog audio inputs and outputs. Each input accommodates microphone through line level signals with a range of -60 to +24dBu. +48V phantom power is individually switchable on each input. Output levels are configurable up to +24dBu.

RGB LEDs on the Iyo's front panel show per channel audio levels and streaming status.

The Iyo family feature an embedded web server, allowing configuration and monitoring of input and output levels. Routing is achieved using the Dante Controller.

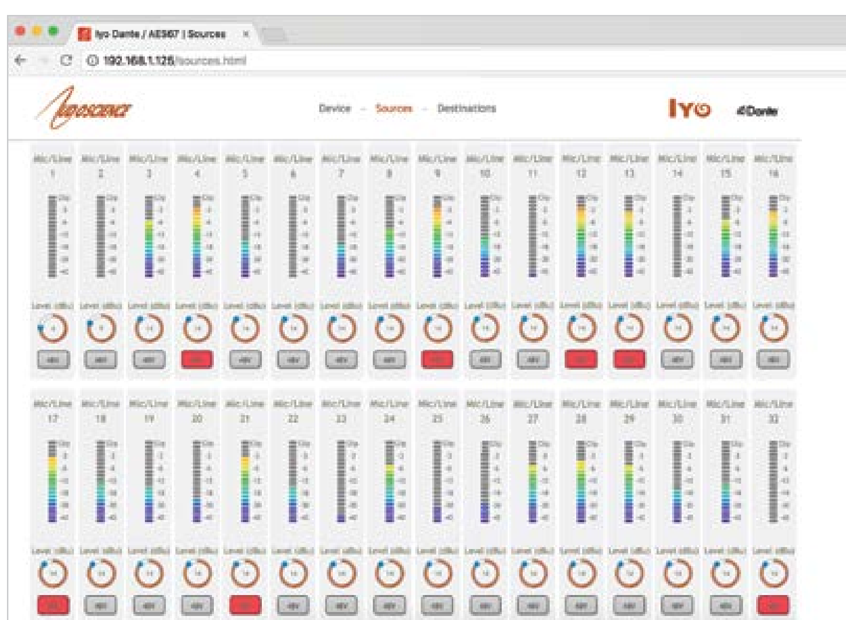
Power is provided from a built-in universal AC power supply. Redundant power is available using an external 12VDC supply via a locking 2.1mm jack.

All units can also be operated in AES67 interoperability mode.

FEATURES

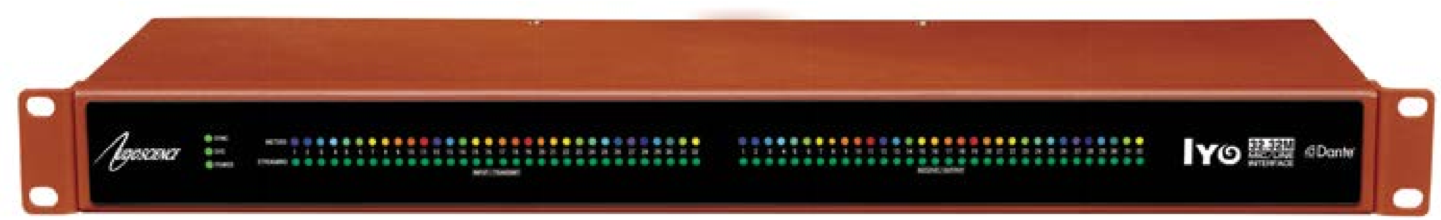
- 8x8, 16x16 or 32x32 channels of Dante® audio-over-IP with AES67 interoperability
- 48 or 96kHz sample rates.
- 1U rack-mount unit.
- Balanced microphone/line level inputs with level range of -60 to +24dBu
- Switchable +48V phantom power on each input
- Balanced line level outputs with level range of 0 to +24dBu.
- 3.81mm Terminal Block terminations.
- RGB front panel LEDs provide per channel metering and stream status
- Built-in web server provides audio level configuration and monitoring
- Dual RJ-45 network jacks can be operated in redundant or switched mode.
- Built-in universal medical grade 90-260VAC power supply.
- Auxiliary +12VDC input provides power supply redundancy.

^ RGB LEDs indicate meter level and stream status

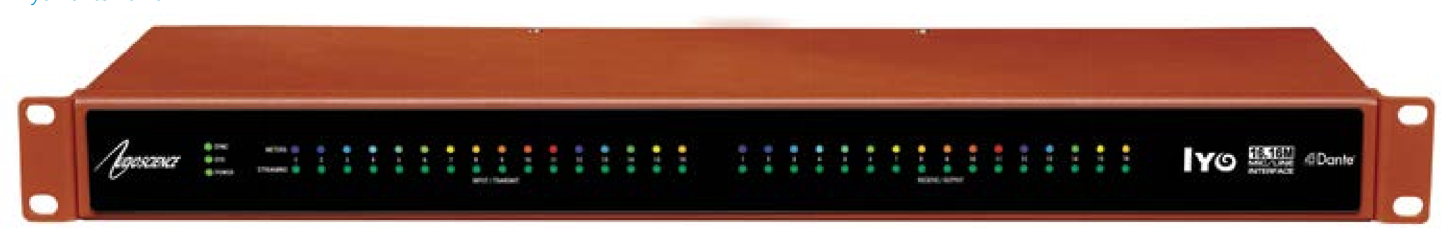


Built in web server provides level configuration and metering >

Iyo Dante 32.32M

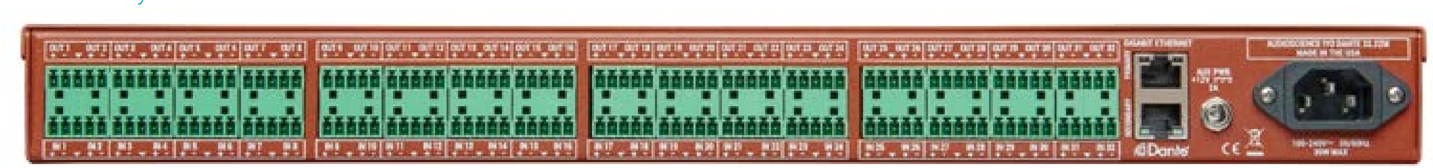


Iyo Dante 16.16M



Iyo Dante 8.8M

Rear view of Iyo Dante 32.32M



Model	Network Protocol	Audio Channels In	Audio Channels Out	Input Type	Output Type	Connectors
Iyo Dante 8.8M	Dante	8	8	Balanced Analog Mic/Line	Balanced Analog Line	Terminal Block
Iyo Dante 16.16M	Dante	16	16	Balanced Analog Mic/Line	Balanced Analog Line	Terminal Block
Iyo Dante 32.32M	Dante	32	32	Balanced Analog Mic/Line	Balanced Analog Line	Terminal Block

HONO AVB CONTROLLER

THE HONO AVB CONTROLLER is a Graphical User Interface (GUI) application intended for use with AudioScience products combined with other 3rd party AVB devices.

