ASI6585/ASI6685

AXIA LIVEWIRE PCI/PCI EXPRESS SOUND CARDS

1 DESCRIPTION

The ASI6585/ASI6685 are professional PCI sound cards designed for use in an Axia Livewire™ IP-audio based radio broadcast facility. Livewire™ is an innovative technology used to transmit low-delay audio over switched Ethernet. For further information on Axia's Livewire, visit their website at: <http://www.axiaaudio.com/livewire/default.htm>

Providing up to 16 play streams that are mixed to 8 stereo outputs and 16 record streams fed from 8 stereo inputs, the ASI6585/ASI6685 features AudioScience's unique "anything to anywhere" mixing and routing.

A choice of uncompressed PCM, MPEG layer 2 and MP3 is available for both recording and playback. All compression is handled by an on-board floating point DSP, allowing the host computer to focus on other tasks.

DSP based functionality includes MRX™ multi-rate mixing technology that allows streams of different sample-rates and formats to be mixed.

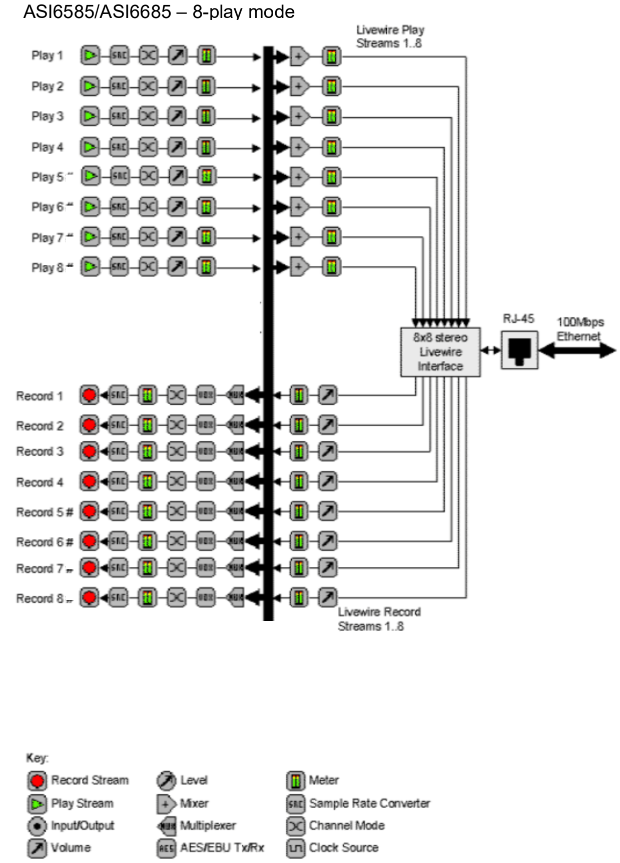
TSX™ time scaling provides compression/expansion of any or all playback streams in real time with no change in pitch.

For emerging surround sound applications, SSX™ mode allows multichannel streams of up to 8 channels to be played, recorded and mixed.

2 FEATURES

- 4, 8, 12 or 16 streams of mono or stereo playback into 8 stereo Livewire streams
- 4, 8 or 16 streams of mono or stereo record from 8 stereo Livewire streams
- Formats include PCM, MPEG layer 2 and MP3 with sample rates from 8kHz 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
- Two SSX™ streams for multichannel record and playback.
- Short length PCI card format (6.6 inches/168mm)
- Up to 4 cards in one system
- Windows 10, 7, Server 2008/2012 and Linux drivers available

ASI6585/ASI6685 – 8-play mode



- The ASI6585 has been discontinued; it is included in this datasheet for informational purposes only. The ASI6585 and ASI6685 are functionally the same except for the PCI bus connection. All information here will apply to either model number unless otherwise noted.

