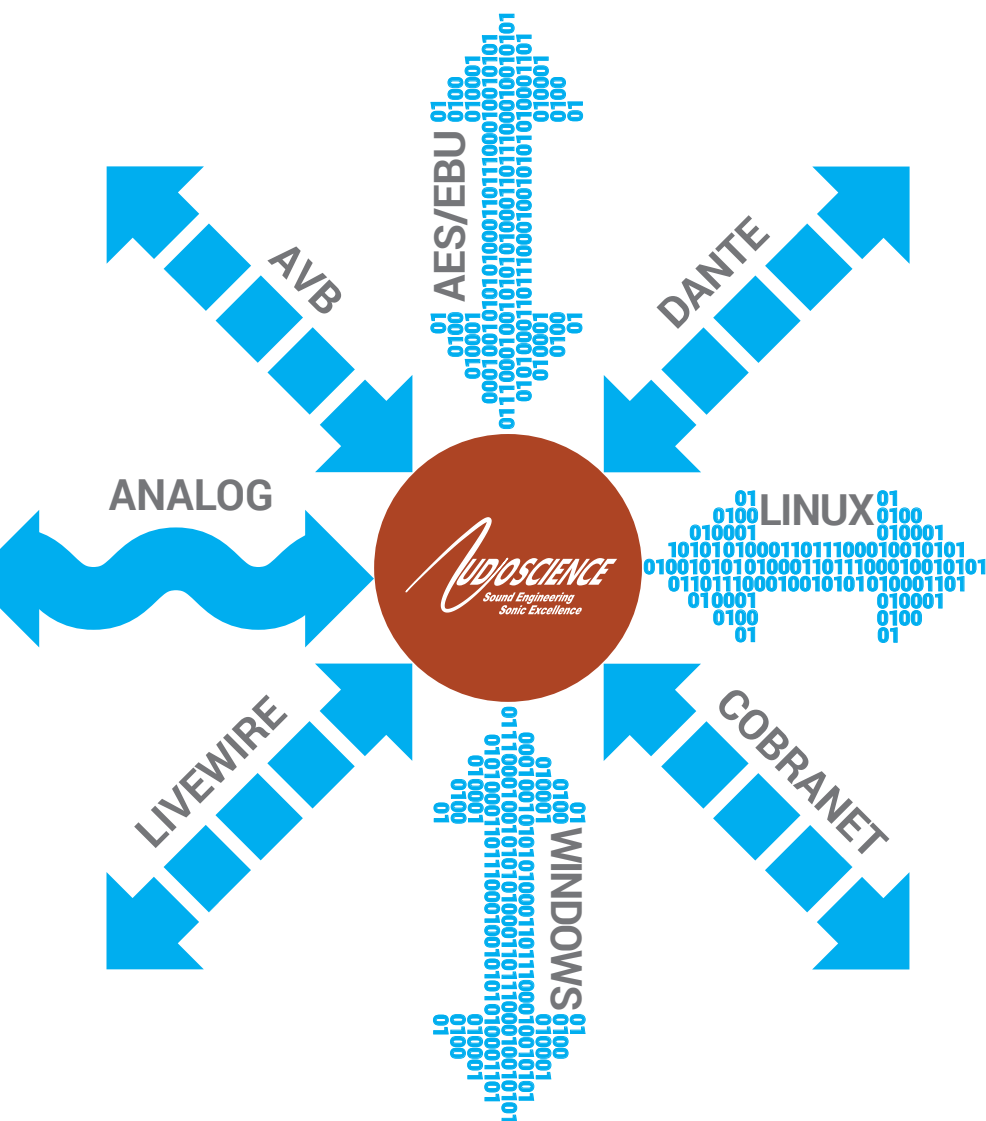


THE AUDIO I/O COMPANY.



LATEST PRODUCTS



AUDIOSCIENCE IS THE AUDIO I/O COMPANY.

For 22 years we have been getting audio in and out of our customers' systems.

Starting with broadcast radio, our PCI Express sound cards provide the conduit for getting analog and AES/EBU audio from the automation system into the studio and on its way to the transmitter.

In 2005 we adopted the first successful audio over network standard, CobraNet, developing products to get this protocol in and out of PCs and to convert it to and from analog and AES/EBU. We then added Audio Video Bridging (AVB), a set of open protocols developed by the IEEE and certified by Avnu.

Now, in our third decade, AudioScience is introducing the Iyo, a family of Dante products designed with all the essential functionality to get microphone and line level audio on and off the Dante and AES67 networks.

CONTENTS

- 4 Iyo Dante®
- 6 Hono™ AVB Controller
- 8 Hono™ AVB Virtual Sound Card (VSC)
- 9 Hono™ AVB Mini
- 10 Hono™ AVB Custom
- 12 Hono™ CobraNet Custom
- 13 Hono™ CobraNet Fixed
- 14 Hono™ CobraNet Mini
- 15 Network Sound Cards
- 16 Broadcast MPEG Sound Cards
- 17 Broadcast Low Profile Sound Cards
- 18 PCM Sound Cards
- 19 PCM Low Profile Sound Cards
- 20 Production Sound Cards
- 21 Processing Sound Card
- 22 Tuner Cards
- 23 OEM Cards
- 24 Breakout Boxes
- 25 Cables
- 26 Software Drivers and Applications



IYO DANTE

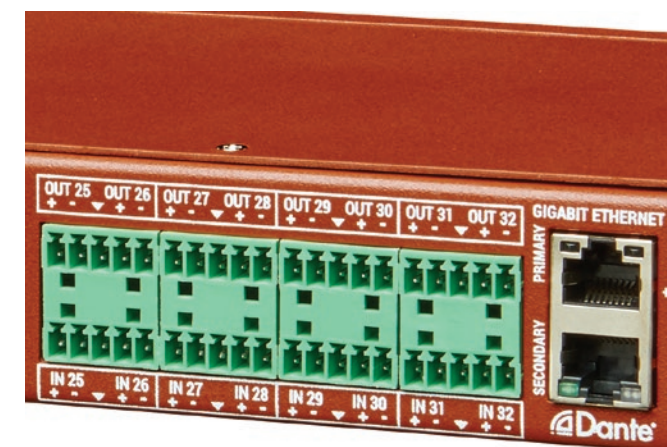
The Iyo Dante® is a cost-effective family of microphone/line Dante audio-over-IP (AoIP) interfaces in a 1U rack mount format.

Three models provide, 8x8, 16x16 or 32x32 balanced analog audio inputs and outputs. Each input accommodates microphone through line level signals. Phantom power is individually switchable on each input.

RGB LEDs on the Iyo's front panel show per channel audio levels and streaming status, while an embedded web server allows configuration and monitoring of input and output levels.

Power is provided from a built-in universal AC power supply. Redundant power is available using an external 12VDC supply via a locking 3.5mm jack.

All units can also be operated in AES67 interoperability mode.



ASI5812

The ASI5812 sound card is designed for use with PC based audio processing software for radio broadcast, such as Thimeo Stereo Tool and Telos OmniaSST.