- View and Adjust Per Stream Setting:
  - Stream name
  - Media format
  - Audio channel mappings
  - IEEE 1722 Presentation time/stream latency
- View Network Status Information:
  - IEEE 802.1AS clock synchronization grandmaster

Logs

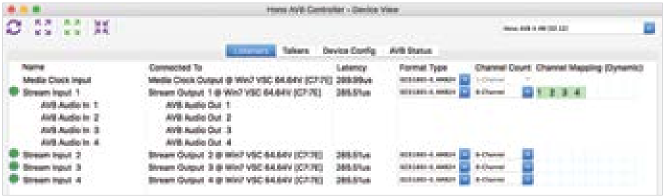
- View device error and notification logs

Operating Systems

- Runs on Windows 7/10 and macOS platforms.

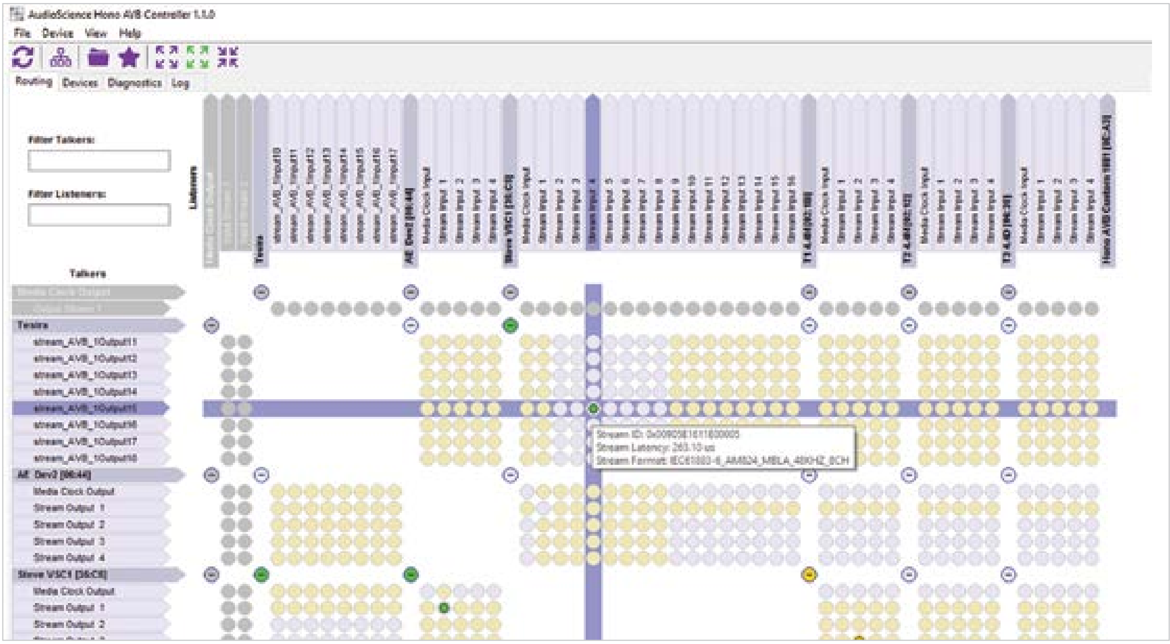
AVB

- View all IEEE 1722.1 enabled AVB devices and their streams.
- Route streams on AVB devices and examine existing stream routes.
- Presets
  - Save audio routing and device configuration presets
  - Apply previously saved presets
  - Edit presets offline and use for new device configurations
- View and adjust per-device settings, if implemented:
  - Lock/Unlock devices
  - Device name
  - Device Sample rate
  - Media clock source
  - Network information



AVB Device dialog allows configuration of stream format and channel mapping

Diagnostics tab shows AVB device state



Routing tab allows connection of AVB streams

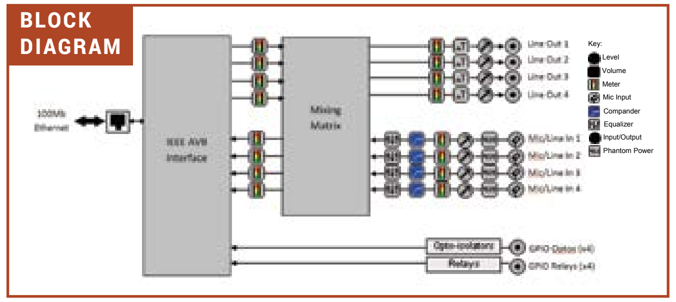
MINI

AUDIOSCIENCE'S HONO™ AVB MINI PRODUCTS ARE PERFECT FOR SMALLER QUANTITY INPUT AND OUTPUT REQUIREMENTS.

The Hono™ AVB 4.4M (and 2.2M) receives four (or two) channels of AVB and sends them to balanced analog audio outputs, while simultaneously inputting four (or two) channels of mic/line level balanced audio and transmitting them as AVB. The Hono™ AVB 4.4D (and 2.2D) receives four (or two) channels of AVB and sends them to their AES/EBU audio outputs, while simultaneously inputting four (or two) channels of AES/EBU audio and transmitting them as AVB.

All models support 48kHz and 96kHz sample rates. The IEEE 1722.1 Discovery and Control protocol allows the Hono to be configured using the AudioScience AVB Controller and 3rd party AVB controllers, such as Riedel AVB Manager. DSP functionality includes a parametric equalizer and compander/limiter on the inputs and programmable delays on the outputs as well as full matrix mixing.

