ASI5111/ASI5211

LINEAR PCI/PCI EXPRESS AUDIO ADAPTERS

1 DESCRIPTION

The ASI5111/ASI5211 are professional PCI audio adapters designed for use in radio broadcast production.

The ASI5111/ASI5211 offer two stereo record stream from either a balanced analog input or AES/EBU digital input and four stereo play streams mixed to both a balanced analog output and an AES/EBU digital output.

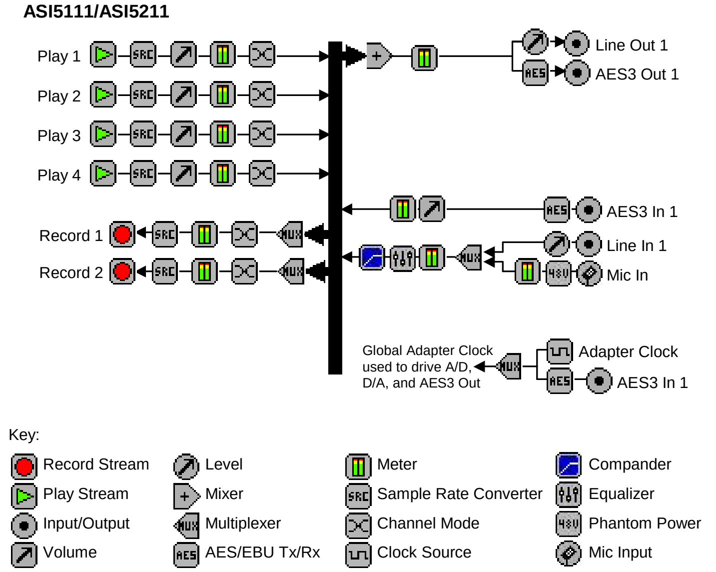
The ASI5111 is a PCI adapter and the ASI5211 is a PCI Express (PCIe) adapter. Additionally, the ASI5211 makes available two opto inputs and two relay outputs via a second bracket attached to the ASI5211 using a 10-pin ribbon cable.

Also included is a microphone input, with low noise pre-amp and a 48V phantom supply.

2 FEATURES

- Four stereo streams of PCM playback
- Two stereo streams of PCM record
- Balanced stereo analog input and output with +24dBu I/O (ASI5211)
- PCI interface (ASI5111) or PCIe interface (ASI5211)
- Two opto inputs and two relay outputs via a second bracket (ASI5211)
- AES/EBU or S/PDIF digital input and output (software selectable)
- Low noise microphone input with 48V phantom supply and DSP based compressor/limiter and 5-band equalizer
- 24bit analog-to-digital and digital-to-analog converters - 100dB SNR and 0.0025% THD+N
- 11 to 96kHz 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 4 cards in one system
- Windows 7, XP, Server 2008/2003, Linux, and OS X software drivers available

ASI5111/ASI5211



3 SPECIFICATIONS

BALANCED INPUT/OUTPUT	
Connector	DB-9 Female
Input Level	-10 to +20dBu (ASI5111) or +24dBu (ASI5211) in 1dBu steps
Input Impedance	20K ohms
Output Level	-10 to +20dBu (ASI5111) or +24dBu (ASI5211) in 1dBu steps
Load Impedance	600ohms or greater
S/N Ratio [1]	> 100dB (record or play)
THD+N [2]	< 0.0025% (record or play)
Sample Precision	24bit Oversampling
Frequency Response	20Hz to 20kHz +/-0.25dB 20Hz to 40kHz +0.25/-5dB[3]

MICROPHONE INPUT	
Connector	¼" TRS jack
Input Gain	20, 40 and 60dB software adjustable
Input Impedance	11K ohms (+ or – to ground)
Phantom Power	48V +/- 4V, software selectable on and off.
S/N Ratio [1]	90dB @ 40dB gain
THD+N [2]	0.005% @ 40dB gain
Frequency Response	20Hz to 20kHz +/-0.5dB 20Hz to 40kHz +0.5/-5dB [3]

DIGITAL INPUT/OUTPUT	
Type	AES/EBU (EIAJ CP-340 Type I / IEC-958 Professional) S/PDIF (EIAJ CP-340 Type II / IEC-958 Consumer) (software selectable)
Connector	DB-9 Male
Sample Rates	32, 44.1, 48, 64, 88.2 and 96kHz
Sample Precision	24bit

SAMPLE RATE CLOCK	
Internal	32, 44.1, 48, 64, 88.2 and 96kHz
AES/EBU In	32, 44.1, 48, 64, 88.2 and 96kHz

SIGNAL PROCESSING	
DSP	Texas Instruments TMS320C6711@135MHz
Memory	8MB
Audio Formats	8 bit unsigned PCM 16bit signed PCM 32bit IEEE floating point PCM

BREAKOUT CABLES (INCLUDED)	
Analog	CBL1001: DB-9 to 2 in and 2 out XLR
Digital	CBL1003: DB-9 to 1 in and 1 out XLR

GENERAL	
Bus	ASI5111: Universal 32bit PCI (3.3V or 5V signaling) ASI5211: X1 PCI Express.
Dimensions	PCI form factor – 6.75" x 3.9" x 0.6" (172mm x 100mm x 15mm)
Weight	8 oz (227g) max
Operating Temperature	0C to 70C
Power Requirements	ASI5111: +5V @ 600mA, +12V @ 150mA, -12V @ 70mA ASI5211: +3.3V @ TBD, +12V @ TBD

[1] - S/N Ratio is the difference between a 1kHz digital full-scale sinewave and digital zero using an A weighting filter
[2] - THD+N measured using a +20dBu 1kHz sinewave sampled at 48kHz and A weighting filter
[3] – Using a 96kHz sampling rate

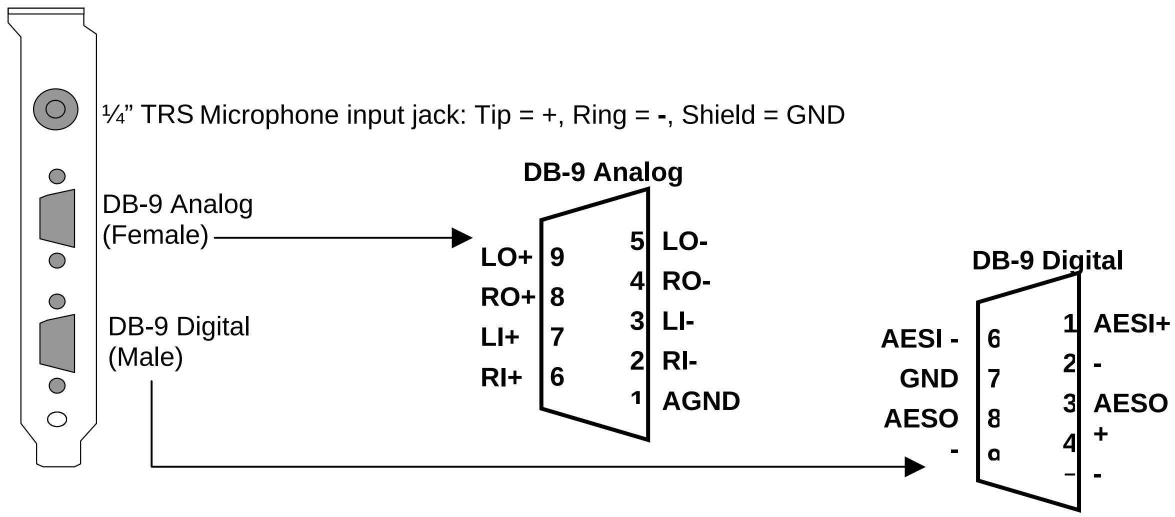
4 REVISIONS

Date	Description
10 June 2009	Updated format slightly. Added new block diagram.
22 July 2010	Updated datasheet format. Added ASI5211 (images to be added later to illustrate GPIO connector).
16 August 2010	Added image of ASI5211.
30 Nov 2010	Update specs for ASI5211
08 August 2011	Add Low Latency mode.
02 October 2011	Updated format.