3 SPECIFICATIONS

LIVEWIRE INPUT/OUTPUT	
Type	100BaseT Ethernet
Connector	RJ-45
Precision	24bit PCM
Sample Rate	48kHz (Livewire sample rate)
Network Interface Latency	750 microseconds
Control Protocol	R/UDP, HTTP/HTML
SIGNAL PROCESSING	
DSP	Texas Instruments TMS320C6713@300MHz
Memory	8MB
Audio Formats	8 bit unsigned PCM 16 bit signed PCM 24 bit signed PCM 32 bit floating point PCM MPEG-1 Layer 2 MPEG-1 Layer 3 (MP3) (MPEG Layer-3 audio coding technology licensed from Fraunhofer IIS and THOMSON multimedia)
GENERAL	
Bus	32bit Universal PCI (PCI-X compatible)
Dimensions	PCI short-length form factor (6.6 inches/168mm long)
Weight	8 oz (227g) max
Operating Temperature	0C to 70C
Power Requirements	+3V@1.5A, +5V @ 100mA
Connectors	The mini DB26 connector has no function; it is only there to help hold the card to the mounting bracket.

4 REVISIONS

Date	Description
04 April 2009	Added DSP Mips table.
05 May 2009	- Section 10; renamed “Inputs” and “Outputs’ to “Stereo Livewire Inputs” and “Stereo Livewire Outputs.” - Section 14: added Note about using only AudioScience recommended firmware. - Section 14.2: added Note about upgrading from firmware versions 2.5.2d_r2 or 2.3.3b → must follow information in Section 14.3.1 first.
01 June 2009	Section 12 Note: for GPIO handling, added that an extra NIC can be used.
17 November 2009	- Section 14.31: Added new firmware info; 2.5.8a_r2. - Added information to Section 14.3.3 for added clarity.
22 December 2009	- Page 1: Updated list of software drivers available. - Section 10: Added Mono mode.
05 April 2011	Added ASI6685 (PCI Express version)
09 August 2011	Added Low Latency mode. Updated format.
24 April 2013	Added note regarding purpose of DB26 connector in Specifications, General table
22 May 2013	Fixed bootps download link Added further explanation as to how to perform IP address recovery.
16 Oct 2014	Updated DSP utilization chart
19 November 2014	Updated operating system and install instructions
19 Sept 2015	Updated bootps download link
25 January 2018	Update firmware instructions
13 April 2018	Additional updates to firmware instructions
13 Oct 2020	Added programing info to GPIO section
Dec 9 2020	Added notes regarding ASI6585 discontinuation
Feb 8 2024	Fixed error in firmware update section

5 CONTENTS

6 DSP UTILIZATION

The ASI6xxx series of adapters have world-class audio signal processing capabilities. The ASI6xxx algorithm complexity has increased at a faster rate than DSP processing power, resulting in a situation where not all available algorithms on an ASI6xxx can run simultaneously.

The following tabulates processing “budgets” so that problem configurations can be identified before system design is completed. The following tables assign a utilization percentage for various operations. By summing up the utilizations for the target (worst case) configuration, one can determine whether audio processing will run without causing dropouts or breakup.