BLOCK DIAGRAM



FEATURES

Inputs

- Hono™ AVB 4.4M and 2.2M: Four or two balanced analog mic/line inputs
- Hono™ AVB 4.4D and 2.2D: Four or two AES/EBU inputs
- Software adjustable, non-volatile, input levels from –50 to +24dBu
- 100dB DNR, -90dB THD+N, -110dBu EIN
- Software selectable +48V phantom power individually available on all inputs
- 3.81mm pluggable terminal block connectors

Outputs

- Hono™ AVB 4.4M and 2.2M: Four or two balanced analog line outputs
- Hono™ AVB 4.4D and 2.2D: Four or two AES/EBU outputs
- Software adjustable, non-volatile output levels from -10 to +24dBu

GPIO

- Four opto-isolated inputs
- Four relay isolated outputs

AVB

- Four or two channels of AVB in and out
- 4 AVB streams in and out
- Stream formats of 1, 2, 4, 8, 16, 24 and 32 channels
- Dedicated media clock input and output stream
- AVnu certified

Power

- Power over Ethernet (PoE) 802.3af compliant
- External +5V power supply if POE not being used

Chassis

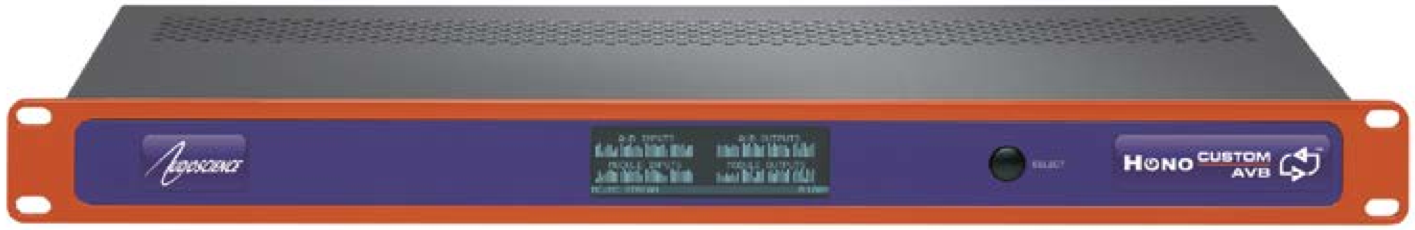
- Rack mountable using optional 1U front panel
- Wings allow easy mounting
- 5.25 inches W x 3.125 inches L x 1.37 inches H

Control

- Controllable from ASIControl, Hono AVB Controller and 3rd party

Model	Audio Network	Audio Channels In	Audio Channels Out	Input Type	Output Type	Connectors
Hono™ AVB 2.2M	AVB	2	2	Balanced Analog Mic/Line	Balanced Analog Line	Terminal Block
Hono™ AVB 4.4M	AVB	4	4	Balanced Analog Mic/Line	Balanced Analog Line	Terminal Block
Hono™ AVB 2.2D	AVB	2	2	AES/EBU	AES/EBU	Terminal Block
Hono™ AVB 4.4D	AVB	4	4	AES/EBU	AES/EBU	Terminal Block

HONO AVB CUSTOM



AUDIOSCIENCE PROVIDES UNIQUE BESPOKE CONFIGURATIONS. From our website's configuration tool, you can easily and quickly design the exact I/O combination that your application requires. You can even specify the connector types: Terminal Blocks, XLR on breakout cable, RJ-45, or ¼ inch jacks (mic-line only).

These custom audio interfaces are built in a 1U rackmount format and provide up to 32 channels of AVB receive and transmit. The units can be populated with up to four function specific modules, allowing up to 32 channels of analog or AES/EBU I/O. Each module has an interchangeable connector that may be configured with either a pluggable terminal block, StudioHub+®, 50pin Centronics connector with XLR breakout cables, or ¼ inch jacks. Each Hono™ AVB custom device features a powerful Texas Instruments 32bit floating point DSP that allows sophisticated switching/mixing. A graphics display on the unit's front panel shows peak meters and status. AudioScience provides application software, ASIControl, that may be used to set up the unit. Controllers such as Hono AVB Controller may be used to set up AVB routing connections between the Hono™ Series units and any other AVB device on the network.

Connector and I/O module combined

FEATURES

AVB

- 32 channels of AVB in and out
- 4 AVB streams in and out
- Dedicated media clock input and output stream
- Interoperable with all AudioScience AVB products and other 3rd party AVB equipment using the IEEE 1722.1 control protocol
- Stream formats of 1, 2, 4, 8, 16, 24 and 32 channels
- AVnu certified

I/O

- Modular architecture allows up to 4 I/O modules to be inserted into the back of the unit
- Module connector options include Terminal Block (Phoenix style), StudioHub+® RJ-45, or 50pin Centronics connector with XLR breakout cables
- Available modules include 8 channel analog I/O, 8 channel AES/EBU I/O, eight-channel microphone preamp and 16x16 GPIO.

Signal Processing

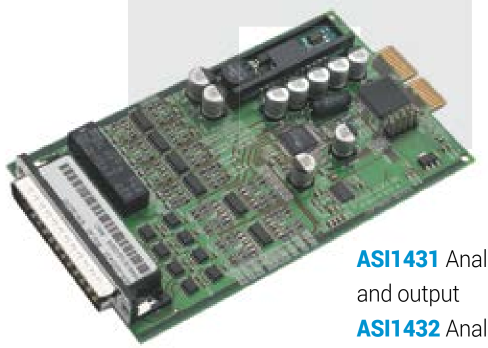
- Metering and up to 20dB gain on all signal paths
- EQ, Compressor/Limiter on microphone inputs

Power

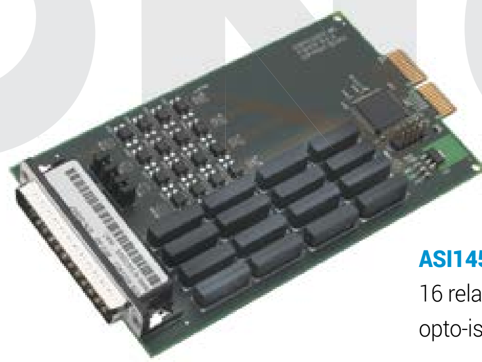
- Built-in 90-260VAC power supply

Rear view of Hono AVB Custom

I/O MODULES



ASI1431 Analog 8 channel input and output
ASI1432 Analog 8 channel input
ASI1433 Analog 8 channel output