5 CONTENTS

6 CONNECTORS

6.1 Mic, Analog, Digital

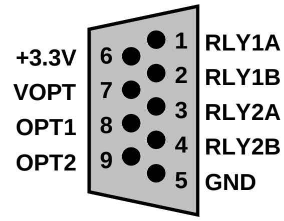


6.2 GPIO Pinouts - ASI5211 Only

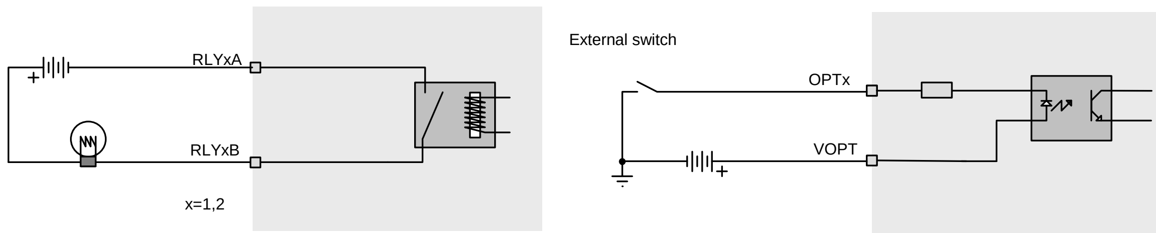
GPIO for the ASI5211 is made available on a second bracket. This second bracket is attached to the ASI5211 via a 10-pin ribbon cable. The red ribbon-edge is “1” and is connected to J9 on the ASI5211, with the bottom, left pin being “1.”

Images of the second bracket, attached to the ASI5211, will be added soon.

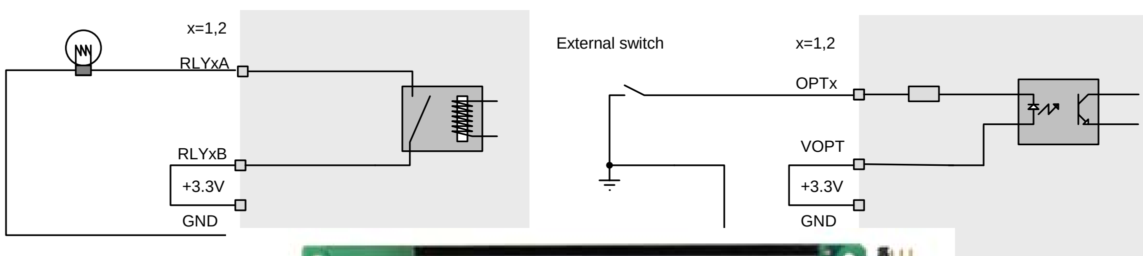
Male DB-9 GPIO



Isolated



TTL Compatible Non-isolated



7 CABLES

The analog cable is CBL1001; a DB-9 to 2 in and 2 out XLR.
The digital cable is CBL1003; a DB-9 to 1 in and 1 out XLR.
The cables are included with the ASI5111/ASI52111.

8 HARDWARE INSTALLATION

This section explains how to install one or more AudioScience adapters in a computer.

8.1 Setting Adapter Index – One Adapter in the PC

1. Make sure your computer is turned off.

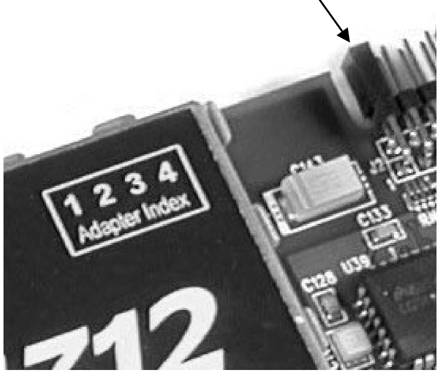
2. PCI adapters should be installed in any empty PCI slot and PCIe adapters should be installed in any x1 (or greater) PCIe slot.

3. Make sure the adapter jumper is set to adapter index #1, the factory default. For a new card no changes need to be made. For an AudioScience adapter from another installation, check that it is set to adapter index #1.

Depending on the adapter family, there are different ways of setting the adapter index.

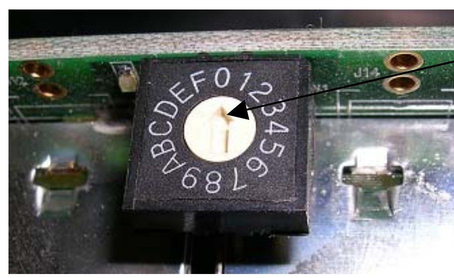
For ASI5000 and ASI6000 families, there is an adapter jumper that must be set. The left most position represents adapter index #1.

Adapter Jumper set to Adapter #1



For ASI5300, ASI6300, ASI8700, and ASI8900 families, there is a rotary switch.

NOTE: Position 0 (zero) represents adapter #1, position 1 is adapter #2, etc.



Adapter Index switch set to Adapter #1

4. Turn on the computer and let it boot. Under Windows, a dialog box will pop up informing you that the computer has detected a new Multimedia Audio card. Cancel out of this dialog box and proceed to the software installation section of this datasheet.

8.2 Setting Adapter Index - Two or More Adapters in the PC

1. Make sure your computer is turned off.

2. PCI adapters should be installed in any empty PCI slots and PCIe adapters should be installed in any x1 (or greater) PCIe slots. Different adapter types can coexist in the same computer; for example, an ASI6416 and ASI8921 will work correctly if installed in the same PC. Different adapter types still require unique adapter index numbers.

3. Each adapter in the PC needs to have its adapter jumper/rotary switch position set to unique numbers. For example if you are installing two adapters, the first one would be set to adapter index #1 and the second to adapter index #2.

3.1. For ASI5000 and ASI6000 families, the position to the right of index #1, when jumpered, represents adapter index #2. The next position represents #3, and the rightmost position, when jumpered, represents #4.

3.2. For ASI5300, ASI6300, ASI8700, and ASI8900 families, rotate the rotary switch to indicate what position is required.

4.
Turn on the computer and let it boot. Under Windows, a dialog box will pop up informing you that the computer has detected a new Multimedia Audio card. Cancel out of this dialog box and proceed to the software installation section of this datasheet.

9 SOFTWARE INSTALLATION

AudioScience makes audio adapters and drivers for various operating systems. Enhancements to an adapter's utility come from the integrators software that uses the audio driver to implement sophisticated audio playback and recording functions.

9.1 Drivers for Windows XP/Server 2003/Server 2008/7

The first step is what type of driver is needed for the adapter. There are two types of drivers for Windows: The WAVE driver and the WDM driver. Typically this will be decided by the application used with the AudioScience adapter. For any application that uses DirectSound, use the WDM driver.