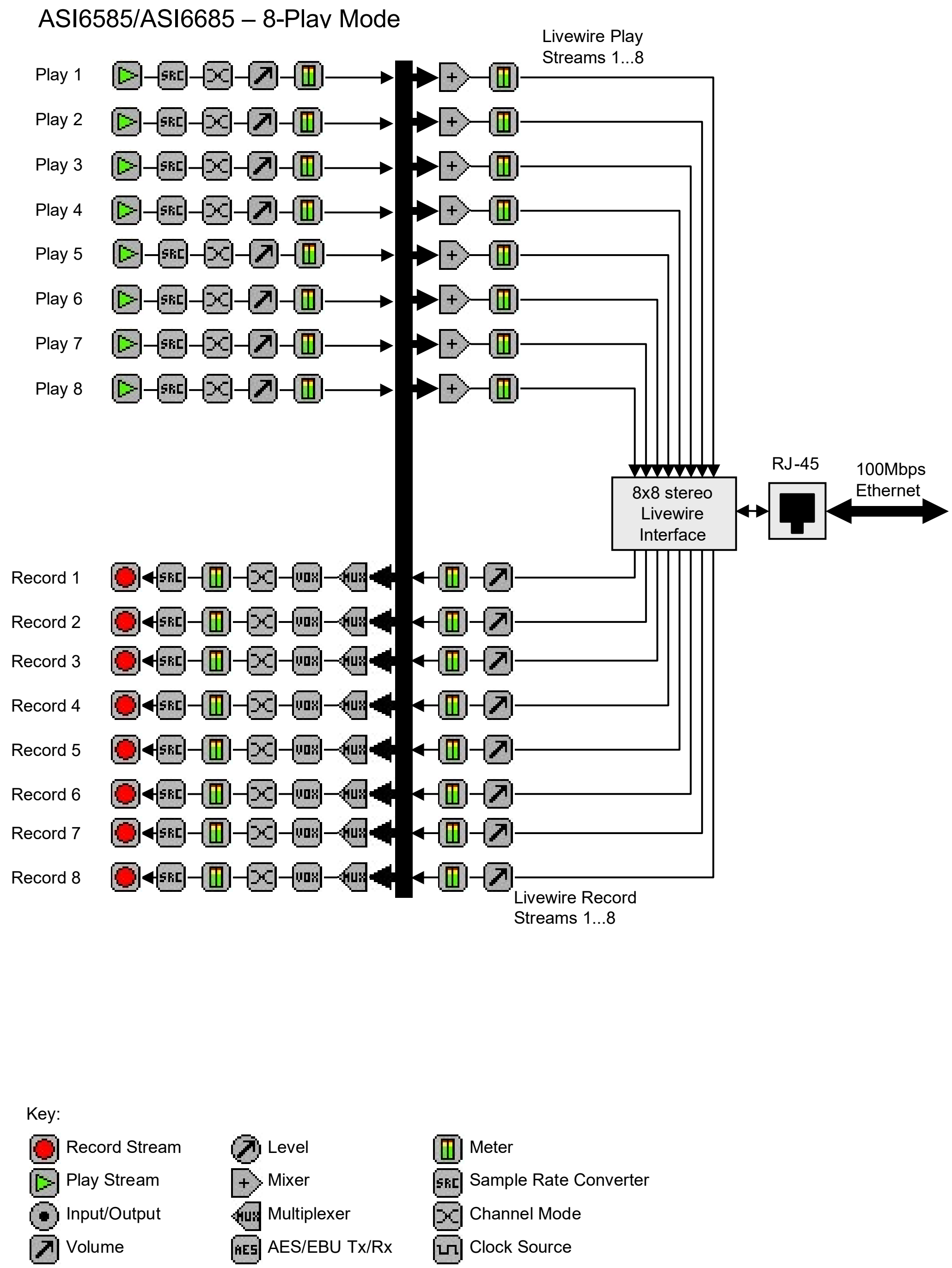
6.1 ASI6585 Rev. D6, samplerate 48kHz, driver 4.10.29