ASI1451 GPIO Module with 16 relay outputs and 16 opto-isolated inputs

ASI1441 AES/EBU 8 channel input and output
ASI1442 AES/EBU 8 channel input
ASI1443 AES/EBU 8 channel output

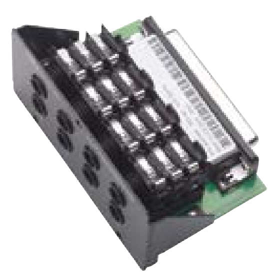
ASI1462 8-channel balanced microphone preamp with 48V phantom supply
ASI1464 8-channel un-balanced microphone preamp with 12V phantom supply

CONNECTOR MODULES

ASI1491
XLR via 50-pin Centronics

ASI1493
Terminal block

ASI1492
StudioHub



ASI1494
1/4" TRS
8 input only

BROADCAST MPEG SOUND CARDS

AUDIOSCIENCE HAS TAKEN SOUND CARD TECHNOLOGY A STEP FURTHER WITH OUR LOW PROFILE SOUNDCARDS BUILT SPECIFICALLY FOR THE BROADCAST INDUSTRY. Our ASI6700 series utilizes the PCI Express interface, for faster bus speeds and compatibility with the latest systems.

All our MPEG sound cards feature essential broadcast-centric technologies, including SoundGuard, TSX time scaling, MRX multi-rate mixing and full MPEG layer 2 and MP3 support.

FEATURES

- Up to 24 stereo streams of playback into up to 8 stereo outputs (depending on model)
- Up to 8 stereo streams of record from up to 4 stereo inputs (depending on model)
- Formats include PCM, MPEG layer 2 and MP3 with sample rates to 96kHz
- MRX™ technology supports digital mixing of multiple stream formats and sample rates
- TSX™ time scaling allows compression/expansion of play streams by up to +/-20% with no pitch shift (on supported models)
- SSX™ mode for multichannel record, playback and mixing (on supported models)
- Balanced stereo analog inputs and outputs with levels to +24dBu
- 24bit ADC and DAC with 110dB DNR and 0.0015% THD+N
- AES/EBU inputs and outputs with sample rate converters on all inputs (on supported models)
- Dedicated AES/EBU and Word clock Sync input (on supported models)
- SoundGuard™ transient voltage suppression on all I/O
- Short length PCI card format (5.4"/138mm)
- Full height bracket available
- Up to 8 cards in 1 system
- 32/64-bit Windows 11, 10, 7, Server 2008/2012, and Linux software drivers available

Product	ASI6714	ASI6718	ASI6720	ASI6722	ASI6740	ASI6744	ASI6788
Interface	PCI Express	PCI Express	PCI Express	PCI Express	PCI Express	PCI Express	PCI Express
Inputs	1 Stereo/2 Mono	1 Stereo/2 Mono	2 Stereo/4 Mono	2 Stereo/4 Mono	4 Stereo/8 Mono	4 Stereo/8 Mono	8 Stereo/16 Mono
Outputs	4 Stereo/8 Mono	8 Stereo/16 Mono	2 Stereo/4 Mono	2 Stereo/4 Mono	4 Stereo/8 Mono	4 Stereo/8 Mono	8 Stereo/16 Mono
Record Streams	1 or 2	1 or 2	2 or 4	2 or 4	4 or 8	4 or 8	8 or 16
Play Streams	4 or 12	8 or 24	2 or 6	2 or 6	4 or 12	4 or 12	8 or 24
Compression Formats	PCM, MPEG Layer 2, MP3	PCM, MPEG Layer 2, MP3	PCM, MPEG Layer 2, MP3	PCM, MPEG Layer 2, MP3	PCM, MPEG Layer 2, MP3	PCM, MPEG Layer 2, MP3	PCM, MPEG Layer 2, MP3
MRX™	•	•	•	•	•	•	•
TSX™	•	•	•	•	•	•	•
Sample Rates	8-96kHz	8-96kHz	8-96kHz	8-96kHz	8-96kHz	8-96kHz	8-96kHz
Analog	•	•	•	•	•	•	•
AES/EBU	•	•		•		•	•
SSX2 Multichannel	•	•	•	•	•	•	•
GPIO	16 in, 4 out	16 in, 4 out	8 in, 2 out	8 in, 2 out	16 in, 4 out	16 in, 4 out	16 in, 4 out
Max Cards/System	8	8	8	8	8	8	8
Size	5.4" X 2.75"	5.4" X 2.75"	5.4" X 2.75"	5.4" X 2.75"	5.4" X 2.75"	5.4" X 2.75"	5.4" X 2.75"
Applications							
Radio Automation	•	•	•	•	•	•	•
Radio Production	•	•	•	•	•	•	•

PCM SOUND CARDS

When on-board compression is not needed, the ASI5700 and ASI5800 series provide multichannel playback and record using the PCI Express interface. Both balanced analog and AES/EBU I/O are offered.