Driver 3.10 and later present the user with three install options during installation:

- Install Standard PCI/PCIe Driver.
- Install Standard + Network Audio Driver.
- Remove all driver components

Traditional installs should select the first of these options. Users of AudioScience CobraNet products should select the second option with the "+Network Audio Driver." in the text.

9.1.1 WAVE Driver

Download the file named ASIWAVE_xxxxxx.EXE from www.audioscience.com and run it (_xxxxxx is the version number). After the EXE has run, reboot the computer and the audio adapter will be operational. If the cover is off the computer, one can see one or two blinking LEDs on top of the card indicating its DSP is running and communicating with the driver.

Verify that the adapter is running using ASIControl (see ASIControl section in this document).

9.1.2 WDM Driver

Download the file named ASIWDM_xxxxxx.EXE from www.audioscience.com and run it (_xxxxxx is the version number). After the EXE has run, reboot the computer and the audio adapter will be operational. If the cover is off the computer, one can see one or two blinking LEDs on top of the card indicating its DSP is running and communicating with the driver.

Verify that the adapter is running using ASIControl (see ASIControl section in this document).

9.1.3 Combo Driver

The Combo driver presents both Wave and WDM devices to the user. Download the file named ASICOMBOV_xxxxxx.EXE from www.audioscience.com and run it (_xxxxxx is the version number). After the EXE has run, reboot your computer and the audio adapter will be operational. If the cover is off the computer, one can see one or two blinking LEDs on top of the card indicating its DSP is running and communicating with the driver.

Verify that the adapter is running using ASIControl (see ASIControl section in this document).

9.1.4 ASIO

All AudioScience drivers also install an ASIO driver interface. It is installed by default.

9.1.5 Driver Failure

In the event that an adapter's driver fails to load correctly, the OS's event viewer should be checked. The event log is viewed as follows:

XP: The system event log is accessed from \Start\Control Panel\Administrative Tools\Event Viewer. The System view should be selected.

7: The system event log is accessed from \Start\Control Panel\System and Maintenance\Administrative Tools\Event Viewer. The Windows Logs\System view should be selected.

If two or more adapters are installed in the same system, the first thing to check is that the adapters were assigned unique adapter numbers. If issues persist, please email support@audioscience.com.

9.2 Drivers for Linux

The latest Linux driver can be downloaded from the AudioScience website – www.audioscience.com

9.3 Applications for Windows

AudioScience provides two application for adapter set-up and configuration: ASIControl and ASIMixer.

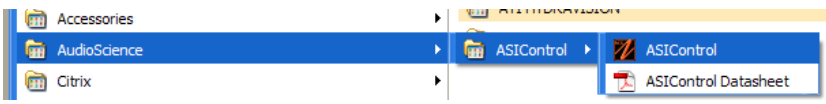
9.3.1 ASIControl

All Windows drivers install an AudioScience application called ASIControl that can be used to setup and verify functionality of adapters. ASIControl provides a common interface for users across all driver types.

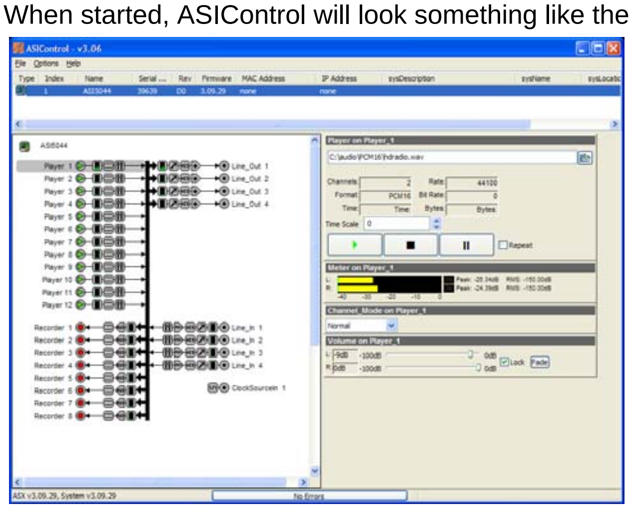
The following list of controls are uniquely supported in ASIControl (as opposed to ASIMixer):

- ASI8700 tuner pre-emphasis
- ASI8900 tuner RBDS
- ASI8900 tuner FM stereo indication
- ASI8914 HD Radio PSD field
- ASI8914 HD Radio Digital status field
- ASI8914 HD Radio Digital program number selection

From the Windows Start menu, navigate to Start→Programs→AudioScience and run the ASIControl program.



When started, ASIControl will look something like the following:



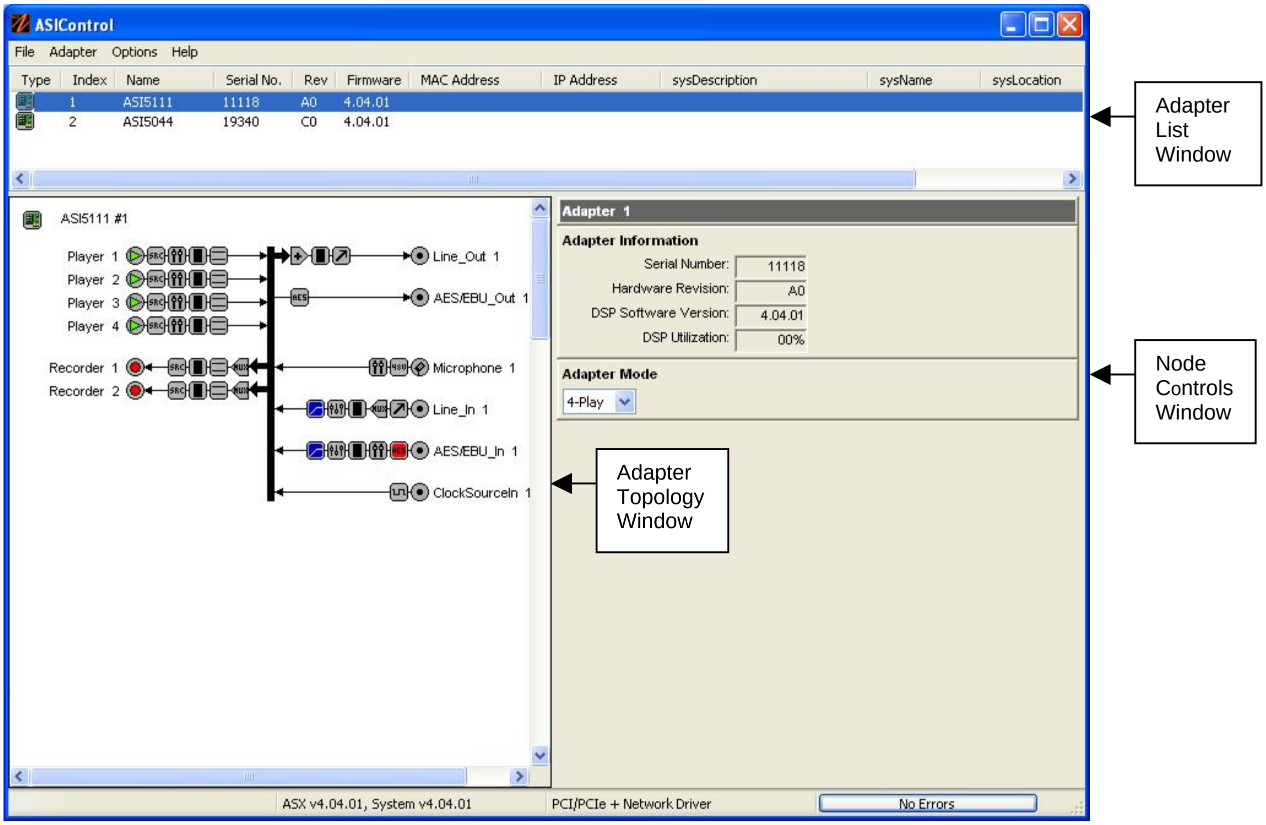
9.3.2 ASIMixer

ASIMixer is specific to the Wave and Combo drivers and is available from the AudioScience website. It uses the Wave/Mixer interface to control AudioScience adapters. Users of driver version 3.10 and later are encouraged to use ASIControl for manipulating adapter controls.