These applications have the ability to generate both an analog FM multiplex (MPX) and an HD-Radio baseband signal. In the past this would have required two audio cards, one to generate the MPX signal at a 192kHz sample rate and a separate card to generate the HD-Radio signal in the digital domain at a 44.1kHz sample rate.

The ASI5812 provides this combined functionality using just one half-height sound card.

IYO DANTE

THE IYO DANTE® IS A COST-EFFECTIVE FAMILY OF MICROPHONE/LINE DANTE AUDIO-OVER-IP (AOIP) INTERFACES IN A 1U RACK MOUNT FORMAT. Three models provide, 8x8, 16x16 or 32x32 balanced analog audio inputs and outputs. Each input accommodates microphone through line level signals with a gain 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.

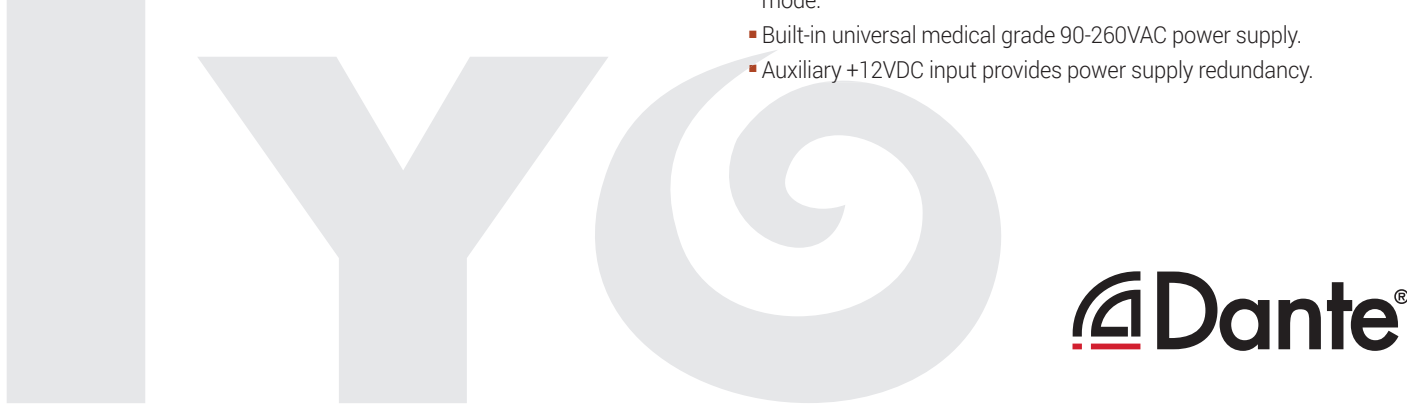
The Iyo family features an embedded web server, allowing configuration and monitoring of input and output levels.

Power is provided from a built-in universal AC power supply. Redundant power is available using an external 12VDC supply via a locking 3.5mm jack.

All units can also be operated in AES67 interoperability mode.

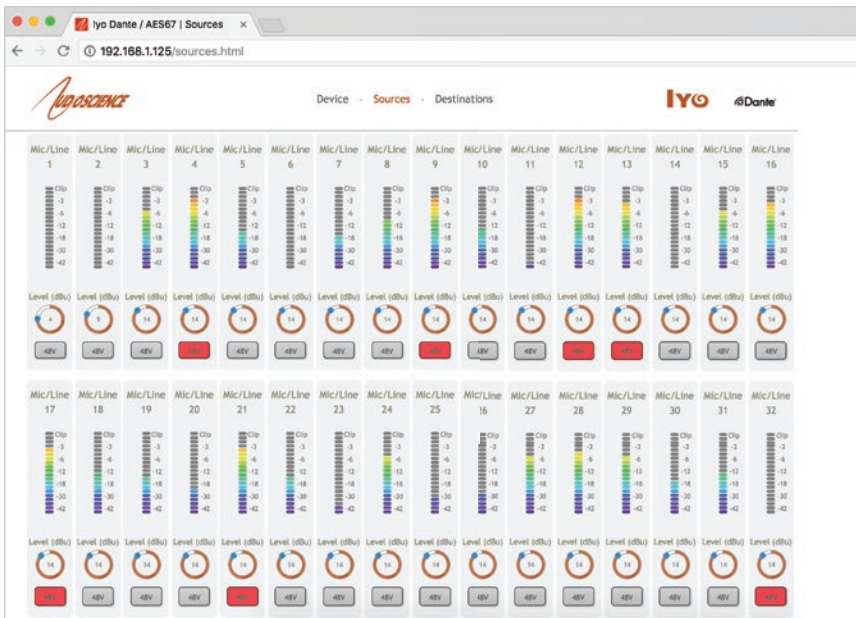
FEATURES

- 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



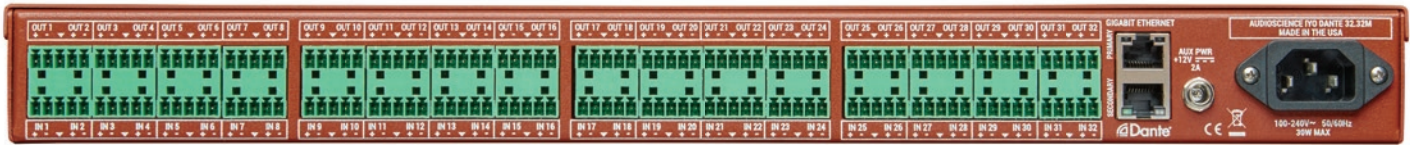
Iyo Dante 16.16M



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

HONO AVB CONTROLLER IS A SOFTWARE APPLICATION WHICH ALLOWS USERS TO CONFIGURE AND ROUTE IEEE1722.1 AVB STREAMS. The Controller displays all IEEE1722.1 AVB compatible devices and streams. Users can edit device settings, route streams, monitor network changes, and apply saved presets.

Per-device configurable settings include device name, sample rate, media clock, and media format. Dynamic stream mappings are supported for all AudioScience AVB devices, allowing individual channels to be selected from incoming streams (up to 32 channels) and routed to outgoing streams.

Helpful stream parameters are readily accessible to the user. Media Formats, Stream IDs, Receive Latency, and packet errors can be easily monitored by hovering the mouse over an applicable stream connection in the main Routing matrix.

The Controller provides extensive AVB diagnostic abilities,

including 802.1AS link delay, 802.1AS GrandMaster status, and 802.1AS GrandMaster transition count.