FEATURES

- Up to 24 stereo streams of playback into up to 8 stereo outputs (depending on model)
- Up to 8 stereo streams of record from up to 4 stereo inputs (depending on model)
- Formats include 8, 16, 24 and 32 bit PCM with sample rates from 32kHz to 96kHz
- SSX™ mode for multichannel record, playback and mixing (on supported models)
- Balanced stereo analog inputs and outputs with levels to +24dBu
- 24bit ADC and DAC with 110dB DNR and 0.0015% THD+N
- AES/EBU inputs and outputs with sample rate converters on all inputs (on supported models)
- Dedicated AES/EBU and Word clock Sync input (on supported models)
- SoundGuard™ transient voltage suppression on all I/O
- Short length PCI card format (5.4"/138mm)
- Up to 8 cards in 1 system
- 32/64-bit Windows 11, 10, 7, Server 2008/2012, and Linux software drivers available

Product	ASI5720	ASI5722	ASI5740	ASI5741	ASI5744	ASI5780	ASI5788
Interface	PCI Express	PCI Express	PCI Express	PCI Express	PCI Express	PCI Express	PCI Express
Inputs	2 Stereo/ 4 Mono	2 Stereo/ 4 Mono	4 Stereo/ 8 Mono	4 Stereo AES	4 Stereo/ 8 Mono	1 Stereo/ 2 Mono	8 Stereo/ 16 Mono
Outputs	2 Stereo/ 4 Mono	2 Stereo/ 4 Mono	4 Stereo/ 8 Mono	4 Stereo AES	4 Stereo/ 8 Mono	8 Stereo/ 16 Mono	8 Stereo/ 16 Mono
Record Streams	2 or 4	2 or 4	4 or 8	4 or 8	4 or 8	1 or 2	8 or 16
Play Streams	4 or 6	4 or 6	4 or 12	4 or 12	4 or 12	8 or 24	8 or 24
Compression Formats	PCM	PCM	PCM	PCM	PCM	PCM	PCM
Sample Rates	32-96kHz	32-96kHz	32-96kHz	32-96kHz	32-96kHz	32-96kHz	32-96kHz
Analog	•	•	•		•	•	•
AES/EBU		•		•	•		•
SSX Multichannel			•	•	•	•	•
Max Cards/ System	8	8	8	8	8	8	8
Size	5.4" X 2.75"	5.4" X 2.75"	5.4" X 2.75"	5.4" X 2.75"	5.4" X 2.75"	5.4" X 2.75"	5.4" X 2.75"
Applications							
Radio Broadcast	•	•	•	•	•	•	•
Installed Sound	•	•	•	•	•	•	•

PRODUCTION SOUND CARDS

THE ASI5810, ASI5811 AND ASI5812 ARE DESIGNED FOR USE IN RADIO BROADCAST PRODUCTION, UTILIZING THE PCI EXPRESS INTERFACE. These adapters offer 2 stereo record streams from either a balanced analog input or AES/EBU digital input and 4 stereo play streams mixed to both a balanced analog output and an AES/EBU digital output. The ASI5811 also includes a microphone input with low noise pre-amp and a 48V phantom supply.

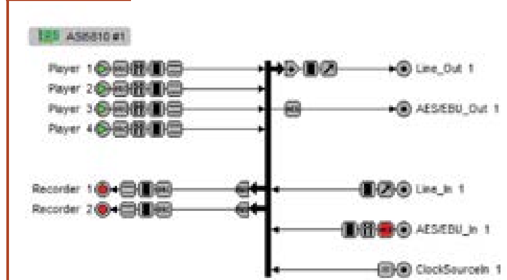
FEATURES

- Four stereo streams of PCM playback
- Two stereo streams of PCM record
- Balanced stereo analog input and output with +24dBu I/O
- PCIe interface
- Four opto inputs and two relay outputs (ASI5810, ASI5811)
- AES/EBU digital input and output with hardware SRC on the input
- Low noise microphone input with 48V phantom supply and DSP based compressor/limiter and 5-band equalizer (ASI5811 only)
- 24bit analog-to-digital and digital-to-analog converters - 100dB SNR and 0.0025% THD+N
- MRX multi-rate mixing technology supports digital mixing of multiple sample rates
- SoundGuard™ transient voltage suppression protects against lightning and other high voltage surges on all I/O
- Up to 8 cards in one system
- 32/64-bit Windows 11, 10, 7, Server 2008/2012, and Linux software drivers available

Product	ASI5810	ASI5811	ASI5812
Interface	PCI Express	PCI Express	PCI Express
Inputs	1 Stereo/2 Mono	1 Stereo/2 Mono + Microphone	1 Stereo/2 Mono
Outputs	1 Stereo/2 Mono	1 Stereo/2 Mono	1 Stereo/2 Mono Analog/2 AES
Record Streams	1 or 2	1 or 2	1 or 2
Play Streams	1 or 4	1 or 4	1 or 4
GPIO	•	•	•
Compression Formats	8, 16, 24, 32 bit PCM	8, 16, 24, 32 bit PCM	8, 16, 24, 32 bit PCM
MRX™	•	•	•
Sample Rates	8 to 192kHz	8 to 192kHz	8 to 192kHz
Analog	•	•	•
AES/EBU	•	•	•
Max Cards/System	8	8	8
Size	4.4" X 2.75"	4.4" X 2.75"	4.4" X 2.75"
Applications			
Radio Broadcast	•	•	•
Radio Production	•	•	•

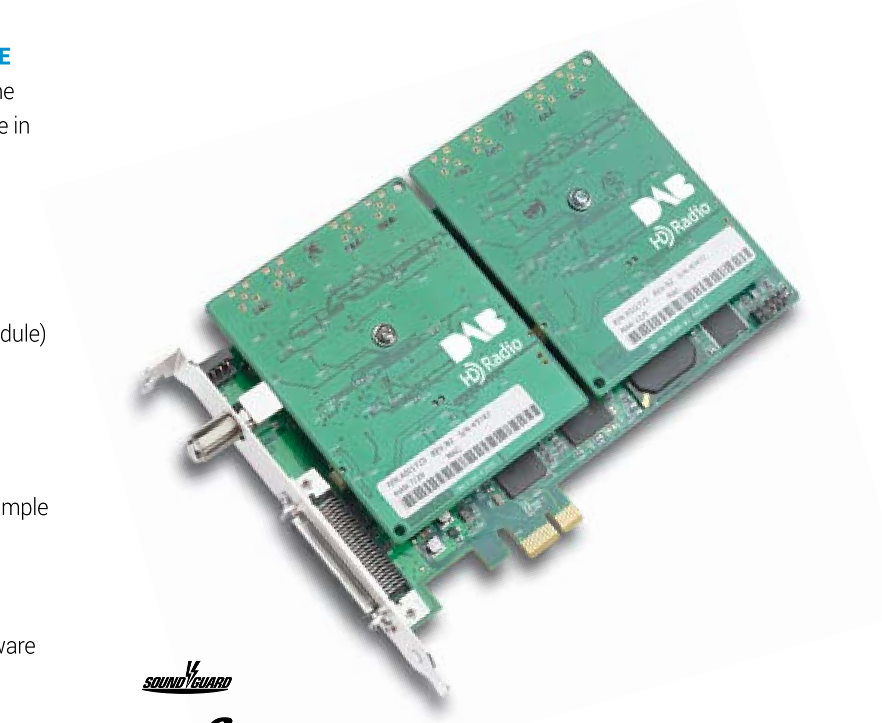
BLOCK DIAGRAM

ASI5810 – 4-PLAY MODE



TUNER CARDS

AUDIOSCIENCE TUNER SOUNDCARDS ARE DESIGNED FOR USE IN RADIO BROADCAST AUDIO MONITORING AND AUDITING. The ASI8821 provides up to 8 channels of AM/FM/RDS audio capture in PCI Express interfaces.