See the list of controls in the previous section that that are only available in ASIControl.

10 OPERATION USING ASICONTROL

Using ASIControl, the ASI5111 will look like so:



10.1 User Interface

ASIControl consists of three main windows: the adapter list in the top portion of the window, the adapter topology view on the left hand side, and the node control list on the right hand side.

10.1.1 Adapter List Window

The top portion of ASIControl shows a list of all the adapters that the application has found. By default, only bus based (i.e. PCI and/or PCI Express) adapters will be shown. If the network portion of the driver is installed (by selecting “Install Standard + Networked Audio Driver” after running the driver installer) and “Local PCI(e) + Networked adapters” is selected from ASIControl’s Options→Configure adapter interface, then AudioScience and other third party CobraNet devices will be shown.

Adapters are listed in order of adapter index. For bus-based adapters, this is determined by the adapter index jumper on the card. For AudioScience CobraNet devices this is calculated from the unit’s MAC address. Third party CobraNet devices are listed last as they have no AudioScience index.

10.1.2 Adapter Topology Window

The left hand side of ASIControl contains the topology view of the adapter. It is essentially a block diagram of the device showing the available physical inputs and outputs on the right hand side of the black, vertical ‘bus’ line. On the left hand side of the bus line, bus-based adapters show player and recorder streams, while CobraNet adapters show their network connections.

Each of the inputs and outputs is referred to as a node and each Node contains one or more controls. The topology shows each control as a small icon. A non-exhaustive list of nodes follows:

Line In	Recorder
Line Out	Tuner
AES/EBU In	Clock Source In
AES/EBU Out	CobraNet In
Player	CobraNet Out

Hovering the mouse over a particular node will highlight it. Clicking on a node will bring up the controls resident on that node in the right hand control list.

There is an adapter node in the top left corner of the topology window. Clicking on this will show adapter-specific controls and properties on the right hand side.

Not all adapters have all nodes.

10.1.3 Node Controls Window

The right hand side of ASIControl shows the controls associated with the selected node in the topology view. The controls are arranged, from top to bottom, in order of audio signal flow, i.e. the audio signal can be viewed as entering the node at the top control and leaving at the bottom control. Controls may be used to either manipulate the audio as it passes through the node, or report back control status information.

For a comprehensive listing of controls and how to operate ASIControl, please see the ASIControl manual available from www.audioscience.com and also installed by the driver. Not all adapters have all controls.

The section below lists some common and any specific controls, as seen in ASIControl, for this adapter.
For further information on controls common to all AudioScience adapters and how to operate ASIControl, please see the ASIControl manual, available from www.audioscience.com and also installed by the driver.

10.2 Controls

10.2.1 Adapter Information

This control displays information about the installed AudioScience product.

10.2.1.1 Interface

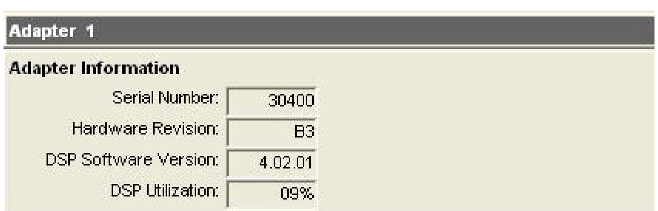


Figure 1. Adapter information seen in right side of ASIControl.

Serial Number:
The serial number is displayed here.

Hardware Revision:
This lists the hardware revision of the AudioScience product.

DSP Software Version:
The DSP software version is displayed; usually the same as the driver version installed.

DSP Utilization:
This shows the loading of the AudioScience product's DSP in percent.
Note: Utilization should be kept below 90%.

10.2.2 Adapter Mode

The Adapter_Mode control changes the number of players/recorders/lineouts that an adapter has. On certain adapters, not all sample rates/formats are supported; changing the mode of the adapter allows for best functionality with certain sample rates/formats. Different adapters will have different modes available, and not all adapters have modes. Please see datasheets on specific adapters, available at www.audioscience.com, to learn more.

10.2.2.1 Interface

Figure 2. Adapter Mode in ASIControl.

Selecting the appropriate mode from the list using the dropdown arrow changes the Adapter_Mode setting. A reboot is necessary after changing adapter mode. The mode setting is saved to the adapter's EEPROM.

The ASI5111/ASI5211 supports three adapter modes: 1-Play, 4-Play, and Low Latency.

10.2.2.2 1-Play

This mode supports 1 Play stream and 1 record stream.

10.2.2.3 4-Play

This modes supports 4 Play streams and 2 Record streams with full mixing capabilities.

10.2.2.4 Low Latency

NOTE: Driver 4.06.00 or later is required.
This mode supports a single multichannel play stream and a single multichannel record stream, enabling live sound processing in ASIO and Core Audio applications. See the Low Latency Mode datasheet for further information.

10.2.3 GPIO – ASI5211

The GPIO interface in ASIControl is located on the adapter node. Note only a few device types/configurations support GPIO.

10.2.3.1 Interface

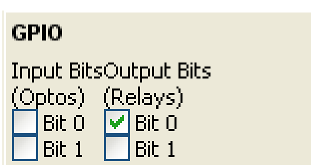


Figure 3. A view of 2 GPIO opto inputs and relay outputs.

10.2.3.2 Developer

Not all GPIO APIs are supported by every device type. See below table:

Device	WAVE – Windows	HPI - Windows	HPI - Linux	ASX - Windows	ASX - Linux	SNMP
ASI2416		•	•	•	•	
ASI2202				•		•
ASI5211	•	•	•	•		

10.2.3.2.1 Windows APIs

WAVE – GPIO inputs and outputs are accessed using MIXERCONTROL_CONTROLTYPE_BOOLEAN and the Windows mixerSetControlDetails() and mixerGetControlDetails() calls.
HPI – GPIO inputs and outputs are accessed using the HPI_GPIOxxx()API.
ASX – GPIO inputs and outputs are accessed using the ASX_GPIO_xxx() API.
SNMP – ASI2202 only

Variable	SNMP address
RLY1	OID : `1.3.6.1.4.1.2680.1.3.3.3.1.2.57` or stdUserInteger.57
RLY2	OID : `1.3.6.1.4.1.2680.1.3.3.3.1.2.58` or stdUserInteger.58

Note: stdUserInteger is defined by the CobraNet MIB (available on request from AudioScience).

10.2.3.2.2 Linux APIs

HPI – GPIO inputs and outputs are accessed using the HPI_GPIOxxx()API.
ASX – GPIO inputs and outputs are accessed using the ASX_GPIO_xxx() API.
SNMP – ASI2202 only (see above table).

10.2.4 Player

The Player control supports playback of an audio file from the computer's hard drive.