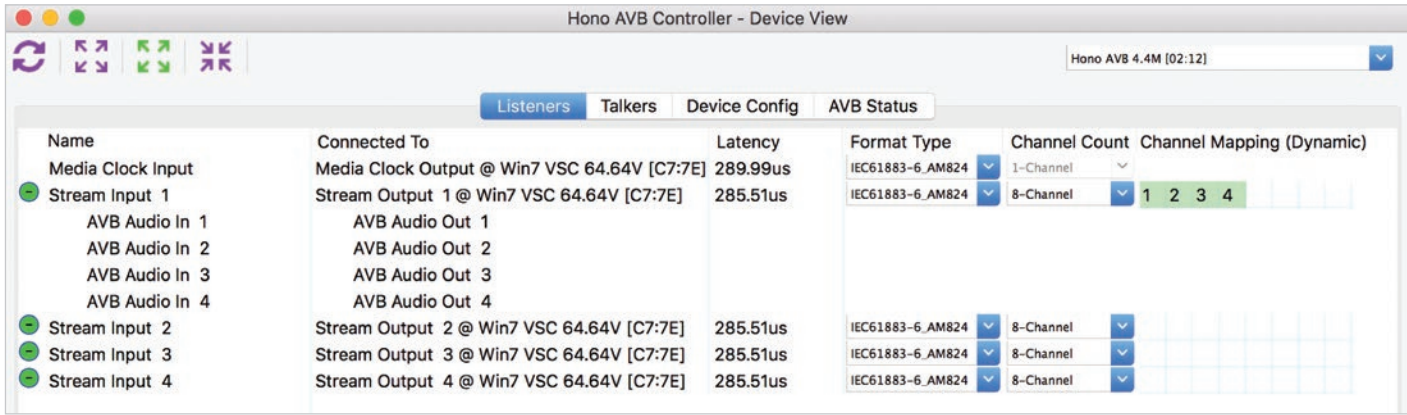
Presets allow a user to backup and restore an AVB network configuration. All applicable device settings and stream connections can be saved. The Controller provides the ability to apply one configuration setting to multiple devices at a time. Presets can be edited offline as necessary.

Hono AVB Controller is built upon the foundation of avdecc-lib, an open source AVB controller library. Avdecc-lib is part of the AVnu OpenAVB effort, an open source project for AVnu software, drivers and building blocks.

The Controller is free when used on a network containing at least one AudioScience AVB device. A paid option is available when using on a network containing only 3rd party AVB devices.

FEATURES

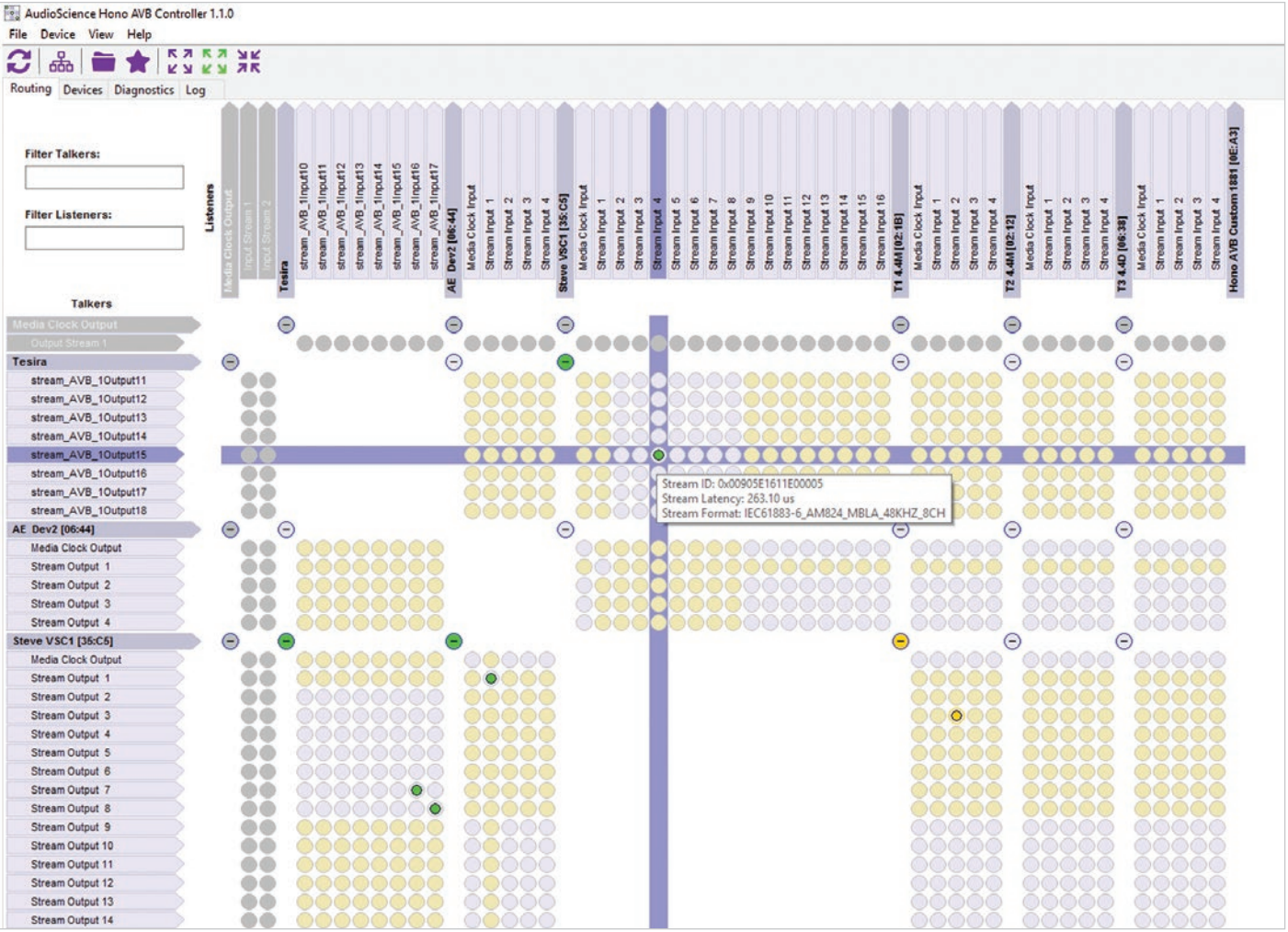
- View all IEEE 1722.1 enabled AVB devices and their streams.
- Route streams on AVB devices and examine existing stream routes.
- Presets
 - Save audio routing and device configuration presets
 - Apply previously saved presets
 - Edit presets offline and use for new device configurations
- View and adjust per-device settings, if implemented, including:
 - Lock/unlock devices
 - Device name
 - Device Sample rate
 - Media clock source
 - Network information
- View and adjust per stream settings including
 - Stream name
 - Media format
 - Audio channel mappings
 - IEEE 1722 Presentation time/stream latency
- View network status information, including IEEE 802.1AS clock synchronisation grandmaster
- View device error and notification logs
- Runs on Windows 7/10 and macOS platforms