Idle utilization in 8-Play mode = 8%, in 16-play mode = 18%

Operation	**Play (utilization/ device)**		**Record (utilization/ device)**	
	8-Play	**16- Play**	**8- Play**	**16- Play**
PCM 16 @ 48kHz	2	3	3	3
MPEG-1 Layer-2, 256 kbps @ 48kHz	7	8	11	12
MPEG-2 Layer-3, 256 kbps @ 48kHz	12	13	33	32
SampleRate Conversion to/from 32kHz PCM 16	7	8	7	7
SampleRate Conversion to/from 44.1kHz PCM 16	7	8	6	6
SampleRate Conversion to/from 32kHz MPEG-1 Layer-2	10	11	13	13
SampleRate Conversion to/from 44.1kHz MPEG-1 Layer-2	10	12	15	15
SampleRate Conversion to/from 32kHz MPEG-2 Layer-3	14	16	28	28
SampleRate Conversion to/from 44.1kHz MPEG-2 Layer-3	16	17	36	34
TimeScale (90%) PCM	5	5	NA	
TimeScale (110%) PCM	5	5	NA	
TimeScale (90%) MPEG-1 Layer 2	5	5	NA	
TimeScale (110%) MPEG-1 Layer 2	5	5	NA	
TimeScale (90%) MPEG-2 Layer 3	5	5	NA	
TimeScale (110%) MPEG-2 Layer 3	5	5	NA	

From the above table, an ASI6585 in 8-Play mode:
4xPCM16 playback (all at 48kHz) = idle (8%) + 4x2% (8%) = ~16%.
4xMP3 playback (all at 44.1khz) = idle (8%) + 4x16% (64%) = ~72%

7 BLOCK DIAGRAM



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.

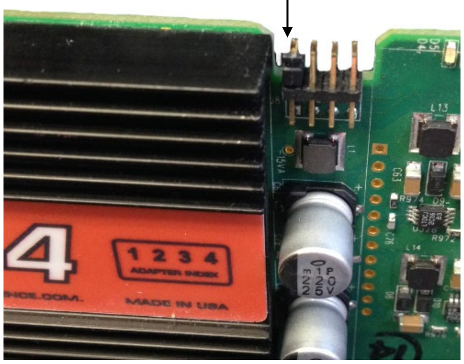
2. PCI adapters should be installed in any empty PCI slot and PCIe adapters should be installed in any x1 (or greater) PCIe slot.

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.

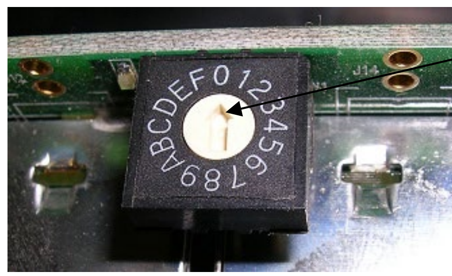
For ASI5000 and ASI6000 families, there is an adapter jumper that must be set. The left most position represents adapter index #1.

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.

8.1.1 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.

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 10, 7, Server 2008, Server 2012

Typically, drivers are not included with the hardware and will need to be downloaded from the AudioScience website. They can be found here: http://www.audioscience.com/internet/download/win_drivers.htm

The first step is to determine what type of driver is needed for your operating system. Drivers are available for 32-bit and 64-bit Windows systems.

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 and AVB products should select the second option with the "+Network Audio Driver." in the text.

9.1.1 Combo Driver

The Combo driver installs WDM devices by default and presents an option to "Install legacy 32-bit WAVE driver" in case your application requires it. Download the file named ASICOMBO_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).

9.1.2 ASIO

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

9.1.3 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 accessed from the Administrative Tools applet in Windows Control Panel under 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

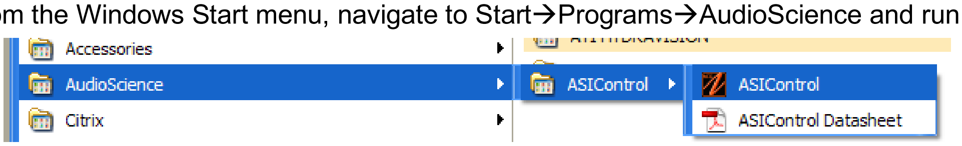
9.3 Applications for Windows

AudioScience provides ASIControl for adapter set-up and configuration.