10.2.4.1 Interface

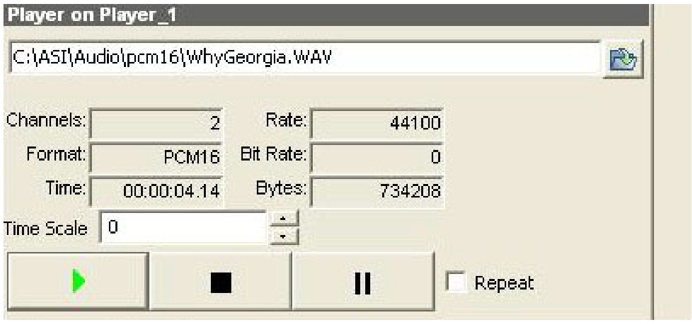


Figure 4. A player in ASIControl.

The first line of static text contains the selected playback file. Below the filename is the file information; playback time and playback bytes, the timescale select options, the player control buttons and the file repeat option.

10.2.4.2 How To Play a File

The first step in playing a file is to select the file to play. Use the **file icon button** to navigate to the desired file. After opening the file, the complete filename, including the path, will appear immediately to the left of the file open icon. At this point the file information is also filled in so that it contains the following fields: **Channels**, **Rate**, **Format**, and **Bit Rate**. Most of there are self-explanatory. The **Rate** refers to the sample rate of the audio recorded in the file. The **Bit Rate** applies only to MPEG compression and is set to 0 for all other formats.

At this point the percentage time scaling without pitch shift can be set if desired. The default of 0 indicates that time scaling is disabled. The valid range of settings is +/- 20 percent.

The **Repeat** check box indicates whether the file should be played again after playback has completed. It can be set either before playback has begun, or while playback is underway.

The file is now ready to be played. To start playback press the **play button**. At this point the **Time** and **Bytes** fields report playback time and the number of bytes of the file that have been played.

Once playback has started, the **stop** and **pause buttons** can be used to stop or pause the playback.

10.2.4.3 Using embedded sine wave generator

Manually typing in a filename of “~” and pressing play will cause a full-scale 1 kHz sine wave to be played at 48 kHz. The format of the filename string is: "~w, c,f,a,m,s,t".

w = **w**aveform = SINE (default=SINE)
c = **c**hannels = 1…8 (default = 2)
f = **f**requency = 1000 for 1kHz (default=1000)
a = **a**mplitude = **-**1 for **-**1dBFs (default=0dBFS, i.e. full scale)
m = channel **m**ask = 10 for left only, 01 for right only, 11 for stereo, etc. (default=1 for all channels)
t = sample **t**ype = (PCM8, PCM16, PCM24, PCM32, FLOAT32) (default=FLOAT32)
s = **s**ample rate = positive integer (default=48000) [validity depends on adapter]

Defaults can be used if the complete string is not specified, i.e.
"~" becomes "~wSINE,c2,f1000,a0,m11,s48000,tFLOAT32"

Any subset of the options may be specified, the remaining options will be set to the defaults. e.g. "~f500" = 500Hz stereo sine wave at 0dBFS, 48kHz sample rate.

10.2.4.4 Developer

10.2.4.4.1 Windows APIs

Wave – waveOutOpen(), waveOutWrite(), waveOutClose() etc.
HPI – Output stream functions documented here.
ASX – ASX Player control functions documented here.
DirectSound – TBD.

10.2.4.4.2 Linux APIs

HPI – TBD.

10.2.5 Level

The levels for the adapter’s analog line_outs and line_ins can be adjusted using the level control.
In the example below, the Line_Out 1 node in the topology view of ASIControl has been selected. Its Level will show up on the right side of ASIControl. The same is done for a Line_In to see its Level.

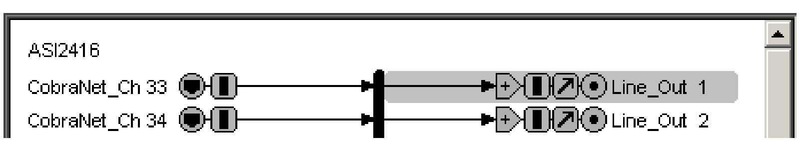


Figure 5. Using ASIControl to select Line_Out 1.

10.2.5.1 Interface

Figure 6. Level displayed by ASIControl for Line_Out 1.

Level:

The level is specified in dBu. It can be adjusted by clicking the arrows or by typing values in the edit box.

The level value represents the maximum output of a Line_Out and the maximum input (before clipping) of a Line_In. For example if you wish to use a +4dBu nominal level with 16dB of headroom, you would set the level to 4+16=+20dBuV.

Consult the specification section for the range of supported levels.

10.2.5.2 Developer

10.2.5.2.1 Windows APIs

Wave/Mixer – Analog levels are controlled using mixerSetControlDetails() on a control of type signed and with the name Level/Trim.
HPI – Analog levels are controlled using the HPI_LevelSet() API.
ASX – Analog level are controlled using the ASX_Level_Set() and ASX_Level_GetRange() API.
DirectSound – TBD.

10.2.5.2.2 Linux APIs

HPI – Analog levels are controlled using the HPI_LevelSet() API.
ASX – Analog level are controlled using the ASX_Level_Set()and ASX_Level_GetRange() API.
ALSA – TBD.

10.2.5.2.3 SNMP APIs

This section applies to specific ASI adapters.

10.2.5.2.3.1 ASI2202

Variable	SNMP address
MIC1 level	OID : `1.3.6.1.4.1.2680.1.3.3.3.1.2.51` or stdUserInteger.51
MIC2 level	OID : `1.3.6.1.4.1.2680.1.3.3.3.1.2.52` or stdUserInteger.52

Note: stdUserInteger is defined by the CobraNet MIB (available on request from AudioScience).

ASIControl level	OID value	Operation
0	0	Applies 0 dB gain to mic input
-6	1	Applies 6 dB gain to mic input
-12	2	Applies 12 dB gain to mic input
-18	3	Applies 18 dB gain to mic input
-24	4	Applies 24 dB gain to mic input
-30	5	Applies 30 dB gain to mic input
-36	6	Applies 36 dB gain to mic input
-42	7	Applies42 dB gain to mic input
-48	8	Applies 48 dB gain to mic input
-54	9	Applies 54 dB gain to mic input

10.2.6 Recorder

The Recorder control supports recording of an audio file.

10.2.6.1 Interface

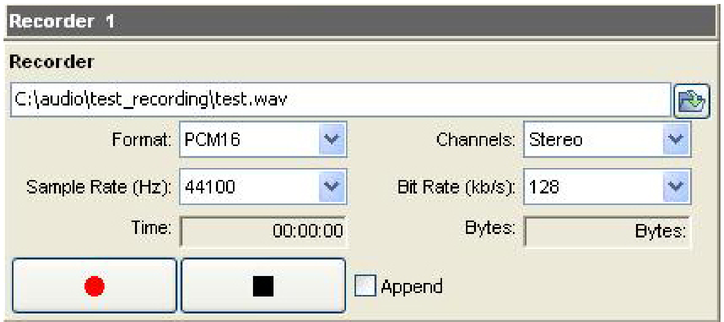


Figure 7. A recorder in ASIControl.

The first line of text contains the name given to the recorded file along with the location where it is to be saved. Below the filename is the file information, the record time and record bytes, the recorder control buttons and the file Append option.