AVB Device dialog allows configuration of stream format and channel mapping

AudioScience Hono AVB Controller 1.1.54													
File Device View Help													
<div><div><div><div></div><div></div><div></div><div></div></div><div><div></div><div></div><div></div><div></div></div><div><div></div><div></div><div></div><div></div></div><div><div></div><div></div><div></div><div></div></div></div><div><div></div><div></div><div></div><div></div></div><div><div></div><div></div><div></div><div></div></div><div><div></div><div></div><div></div><div></div></div></div>													
Routing		Devices	Diagnostics		Log								
State	Name	Group Name	Vendor	Model Name	MAC Address	Firmware Version	Entity ID	AVB	Grandmaster ID	GM Changes	pDelay (ns)	Link Up	Link Down
<div></div>	AE - Hono AVB 4.4D [06:38]	ASI	AudioScience, Inc.	Hono AVB 4.4D [06:38]	001C:F7:00:06:38	1.2.1	001CF7FFFF000638	<div></div>	000499FFFF7E1910	2	44	1	0
<div></div>	Hono AVB 4.4M [02:12]	ASI	AudioScience, Inc.	Hono AVB 4.4M [02:12]	001C:F7:00:02:12	1.2.1	001CF7FFFF000212	<div></div>	000499FFFF8EC939	1	57	1	0
<div></div>	FS AVB 64.64V [70:2F]	ASI	AudioScience, Inc.	Hono AVB 64.64V [70:2F]	A0:36:9F:66:70:2F	4.21.23	A0369FFFFF6702F	<div></div>	000499FFFF8EC93F	1	47	1	0
<div></div>	Hono AVB 64.64V [35:C5]	ASI	AudioScience, Inc.	Hono AVB 64.64V [35:C5]	0C:C4:7A:31:35:C5	4.21.23	0CC47AFFFF3135C5	<div></div>	000499FFFF9AF16C	1	46	1	0
<div></div>	Hono AVB Custom 1111 [0E:AC]	ASI	AudioScience, Inc.	Hono AVB Custom 1111 [0E:AC]	001C:F7:00:0E:AC	1.3.0-dev0	001CF7FFFF000EAC	<div></div>	000499FFFF9AF16C	1	77	1	0
<div></div>	Galileo-GALAXY	Unconfigured-Galileo-GALAXYs	Meyer Sound Laboratories, Inc.	Galileo-DEFAULT	001C:AB:00:6D:74	1.1.0-R4-2016-12-20-3219	001CABFFFF0006D74	<div></div>	000499FFFF9AF16C	0	84	1	0

Diagnostics tab shows AVB device state



Routing tab allows connection of AVB streams

HONO AVB VSC

MINI

AUDIOSCIENCE HAS LEVERAGED ITS EXPERTISE IN WINDOWS AUDIO DRIVERS AND NETWORK AUDIO TO DEVELOP THE HONO AVB VIRTUAL SOUND CARD. Available in three versions, our Hono AVB virtual soundcards feature standard WDM and ASIO interfaces and up to 64x64 AVB audio channels as well as an additional media clock input and output stream for syncing. They all utilize the RTX Real-time operating system from Interval Zero to deliver the same performance as high end digital audio hardware. Supporting the IEEE 1722.1 Discovery and Control protocol, our Virtual Soundcards can be configured using ASiControl, Hono AVB Controller and 3rd party AVB controllers. Windows 10 64bit is supported. All models support 48kHz and 96kHz sample rates.

FEATURES

Software Interfaces

- DirectSound, WAVE, ASIO and AudioScience HPI

AVB I/O

- Up to 64 channels in and out
- Up to 16 AVB streams in and out
- Stream formats of 1, 2, 4, 8, 16, 24 and 32 channels
- Dedicated media clock input and output stream

OS Support

- Windows 10 64bit

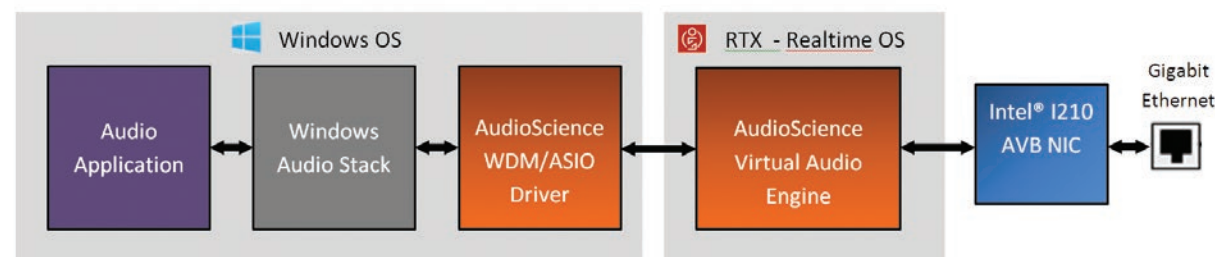
Control

- Controllable from ASiControl, Hono AVB Controller and 3rd party IEEE 1722.1 Controller software



IEEE 802.1 AVB

BLOCK DIAGRAM



Model	Audio Network	Audio Channels In	Audio Channels Out	AVB Streams In & Out	Software interfaces
Hono™ AVB 16.16V	AVB	16	16	8x8	DirectSound, WAVE, ASIO
Hono™ AVB 32.32V	AVB	32	32	8x8	DirectSound, WAVE, ASIO
Hono™ AVB 64.64V	AVB	64	64	16x16	DirectSound, WAVE, ASIO

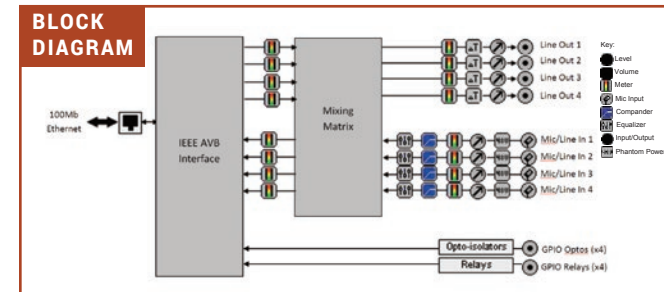


IEEE 802.1 AVB AVB

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.



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

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

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.