FEATURES

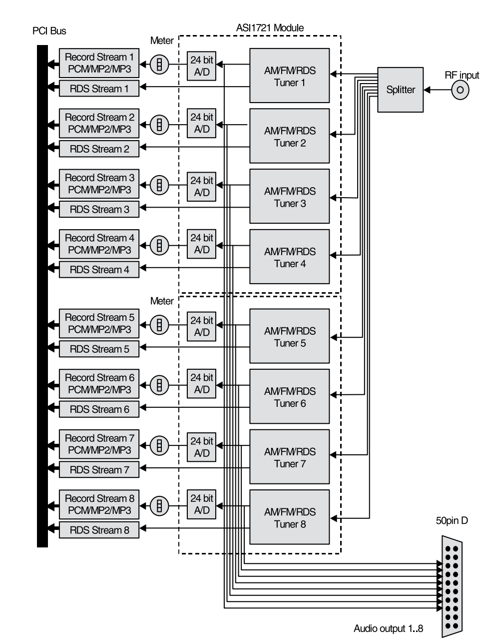
- Up to 8 channels of HD-Radio or DAB/DAB+ (with appropriate module)
- Up to 8 channels of AM/FM audio capture (with appropriate module)
- PCIe interface
- Up to 8 channels of FM RBDS/RDS data capture
- AM/FM tuners can be fed from individual external antennas
- Audio monitoring of all tuners simultaneously
- MRX technology allows each stream to have an independent sample rate of between 8 and 48kHz
- PCM and MPEG-1 Layer 2 and MP3 recording formats
- Up to 8 cards in one system
- 32/64-bit Windows 11, 10, 7, Server 2008/2012, and Linux software drivers available

For full specifications, visit www.audioscience.com

Product	ASI8821
Interface	PCI Express
Inputs	RF
Tuners	4 or 8 HD-Radio/ DAB/DAB+/FM 4 or 8 AM/FM/RDS
Record Streams	4 or 8
Compression Formats	PCM, MPEG Layer 2, MP3
Sample Rates	8-48kHz
Max Cards/System	8
Size	3.9" x 6.6"
Applications	
Broadcast Monitoring	•
Broadcast Logging	•

BLOCK DIAGRAM

ASI8821 8-STREAM MODE



OEM PRODUCTS

AUDIOSCIENCE HAS THE CAPABILITY AND EXPERTISE TO DESIGN AND MANUFACTURE CUSTOM AUDIO PRODUCTS. AudioScience has been busy behind the scenes working with our OEM partners on a variety of exciting new audio projects. Looking to bring your ideas to life? Take advantage of our decades of experience in designing and building audio gear. Contact our sales team today and let us help you create the products that will power your company's future For more information, please contact Richard Gross @ 1-302-235-7109

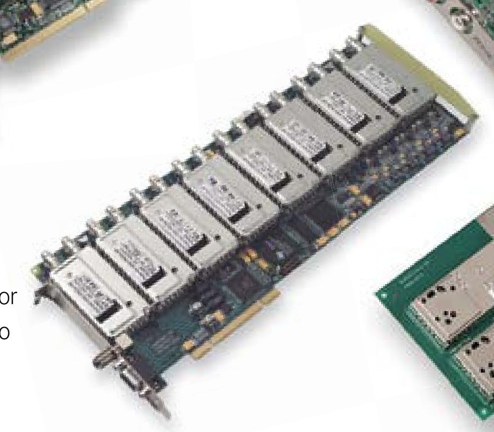
BEINCOURT

LEGACY PRODUCTS

ASI4601
MPEG audio adapter for Prophet Systems/Sirius satellite radio

ASI4401
Audio adapter for in-store advertising

BLU-PCI
Custom BLU Link network audio card



ASI8801
Audio adapter for TV and FM radio advertisement monitoring

ASI8401
Audio adapter for TV and FM radio advertisement monitoring

BREAK-OUT BOXES

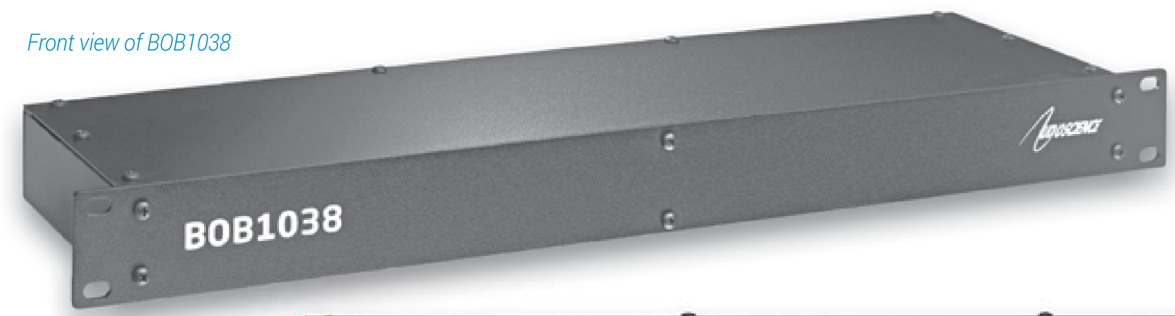
THE BOB1038 AND HUB-12 ARE 1U RACKMOUNT BREAK-OUT BOXES THAT TERMINATE THE ANALOG OR AES/EBU CONNECTORS ON SUPPORTED AUDIOSCIENCE ADAPTERS.