10.2.6.2 How To Record a File

The first step in recording a file is to have audio coming into the adapter. This can be from a line-in or from one of the players in ASIControl. See appropriate sections in this datasheet to accomplish this. Next, the new file needs a name and place to be saved, or an existing audio file can be selected to be overwritten or appended to. Use the **file icon button** to navigate to the location to create the file and to give it a name, or to open a previously recorded file to overwrite or append to it. Next, from the dropdown arrows, select the number of "**Channels"**, the "**Sample Rate"**, the "**Format"**, and the "**Bitrate"** that the file should be recorded in.

Check the **Append** checkbox to save the audio to the end of an already existing file.

The file is now ready to be recorded. To start recording, press the **record button**. At this point the "**Time'** and "**Bytes'** fields report record time and the number of bytes of the file that have been recorded.

Once recording has started, the **stop** and **pause buttons** can be used to stop or pause the playback.

10.2.6.3 Developer

10.2.6.3.1 Windows APIs

Wave – use waveInOpen(), waveInStart() etc.
HPI – use HPI_InStreamxxx() functions.
ASX – use ASX_Recorder_xxx() functions.
DirectSound – TBD.

10.2.6.3.2 Linux APIs

HPI – use HPI_InStreamxxx() functions.
ASX – use ASX_Recorder_xxx() functions.
ALSA – TBD

10.2.7 Volume

The Volume control allows the audio signal's gain to be altered in the range of –100 to +20dB.

10.2.7.1 Interface

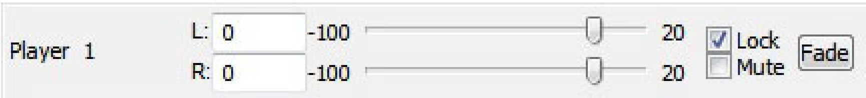


Figure 8. A Volume of a Player in ASIControl.

Left and **Right** display boxes:
Displays the gain settings that the slider bars are set to.

Slider Bars:
Click on the bar with the mouse and drag to desired gain. Once the bars are selected, the left and right arrow keys can also be used to change the settings.

Lock:
When checked, locks the left and right channels to the same gain value. When unchecked, allows the left and right channels to have independent gains. (Note that if an adapter is in SSX2 mode, the Player volumes cannot be unlocked to move the left and right channels independently.)

Mute:
Check this box to mute the volume.

Fade:
When pressed, automatically fades the volume to the opposite end of the scale.

10.2.7.2 Developer

10.2.7.2.1 Windows APIs

Wave/Mixer – MIXERCONTROL_CONTROLTYPE_VOLUME
This is a Windows standard volume control. Settings are in the range of 0 to 65535, where 0 completely mutes the output and 65535 is the maximum volume.
HPI – HPI_Volume APIs.
ASX – ASX_Volume APIs.
DirectSound – TBD.

10.2.7.2.2 Linux APIs

HPI –HPI_Volume APIs.
ASX –ASX_Volume APIs.
ALSA – TBD.

10.2.8 Meter

Meters in ASIControl are located on audio nodes and display the audio level as the audio signal passes through the node. Most AudioScience devices return both RMS and peak level readings and ASIControl displays both simultaneously.

10.2.8.1 Interface

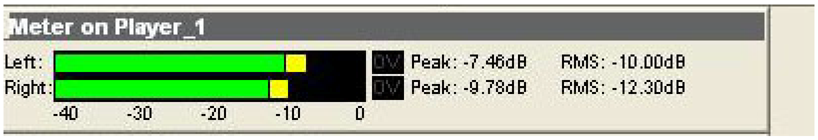


Figure 9. A stereo peak meter display. The RMS is the green bar and the peak is the yellow bar.

To the right of the peak meter is the absolute readings in dBFS. These can be useful when testing input tones of a specific known level.

10.2.8.2 Developer

10.2.8.2.1 Windows APIs

Wave/Mixer – Meters are read using mixerGetControlDetails() on a control of type signed and with type "Peak" the name "Peak Meter". A minimum value is 0 and maximum is 32767. The interface returns the peak readings only, not the RSM level. It confirms to expected Windows functionality.
HPI – Meters are read using the HPI_Meterxxx() API.
ASX – Meters are read using the ASX_Meter_xxx() API.
DirectSound – TBD.

10.2.8.2.2 Linux APIs

HPI – Meters are read using the HPI_Meterxxx() API.
ASX – Meters are read using the ASX_Meter_xxx() API.
ALSA – TBD.

10.2.9 Channel_Mode

The channel mode is a mechanism for handling mono to stereo conversions and directing the output to either left or right channels, as well as outputting left to stereo and right to stereo.

10.2.9.1 Interface

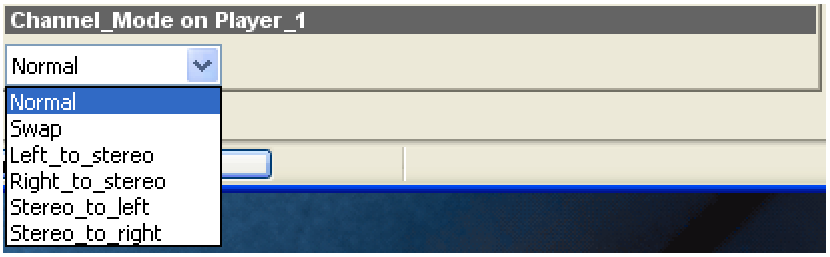
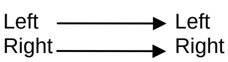


Figure 10. ASIControl view of a player's channel mode control.

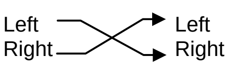
Default playback of either mono or stereo files causes audio to be output from the player on both the left and right audio channels. The channel mode control can allow the audio to be directed to either the left only or the right only. Select a channel mode setting from the dropdown list.

Valid settings are:

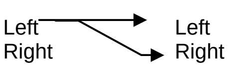
Normal – left channel out left channel, right channel out right channel



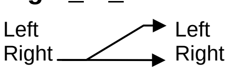
Swap – left channel out right channel and right channel out left channel



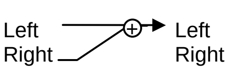
Left_to_stereo – left channel out to both left and right channels



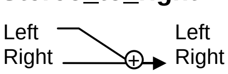
Right_to_stereo – right channel out to both left and right channels



Stereo_to_left – left and right channels out to left channel



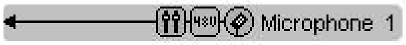
Stereo_to_right – left and right channels out to right channel



The Stereo_to_left and Stereo_to_right operations perform a sum of the left and right channels and then divides the result by 2.

10.2.10 Microphone Input

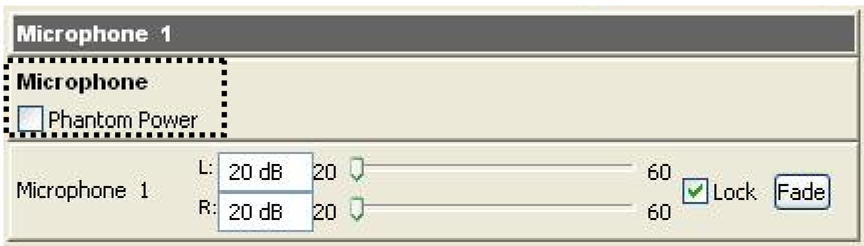
The ASI5111/ASI5211 has a balanced microphone input using a ¼” stereo jack. Click on the Microphone 1 node in the topology pane of ASIControl to access Phantom Power and the Microphone 1 gain.



10.2.10.1 Phantom Power

When phantom power is enabled, +48V is present on both the + and – signal inputs (tip and ring of ¼” jack). This is used to drive professional condenser type microphones. If you are using a dynamic microphone, make sure that the phantom power is off as it may damage the mic.