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

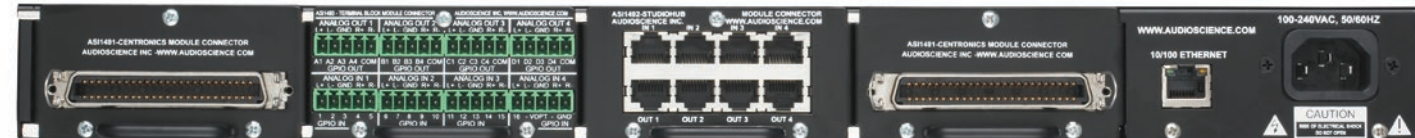
Power

- Built-in 90-260VAC power supply

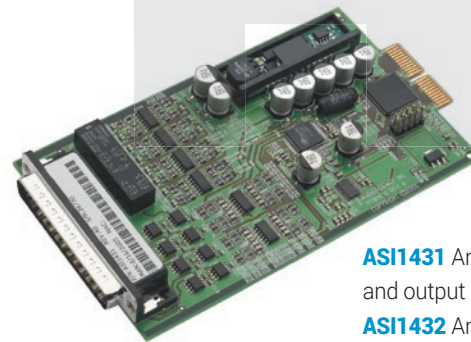


Connector and I/O module combined

Rear view of Hono AVB Custom



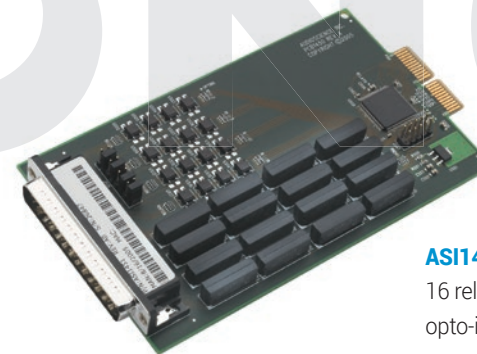
I/O MODULES



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

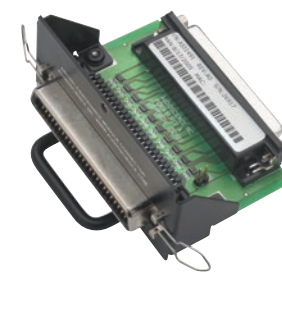
ASI1443 AES/EBU 8 channel output



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



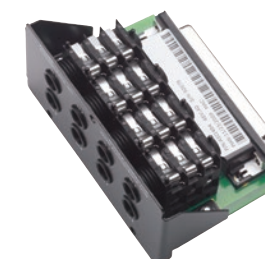
ASI1493

Terminal block



ASI1492

StudioHub



ASI1494

1/4" TRS
8 input only

HONO COBRANET

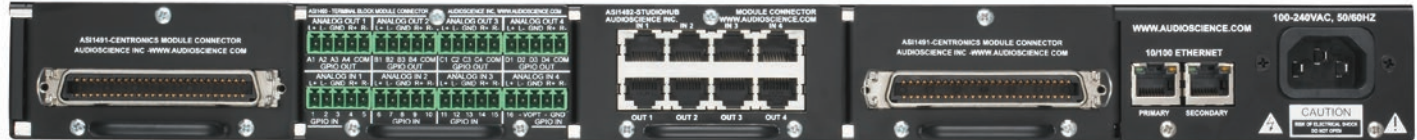
CUSTOM

FIXED



AUDIOSCIENCE PROVIDES UNIQUE BESPOKE CONFIGURATIONS WITH OUR ASI2416. 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 16 channels of CobraNet 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 terminalblock, StudioHub+®, 50pin Centronics connector with XLR breakout cables, or ¼ inch jacks. Each Hono™ CobraNet "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 CobraNet status. AudioScience provides application software that may be used to set up the unit. ASIControl sets up all internal features of the unit and allows CobraNet routing connections to be set up between the Hono™ Series units and any other CobraNet device on the network.

FEATURES

- 16 CobraNet® input channels and 16 CobraNet output channels on 100Mbit Ethernet with redundant RJ-45 connectors
- 4 CobraNet Transmitters and 8 CobraNet Receivers
- 1U rackmount unit
- Modular architecture allows up to 4 I/O modules to be inserted into the back of the unit.
- Modules connector options include Terminal Block (Phoenix style), StudioHub+® RJ-45, 50pin Centronics connector with XLR breakout cables, or ¼ inch jacks
- Available modules include 8 channel analog I/O, 8 channel AES/EBU I/O, 8 channel mic preamp and 16 channel GPIO
- Powerful floating point DSP provides metering, level control and up to 20 dB gain on all signal paths
 - Interoperable with all AudioScience CobraNet devices and other 3rd party CobraNet equipment
 - Built-in 90-260VAC power supply
 - Lua scripting support

I/O MODULES

Available modules are described on pg 11.

AUDIOSCIENCE'S HONO™ COBRANET SERIES HAS 12 PRE-CONFIGURED DEVICES FOR THE MOST POPULAR I/O COMBINATIONS.

These networked audio interfaces are built in a 1U rackmount format and provide up to 16 channels of CobraNet receive and transmit. These units feature 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 CobraNet status. AudioScience provides application software that may be used to set up all of the Hono™ Series units. ASIControl sets up all internal features of the unit and allows CobraNet routing connections to be set up between the Hono™ Series units and any other CobraNet device on the network.

FEATURES

- Up to 16 CobraNet input channels and 16 CobraNet output channels on 100Mbit Ethernet with redundant RJ-45 connectors
- 4 CobraNet Transmitters and 8 CobraNet Receivers
- 1U rackmount unit
- Powerful floating point DSP provides metering, level control, and up to 20 dB gain on all signal paths
- Interoperable with all AudioScience CobraNet Sound Cards and other 3rd party CobraNet equipment
- Built-in 90-260VAC power supply



Model	Network Protocol	Audio Channels In	Audio Channels Out	Input Type	Output Type	Connectors
Hono™ 8.0M	CobraNet	8	0	Balanced Analog Mic/Line	NA	Terminal Block
Hono™ 8.0L	CobraNet	8	0	Balanced Analog Line	NA	Terminal Block
Hono™ 16.0M	CobraNet	16	0	Balanced Analog Mic/Line	NA	Terminal Block
Hono™ 16.0L	CobraNet	16	0	Balanced Analog Line	NA	Terminal Block
Hono™ 0.8L	CobraNet	0	8	NA	Balanced Analog Line	Terminal Block
Hono™ 0.16L	CobraNet	0	16	NA	Balanced Analog Line	Terminal Block
Hono™ 8.8M	CobraNet	8	8	Balanced Analog Mic/Line	Balanced Analog Line	Terminal Block
Hono™ 8.8L	CobraNet	8	8	Balanced Analog Line	Balanced Analog Line	Terminal Block
Hono™ 16.16M	CobraNet	16	16	Balanced Analog Mic/Line	Balanced Analog Line	Terminal Block
Hono™ 16.16L	CobraNet	16	16	Balanced Analog Line	Balanced Analog Line	Terminal Block
Hono™ 8.8D	CobraNet	8	8	AES/EBU	AES/EBU	Terminal Block
Hono™ 16.16D	CobraNet	16	16	AES/EBU	AES/EBU	Terminal Block

MINI NETWORK SOUND CARDS

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

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

The Hono™ CobraNet Mini products: the ideal solution for networked audio systems that need a small quantity of additional inputs or outputs.