BOB1038 provides eight stereo in and eight stereo out I/O and GPIO, allowing the full functionality of the ASI5788/ASI6788 to be realized. Breakouts use 3.81mm terminal blocks with screw-locks. It can be operated as either an analog and GPIO breakout or an AES/EBU and GPIO breakout. Adapter cables allow the ASI5600, ASI6600, ASI5700 and ASI6700 family to be used.

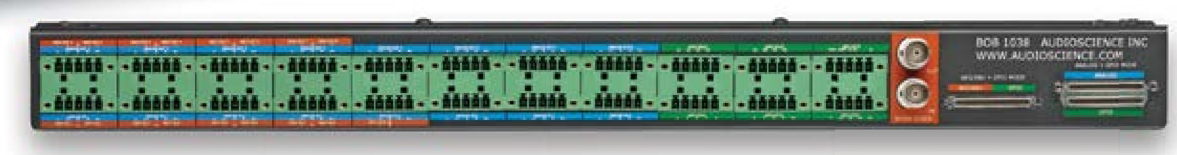
HUB-12 (formerly BOB1025) provides balanced I/O on StudioHub (RJ45) connectors. It can be operated as either an analog breakout (4 stereo in and 8 stereo out) or AES/EBU digital breakout (4 stereo in, 6 stereo out and sync in/out). Connections to the audio adapter are via a 50-pin high density SCSI type connector for analog and a 26-pin high density connector for digital (AES/EBU). Purchase from Angry Audio at www.angryaudio.com.

The HUB-12 includes the necessary cables to connect to AudioScience sound cards.

Front view of BOB1038



Rear view of BOB1038



CABLES

AudioScience provides a variety of broadcast-quality cables that provide XLR terminations for both the analog and AES/EBU connectors on our sound cards. Connections include DB-9, Mini DB50 and Centronics 50-pin.

Cable	Card	Description
CBL1044		50-pin to XLR, balanced analog, 4 stereo in, 4 stereo out. Needs to be paired with CBL4004.
CBL1144	ASI6744	50-pin Centronics to XLR, AES/EBU, 4 stereo in, 4 stereo out, AES sync in, word clock I/O (BNC). Needs to be paired with CBL4104.
CBL1311	ASI5811	HD DB-26 to analog XLR, AES/EBU XLR and GPIO on DB15
CBL1312	ASI581x	
CBL4004	ASI57xx ASI67xx	VHDCI to Centronics 50-pin adapter, analog
CBL4011	ASI5711 ASI6711	VHDCI to 1 stereo in, 1 stereo out, analog XLR
CBL4022	ASI572x ASI672x	VHDCI to 2 stereo in, 2 stereo out, analog XLR
CBL4104	ASI57xx ASI67xx	VHDCI to Centronics 50-pin adapter, AES/EBU and GPIO
CBL4111	ASI5711 ASI6711	VHDCI to 1 in, 1 out, Sync in, word clock I/O, AES/EBU XLR
CBL4122	ASI5722 ASI6722	VHDCI to 2 in, 2 out, Sync in, word clock I/O, AES/EBU XLR

DRIVERS

AUDIOSCIENCE IS THE AUDIO I/O COMPANY.

SOFTWARE DRIVERS

AudioScience offers a complete range of audio driver interfaces for all of our audio adapters. Operating system support includes 32/64-bit Windows 11, 10, 7, Server 2008/2012 and Linux. The sections below outline the specific interfaces supported.

Windows WAVE, WDM and Combo

The AudioScience Microsoft Windows drivers enable multi-stream recording, reproduction and mixing of digital audio on a PC platform. The Microsoft multimedia wave and mixer APIs are supported under 32/64-bit Windows 11, 10, 7, and Server 2008/2012. The drivers utilize large adapter buffers to provide high performance, glitch-free audio under all operating conditions.

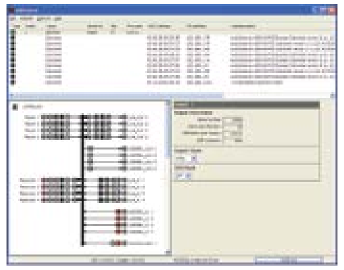
There are 3 versions of the Windows driver. The WAVE driver supports the waveOut, waveIn and mixer interfaces. It communicates directly with the hardware to support compressed audio playback/recording/mixing using the on-board DSP. The WDM driver supports the DirectSound interface for low-latency PCM only recording and playback. The Combo driver contains both the WAVE and WDM interfaces; a 64bit version is available. All drivers support AudioScience's ASX and HPI APIs, TSX™ time scaling, and SSX™/SSX2™ extensions.