Phantom power is turned on and off by checking or unchecking the Phantom Power checkbox in ASIControl:



10.2.10.2 Developer

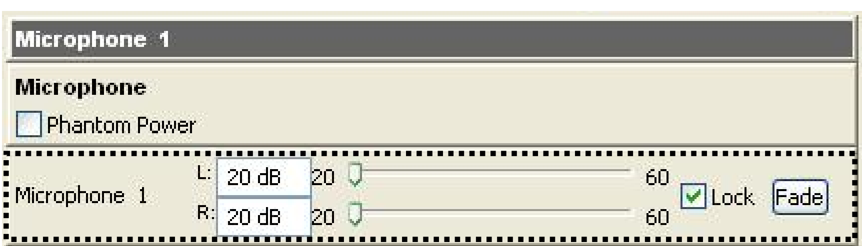
10.2.10.2.1 Windows APIs

HPI – Phantom power is controlled using the HPI_Microphone_SetPhantomPower() API

10.2.10.3 Programmable Gain

The microphone preamp has a software programmable gain of +20, +40 or +60dB.
Note: The gain interface is the same one used on stereo line ins and line outs. For the Microphone 1 on the ASI5111/ASI5211, the Fade button and the Lock checkbox will not work properly and the right channel is ignored. Though it is in a slider format, only values of +20, +40, or +60 are accepted.

Microphone gain is adjusted using the following control in the ASIControl:



10.2.10.4 Developer

10.2.10.4.1 Windows APIs

HPI – Microphone is controlled using a Volume control on the MICROPHONE source node. Use HPI_VolumeSetGain() API.

10.2.11 ClockSourceIn

In the topology pane of ASIControl, click on Clock Source 1

and in the node pane, select where the adapter is to get its clock source from using the Clock Source dropdown list, as well as the sample rate to use if clocking from adapter.

Note that for CobraNet and Livewire devices, the sample rate is fixed at 48kHz.

10.2.11.1 Interface

Figure 11. Clock Source information as seen in ASIControl.

Local Rate:
Select from the dropdown list the supported rates of the adapter.

Clock Source:
From the dropdown list, select the source for the adapter's clocking. Selections, depending on the adapter, include:

- Local – adapter rate is used; select a supported sample rate in Local Rate dropdown list
- Word – Word clock from Word clock BNC connector on digital cable loom (or BOB1024)
- WordHeader – Word clock from header on adapter (ASI61xx only)
- AES/EBU Sync – AES/EBU Sync from AES/EBU Sync XLR connector on digital cable loom (or BOB1024)
- AES/EBU In 1-4 – rate taken from specific digital input
- AES/EBU Auto – rate taken from first valid digital input; looks at digital input 1 first, then up to digital input 4

Adapter Rate:
Displays current adapter rate.

10.2.12 COMPANDER

The ASI5111/ASI5211 contains a compressor/expander (Compander), which is used to reduce or expand the dynamic range of the signal it acts on. It is located on the LineIn input and maybe used on both the Line In and Microphone signals.

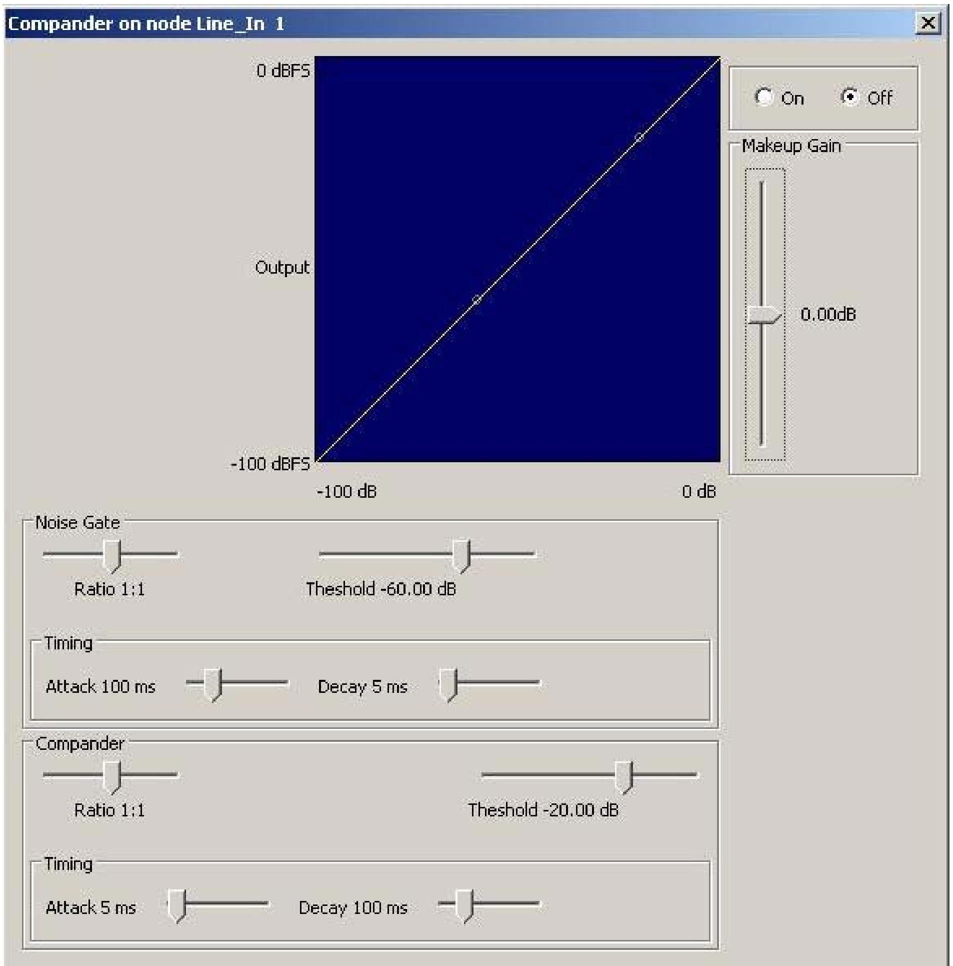
The Compander is accessed from the ASI Mixer by clicking on the "Compander" button on the LineIn panel. The following parameters can be set:

Compression Threshold – the input signal level at which the compression starts
Compression Ratio – The ratio of the input signal level to the output signal level
Makeup Gain – additional gain applied the compressed/expanded signal
Attack – Attack time of compander in milliseconds. Sets the time that the compressor takes to act
Decay – Decay time of compander in milliseconds. Sets the time for the signal gain to return to normal after compression
Noise Gate –

10.2.12.1 Developer

10.2.12.1.1 Windows APIs

Wave – Use the Compander control – see the "AudioScience WavX Specification" (SPCWAVX.PDF)
HPI – Use the HPI_Compandor_XXXX APIs - see the "AudioScience HPI Specification" (SPCHPI.PDF)

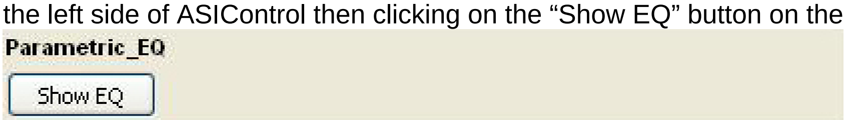


10.2.13 Parametric Equalizer

The AudioScience parametric equalizer is a 5 band parametric equalizer. It is located on the Line_In and AES/EBU_In nodes and may be used on both the Line In, AES/EBU In, and Microphone signals. Each of the equalizers 5 bands may be individually programmed with filter type (Bypass, Lowshelf, Highshelf, Equalizer, Lowpass, Highpass, Bandpass, and Bandstop), Q (sharpness), and center frequency.