FEATURES

Inputs

- Hono™ 4.4M and 2.2M: Four or two balanced analog mic/line inputs
- Hono™ 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™ 4.4M and 2.2M: Four or two balanced analog line outputs
- Hono™ 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

CobraNet

- Redundant Cobranet
- Eight or four channels of Cobranet in and out
- Four CobraNet transmitters and eight receivers

DSP

- Peak and RMS meters on all audio inputs and outputs
- Mixing of any input to any output
- Programmable delay on all audio outputs
- Silence detect on all audio outputs



- 5 band parametric equalizer and Compander/Limiter on all mic inputs
- Lua scripting support

Power

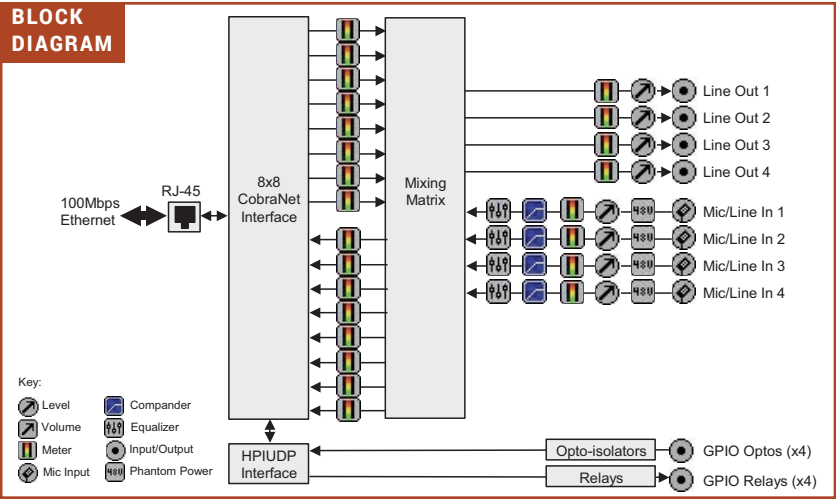
- Power over Ethernet (PoE) 802.3af compliant
- External +5V power supply if POE not being used

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

- All settings adjustable from ASiControl software



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

AudioScience has a selection of soundcards that support the Cirrus CobraNet and Axia Livewire audio-over-network protocols. The ASI6300 and ASI5300 CobraNet series utilize the PCI Express interface. The ASI6685 provides Livewire on PCI Express.



Product	CobraNet			Livewire
	ASI6316	ASI5316	ASI5308	ASI6685 LIVEWIRE
Interface	PCI Express	PCI Express	PCI Express	PCI Express/Livewire
Inputs	8 Stereo/16 Mono	8 Stereo/16 Mono	4 Stereo/8 Mono	8 Stereo/16 Mono
Outputs	8 Stereo/16 Mono	8 Stereo/16 Mono	4 Stereo/8 Mono	8 Stereo/16 Mono
Record Streams	8/16	8/16	4/8	8/16
Play Streams	8/16	8/16	4/8	8/16
Compression Formats	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
AES/EBU				
Max Cards/System	8	8	8	4
Size	2.5" x 6.6"	2.5" x 6.6"	2.5" x 6.6"	3.9" x 6.6"
Applications				
Radio Broadcast	•	•	•	•
Radio Production	•	•	•	•
Live Sound	•	•	•	•

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 (6.6 inches/168mm)
- Full height bracket available
- Up to 8 cards in 1 system



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	2.75" X 5.4"	2.75" X 5.4"	2.75" X 5.4"	2.75" X 5.4"	2.75" X 5.4"	2.75" X 5.4"	2.75" X 5.4"
Applications							
Radio Automation	•	•	•	•	•	•	•
Radio Production	•	•	•	•	•	•	•

AUDIOSCIENCE HAS AN EXTENSIVE SELECTION OF PCI EXPRESS SOUNDCARDS BUILT SPECIFICALLY FOR THE BROADCAST INDUSTRY. Our ASI6600 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 16 /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 (6.6 inches/168mm)
- Up to 4 cards in 1 system
- Windows and Linux drivers available



Product	ASI6614	ASI6618	ASI6620	ASI6622	ASI6640	ASI6644
Interface	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
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
Record Streams	1 or 2	1 or 2	2 or 4	2 or 4	4 or 8	4 or 8
Play Streams	4 or 12	4 or 12	4 or 6	4 or 6	4 or 12	4 or 12
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
MRX™	•	•	•	•	•	•
TSX™	•	•	•	•	•	•
Sample Rates	8-96kHz	8-96kHz	8-96kHz	8-96kHz	8-96kHz	8-96kHz
Analog	•	•	•	•	•	•
AES/EBU	•	•	•	•	•	•
SSX2 Multichannel	•	•	•	•	•	•
Max Cards/System	4	4	4	4	4	4
Size	3.9" x 6.6"	3.9" x 6.6"	3.9" x 6.6"	3.9" x 6.6"	3.9" x 6.6"	3.9" x 6.6"
Applications						
Radio Automation	•	•	•	•	•	•
Radio Production	•	•	•	•	•	•



When on-board compression is not needed, the ASi5700 series provides 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 (6.6 inches/168mm)
- Up to 8 cards in 1 system
- 32/64-bit Windows 10, 7, Server 2008/2012, and Linux software drivers available

Product	ASi5744	ASi5780	ASi5788
Interface	PCI Express	PCI Express	PCI Express
Inputs	4 Stereo/8 Mono	8 Stereo/16 Mono	8 Stereo/16 Mono
Outputs	4 Stereo/8 Mono	8 Stereo/16 Mono	8 Stereo/16 Mono
Record Streams	4 or 8	8 or 16	8 or 16
Play Streams	4 or 12	8 or 24	8 or 24
Compression Formats	PCM	PCM	PCM
Sample Rates	32-96kHz	32-96kHz	32-96kHz
Analog	•	•	•
AES/EBU	•	•	•
SSX Multichannel	•	•	•
Max Cards/System	8	8	8
Size	2.75" x 5.4"	2.75" x 5.4"	2.75" x 5.4"
Applications			
Radio Broadcast	•	•	•
Installed Sound	•	•	•



When on-board compression is not needed, the ASi5600 series provides multichannel playback and record using the PCI Express interface. Both balanced analog and AES/EBU I/O are offered.

FEATURES

- 12 stereo streams of playback into 4 stereo outputs
- 8 stereo streams of record from 4 stereo inputs
- Formats include 8, 16, 24 and 32 bit PCM with sample rates from 32kHz to 96kHz
- SSX® mode for multichannel record, playback, and mixing
- Short length PCI card format (6.6 inches/168mm)
- Up to 4 cards in one system
- 32/64-bit Windows 10, 7, Server 2008/2012, and Linux software drivers available

Product	ASi5620	ASi5622	ASi5640	ASi5641	ASi5644	ASi5680
Interface	PCI Express	PCI Express	PCI Express	PCI Express	PCI Express	PCI Express
Inputs	2 Stereo/8 Mono	2 Stereo/8 Mono	4 Stereo/8 Mono	4 Stereo/8 Mono	4 Stereo/8 Mono	1 Stereo/2 Mono
Outputs	2 Stereo/8 Mono	2 Stereo/8 Mono	4 Stereo/8 Mono	4 Stereo/8 Mono	4 Stereo/8 Mono	8 Stereo/16 Mono
Record Streams	4	4	4 or 8	4 or 8	4 or 8	1
Play Streams	6	6	4 or 12	4 or 12	4 or 12	8 or 16
Compression Formats	PCM	PCM	PCM	PCM	PCM	PCM
Sample Rates	32-96kHz	32-96kHz	32-96kHz	32-96kHz	32-96kHz	32-96kHz
Analog	•	•	•	•	•	•
AES/EBU	•	•	•	•	•	•
SSX Multichannel	•	•	•	•	•	•
Max Cards/System	4	4	4	4	4	4
Size	3.9" x 6.6"	3.9" x 6.6"	3.9" x 6.6"	3.9" x 6.6"	3.9" x 6.6"	3.9" x 6.6"
Applications						
Radio Broadcast	•	•	•	•	•	•
Installed Sound	•	•	•	•	•	•