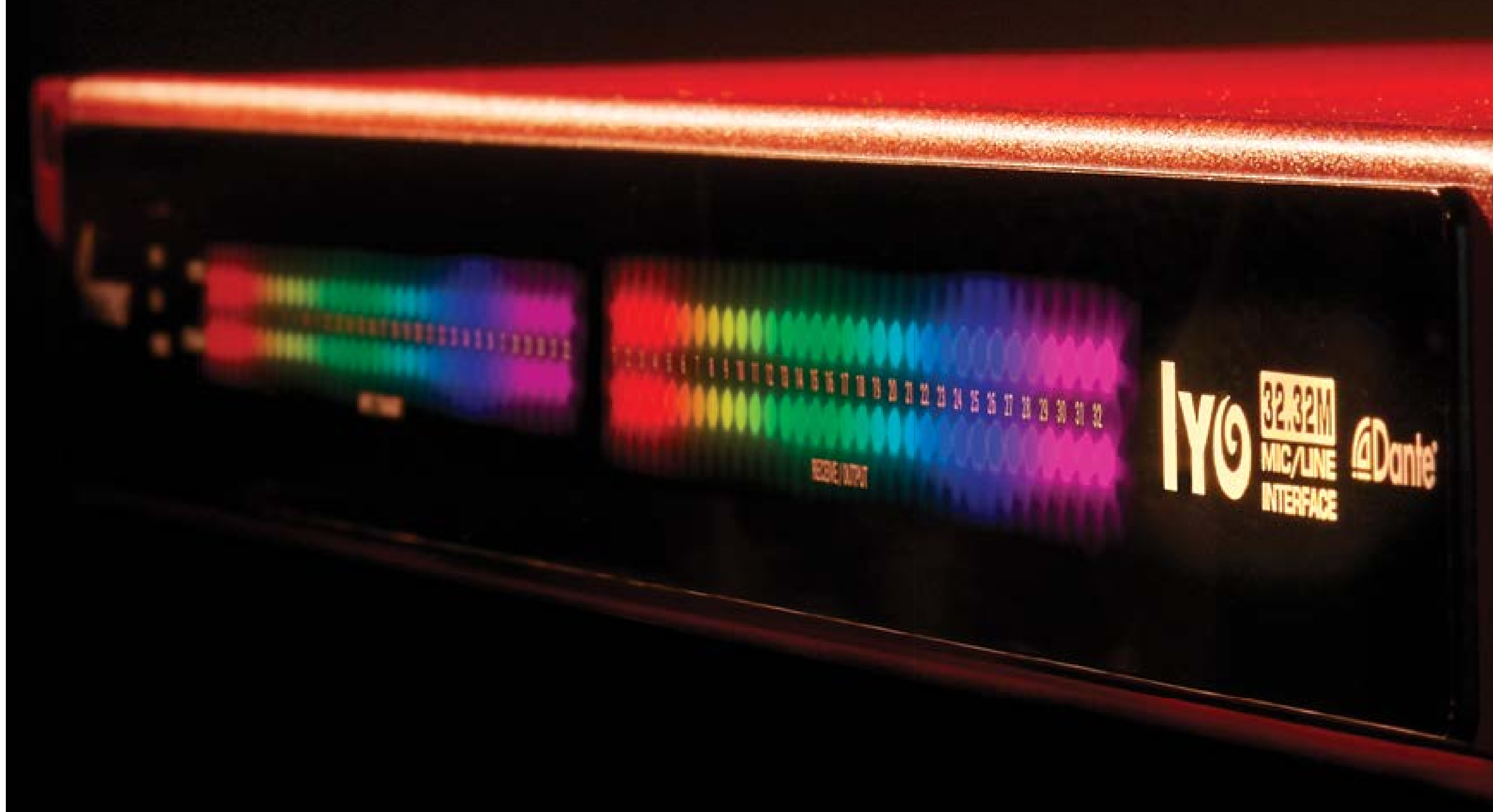
ASIO

The AudioScience Audio Stream In/Out (ASIO) 2.0 driver enables multi-track recording, reproduction, and mixing of PCM digital audio on a PC platform under the 32/64-bit Windows 11, 10, 7, and Server 2008/2012 OS. This driver follows the ASIO philosophy of providing a simple low latency PCM audio pass-through from the hardware to the application. The ASIO driver interface is integrated into the AudioScience Windows WAVE, WDM or Combo drivers; it's built in and installs automatically when one of our drivers is installed. As such, ASIO applications may share the audio hardware's resources with other audio applications that use WAVE, DirectSound,® or the HPI interfaces.

Linux ALSA/HPI

The AudioScience Linux drivers enable multi-stream recording, reproduction, and mixing of multi-stream digital audio on a PC running the Linux operating system. The AudioScience HPI and ASX audio APIs are supported under various kernel versions and distributions including kernel 2.6. Additionally ALSA supports AudioScience cards with the "snd-asihpi" module. The driver is released under GPL, the libraries under a modified BSD style license.





CORPORATE AND SALES

7209 Lancaster Pike #4-323
Hockessin, Delaware 19707
Sales Toll free: 1-855-AUDIOSC (283-4672)
Phone: +1-302-235-7109
FAX: +1-302-235-7110
salesasi@audioscience.com
Language: English

TECHNICAL SUPPORT

Rochester, NY
585-271-8870 (9AM-5PM EST)
949-650-6263 (2PM-5PM PST)
support@audioscience.com
Language: English

ENGINEERING AND MANUFACTURING

760 W. 16th St, Bldg. L
Costa Mesa, California 92627
Phone: +1-949-650-6263
FAX: +1-949-650-6291
engineering@audioscience.com

AUDIOSCIENCE ASIA

222 Jalan Eunos
EuHabitat, #03-105
Singapore 415871
Phone: +65 6209 6252
Fax : +65 6383 0401
salesasia@audioscience.com
Languages: Simplified Chinese and English

AUDIOSCIENCE EUROPE

Malaga, Spain
Phone: +44 203 287 3541
saleseu@audioscience.com
Languages: English, Spanish, Italian

AUDIOSCIENCE was founded in 1996 to provide high-level design and manufacturing solutions to strategically targeted areas of the digital audio reseller market. Over the last 26 years AudioScience has pioneered ground-breaking innovations in soundcard and software design, starting with the first DSP-based digital audio adapter ever offered to the digital broadcast market. AudioScience's four main product areas include Broadcast Sound Cards for digital automation for radio, Tuner Cards for monitoring and verification of AM, FM and HD Radio™, Network Peripherals utilizing industry standard network protocols, and Custom OEM Products.

STANDARDS-BASED TECHNOLOGY Instead of using proprietary designs that limit choice and flexibility, AudioScience developed Standards-Based Technology. This ensures maximum compatibility of AudioScience products with other products and systems, including our competition.

TECHNICAL INNOVATION AudioScience developed its exclusive MRX Multi-Rate Mixing technology to allow the playback and mixing of stereo streams of MPEG Layer II & MP3 audio with different sample rates, while recording stereo streams at different sample rates. Our Standards-Based Technology allows AudioScience soundcards to integrate seamlessly into nearly any professional audio environment and to be compatible with the most popular broadcast software and hardware. AudioScience uses large RAM buffers to provide users with consistent, glitch-free digital audio playback, even in the most data-intensive applications.

LOOKING FORWARD "With our technical expertise, we absolutely have the ability to provide a better product that does more for less," says founding partner Stephen Turner. "We leverage our combined design experience and market knowledge to deliver products our customers will need next year." AudioScience is committed to staying ahead of the curve for the next 20 years, and beyond.