10.2.14 Interface

The Parametric Equalizer is accessed from the ASIControl by clicking on either a Line_In or an AES/EBU_In in the left side of ASIControl then clicking on the “Show EQ” button on the right side of ASIControl.



The Parametric EQ opens, shown below.

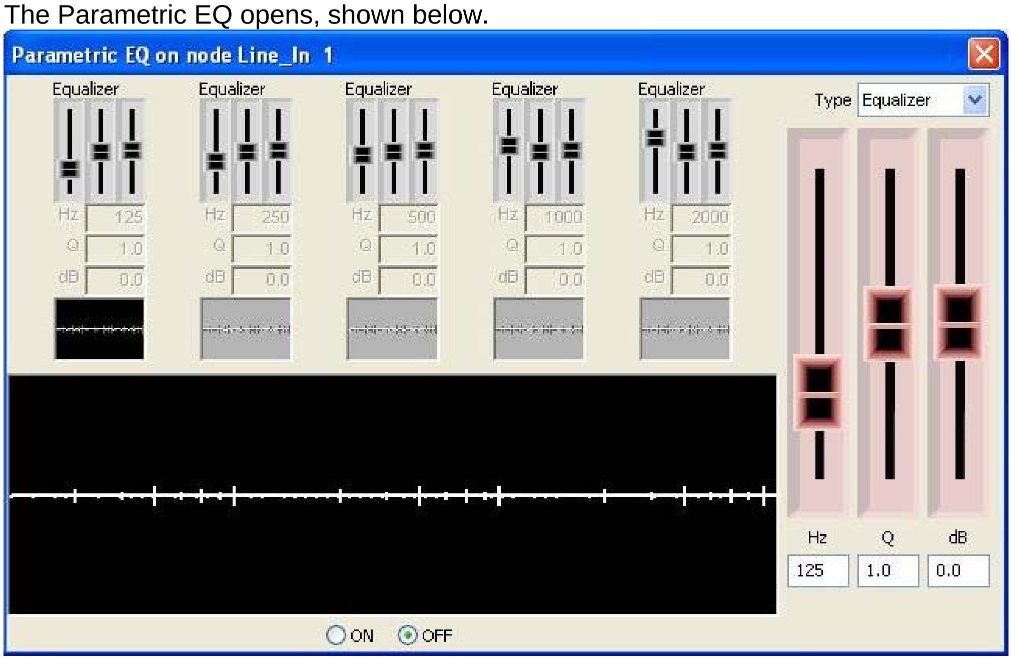


Figure 12 The Parametric EQ as seen in ASIControl

The EQ window contains controls for setting the filter parameters of each of the 5 bands, with a graph showing the combined frequency response of the 5 bands.

Clicking on one of the bands highlights it by changinging its small graph display black, as shown on the left band in the image above. Select the type of graph you want from the Type selection box in the upper right corner, and adjust levels by sliding the large sloders on the right. Click on the next equalizer and change its parameters as needed.

At the bottom of the ASI Parametric EQ pop up, click on the On radio button to activate it.

Each filter band has the following parameters:
Filter Type – The shape of the filter. Supported filter types are:

- Bypass – filter is turned off
- Low Shelf – EQ low shelf
- High Shelf – EQ high shelf
- Equalizer – EQ band (default)
- Low Pass – Standard low pass
- High Pass – Standard high pass
- Band Pass – Standard band pass
- Band Stop – Standard band stop/notch

Filter Hz (Freq) – The center frequency of the filter.
Filter Q – The sharpness of the filter. The higher the Q, the more selective the filter is.
Filter dB (Gain) – The gain of the filter at the center frequency.

10.2.15 Developer

10.2.15.1 Windows APIs

Wave – Use the equalizer mixer control – see "AudioScience WavX Specification"
HPI – Use the HPI_ParametricEQ_XXXX APIs – see "AudioScience HPI Specification"
ASX – TBD
DirectSound – TBD

10.2.15.2 Linux APIs

HPI – TBD
ASX – TBD
ALSA – TBD

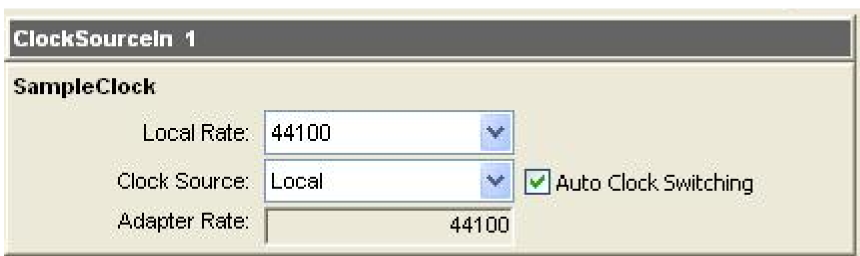
10.2.16 Sample Rate Clock and MRX Mixer

The ASI5111/ASI5211 sample rate clock is used to drive the MRX digital mixer, Analog to Digital Converter (ADC), Digital to Analog Converter (DAC) and AES/EBU output. There are two sources of sample rate clock – internal and the AES/EBU input.

The internal adapter clock is generated from a low jitter frequency synthesizer and may be set to 32, 44.1, 48, 64, 88.2 and 96kHz. When a valid AES/EBU bitstream is connected to the AES/EBU input, the ASI5111/ASI5211 will **automatically** switch to using this as the sample rate clock. This is needed so that digital audio from the AES/EBU input can be synchronized with the other audio streams present in the mixer. There is no way to override this.

Note that the sample rate clock does not determine the sample rates of the audio streams that may be played and recorded. These are independently set using the MRX multi rate mixer, so that, for instance, you can have the adapter running at 96kHz, but be playing files of 44.1 and 48kHz and recording files of 32 and 88.2khz.

In ASIControl, click on the ClockSourceIn 1 node in the topology pane to access the Local Rate, Clock Source, and Adapter Rate. Local Rate is used to select the internal adapter rate. Note the SampleClk source control is not user selectable as the adapter automatically switches depending whether a valid AES/EBU input is present.



10.2.17 Developer

10.2.17.1 Windows APIs

HPI – Use the HPI_SampleClock_XXXX APIs.

11 AUDIO FORMATS

The ASI5111/ASI5211 supports record and play of the following formats:

Format	HPI format	Windows format
8 bit unsigned PCM	HPI_FORMAT_PCM8_UNSIGNED	WAVE_FORMAT_PCM, wBitsPerSample=8
16 bit signed PCM	HPI_FORMAT_PCM16_SIGNED	WAVE_FORMAT_PCM, wBitsPerSample=16
32 bit signed PCM	HPI_FORMAT_PCM32_SIGNED	WAVE_FORMAT_PCM, wBitsPerSample=32
32 bit floating point PCM (+/-1.0)	HPI_FORMAT_PCM32_FLOAT	WAVE_FORMAT_IEEE_FLOAT

<end>

12 REFERENCES

12.1 Specifications

SPCWAVX.PDF - WavX - AudioScience Windows Multimedia Extensions

SPCHPI.PDF - Hardware Programming Interface (HPI) Specification

All these documents are available from www.audioscience.com in the Technical Info section