PRODUCTION PROPROCESSING

SOUND CARDS SOUND CARD



THE ASI5810, ASI5811 AND ASI5211 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 ASI5211 and ASI5811 also include 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 and ASI5811)
- Two opto inputs and two relay outputs via a second bracket (ASI5211)
- 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 (ASI5211 & 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 10, 7, Server 2008/2012, and Linux software drivers available

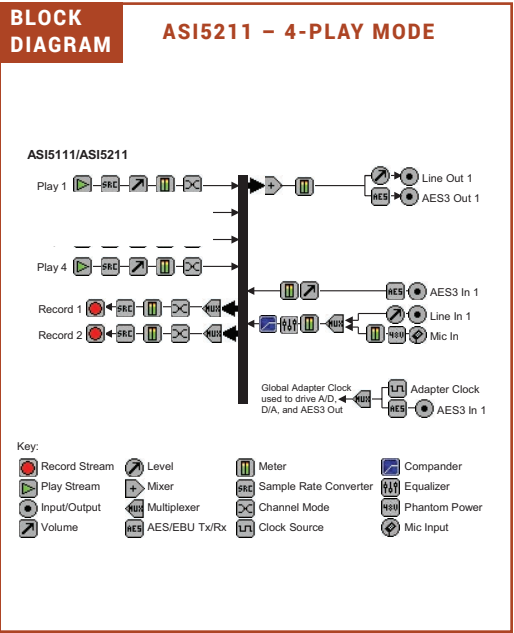


THE ASI5812 IS A PROFESSIONAL PCI EXPRESS AUDIO ADAPTER DESIGNED FOR USE IN RADIO BROADCAST FM MPX GENERATION. The adapter offer two stereo record streams from either a balanced analog input or AES/EBU digital input and four stereo play streams mixed to a balanced analog output and two AES/EBU digital outputs. The second AES/EBU output includes a Sample Rate Converter (SRC) and independent sample rate clock. The adapter also includes GPIO in the form of four opto-isolated inputs and two relay isolated outputs.

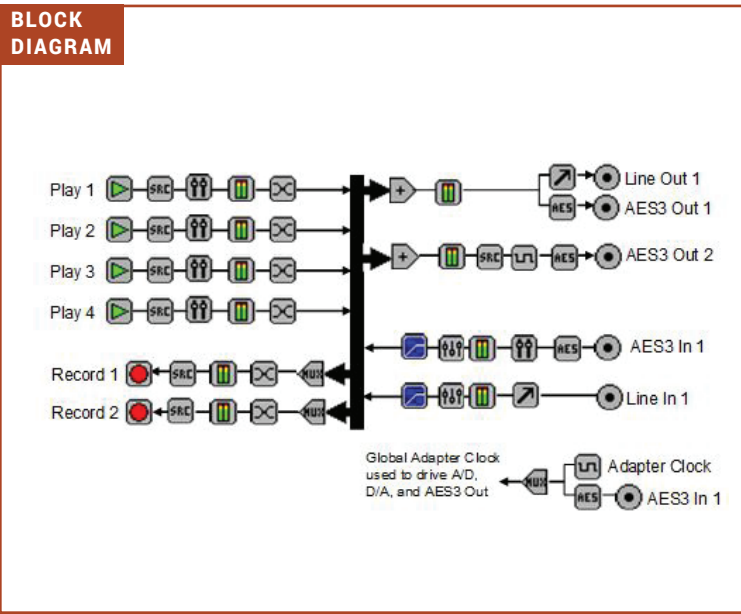
FEATURES

- Four stereo streams of PCM playback
- Two stereo streams of PCM record
- Balanced stereo analog input and output
- PCIe interface
- GPIO: Four opto inputs and two relay outputs
- AES/EBU or S/PDIF digital input and output (software selectable)
- 24bit analog-to-digital and digital-to-analog converters - 100dB SNR and 0.0025% THD+N
- 11 to 192kHz sample rates
- 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
- Half-height design and included mounting bracket allow for installation in small form factor systems
- Windows and Linux software drivers available

Product	ASI5810	ASI5811	ASI5211
Interface	PCI Express	PCI Express	PCI Express
Inputs	1 Stereo/2 Mono	1 Stereo/2 Mono + Microphone	1 Stereo/2 Mono + Microphone
Outputs	1 Stereo/2 Mono	1 Stereo/2 Mono	1 Stereo/2 Mono
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	11-96kHz
Analog	•	•	•
AES/EBU	•	•	•
Max Cards/System	8	8	4
Size	2.75" x 4.4"	2.75" x 4.4"	3.9" x 7.0"
Applications			
Radio Broadcast	•	•	•
Radio Production	•	•	•



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



TUNER CARDS OEM PRODUCTS

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 and/or NTSC/PAL/SECAM audio capture in both PCI and PCI Express interfaces.

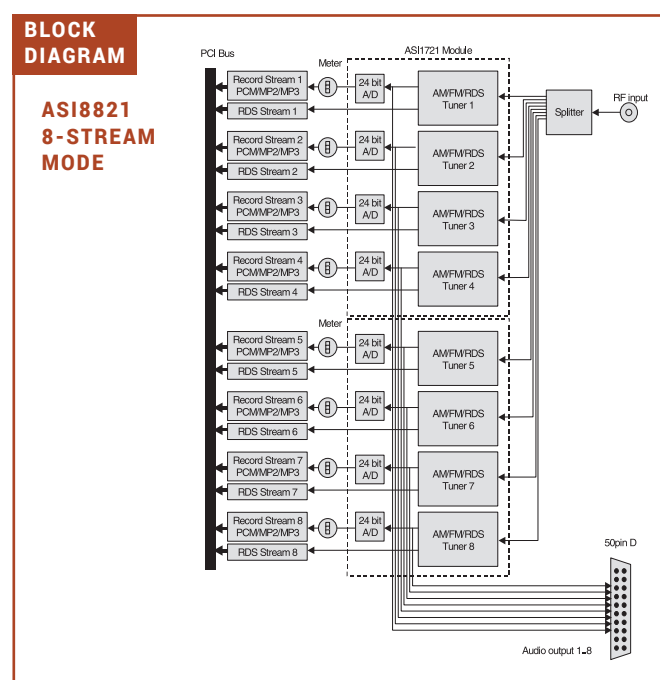
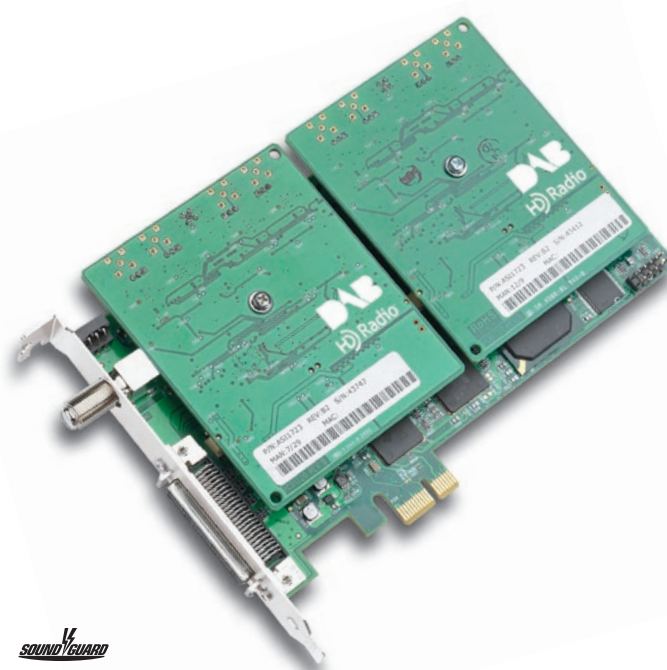
FEATURES

- 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
- Half-length PCI card
- Up to 8 cards in one system
- 32/64-bit Windows 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	•



AUDIOSCIENCE HAS THE CAPABILITY AND EXPERTISE TO DESIGN AND MANUFACTURE CUSTOM AUDIO PRODUCTS. Here are some examples of what we have delivered. For more information, please contact Richard Gross @ 1-302-324-5333

BLU-PCI

Custom BLU Link network audio card



ASI4401

Custom audio adapter for in-store advertising



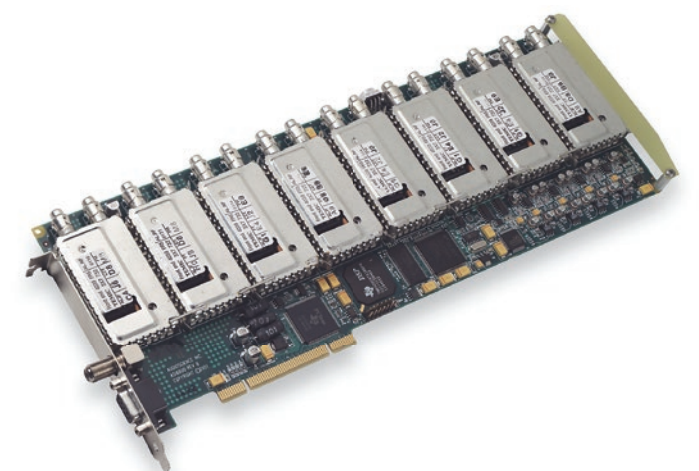
ASI4601

Custom MPEG audio adapter for Prophet Systems/Sirius satellite radio



ASI8801

Custom audio adapter for TV and FM radio advertisement monitoring/verification.



BREAK-OUT BOXES CABLES

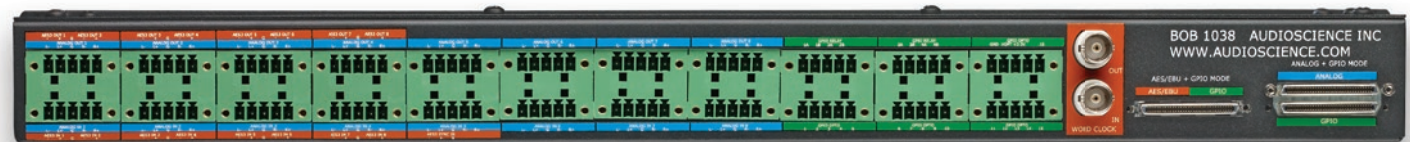
THE BOB1038 AND HUB-12 ARE 1U RACKMOUNT BREAK-OUT BOXES THAT TERMINATE THE ANALOG OR AES/EBU CONNECTORS ON SUPPORTED AUDIOSCIENCE ADAPTERS.

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.

Front view of BOB1038



Rear view of BOB1038



Front view of HUB-12



Rear view of HUB-12



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 Radio Systems at www.radiosystems.com.

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

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
CBL1001	ASI5111 ASI5211	DB-9 male to 1 stereo XLR in and out
CBL1003	ASI5111 ASI5211	DB-9 male to AES3 XLR in and out
CBL1004	ASI554x ASI564x ASI5680 ASI6514 ASI654x ASI6614 ASI664x	Mini DB50 to Centronics 50-pin adapter, analog
CBL1022	ASI5520 ASI652x ASI662x	Mini DB50 to XLR, balanced analog, 2 stereo in, 2 stereo out
CBL1044	ASI554x ASI564x ASI5680 ASI5744 ASI6514 ASI654x ASI6614 ASI664x ASI6744	50-pin to XLR, balanced analog, 4 stereo in, 4 stereo out. Needs to be paired with CBL1004.
CBL1101	ASI554x ASI564x ASI6514 ASI6544 ASI6614 ASI6644	Mini DB26 to Centronics 50-pin adapter, AES/EBU
CBL1122	ASI6522 ASI6622	Mini DB26 to XLR, AES/EBU, 2 stereo in, 2 stereo out, AES sync in, word clock I/O (BNC)
CBL1144	ASI554x ASI564x ASI6514 ASI6544 ASI6614 ASI6644	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 CBL1101.
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

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 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 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 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.

Copyright notice: ASIO is a trademark and software of Steinberg Media Technologies GmbH

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.



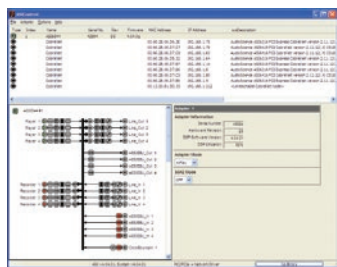
SOFTWARE APPLICATIONS

AudioScience supplies several applications that enable the configuration of our sound cards. For a current list, visit www.AudioScience.com.

ASiControl

This application is used to configure your audio adapter when running the WDM driver. It runs under 32/64-bit Windows 10, 7, and Server 2008/2012. It can be used to:

- Set routing and mixing of streams to physical inputs and outputs
- Set input and output analog levels
- Set tuner parameters (ASi8800/8900 series)
- Record and play audio streams to and from files
- Configure CobraNet bundles and channels
- Configure AVB endpoints and connect AVB streams



AUDIO SCIENCE IS THE AUDIO I/O COMPANY.





CORPORATE AND SALES

42 Reads Way
New Castle, Delaware 19720
Phone: +1-302-324-5333
FAX: +1-302-235-7110
salesasi@audioscience.com

TECHNICAL SUPPORT

Phone: +1-585-271-8870
support@audioscience.com

ENGINEERING AND MANUFACTURING

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

AUDIOSCIENCE ASIA

111 North Bridge Road
#21-01 Peninsula Plaza
Singapore 179098
Phone: +65 6507 4591
Fax : +65 6383 0401
salesasia@audioscience.com

AUDIOSCIENCE EUROPE

Calle Miguel Carrera, 33
Bloque 2, Escalera 2, 4D
29010 Malaga, Spain
Phone +44 203 287 3541
saleseu@audioscience.com

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 20 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.