9.3.1 ASIControl

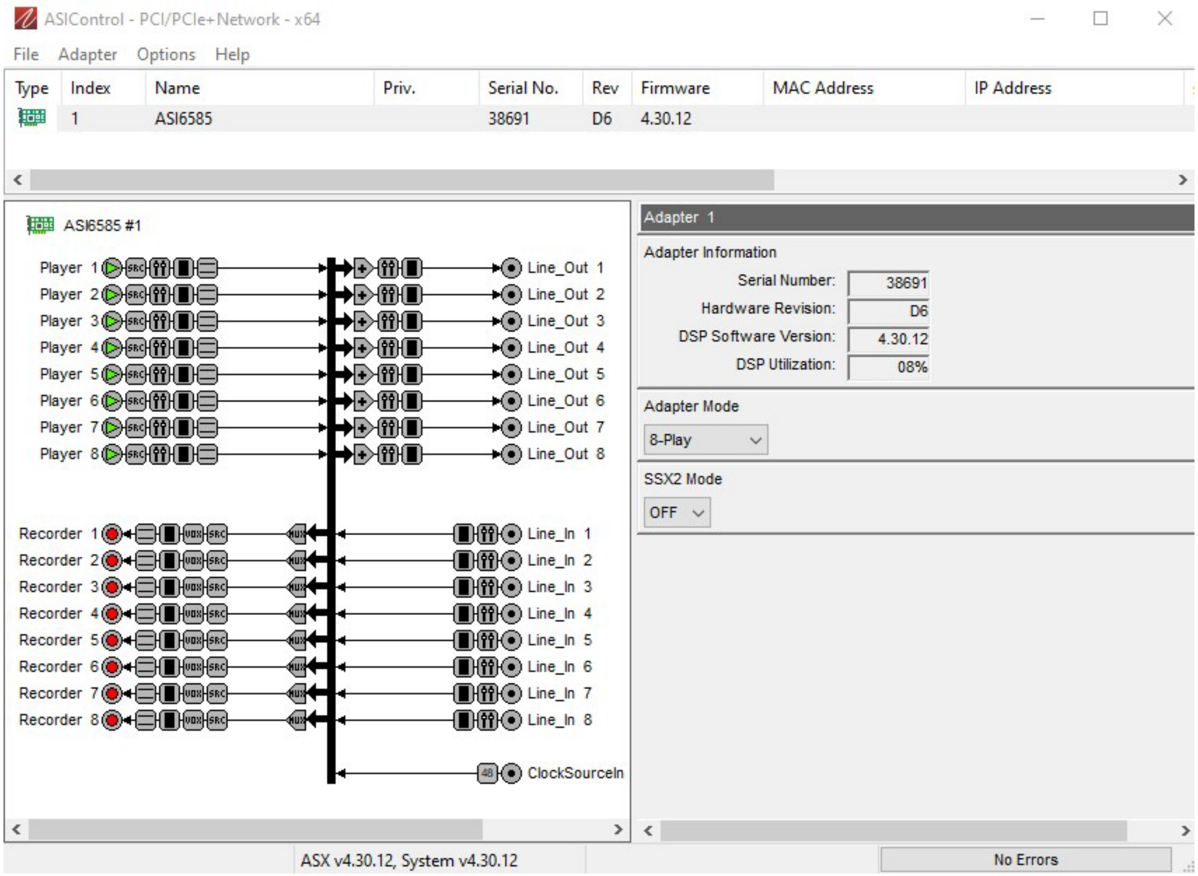
All Windows drivers install an AudioScience application called ASIControl that can be used to setup and verify functionality of adapters. ASIControl provides a common interface for users across all driver types.

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



10 OPERATION USING ASICONTROL

Using ASIControl, the ASI6685 will look similar to this:



11 USER INTERFACE

11.1 ASIControl Layout

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.

11.1.1 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.

11.1.2 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	Player	CobraNet In

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

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.

11.1.3 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.

The ASI6585/ASI6685 supports six adapter modes, 4-Play, 8-Play, 12-Play, 16-Play, Mono, and Low Latency.

Using ASIControl (installed with driver), select the Mode by clicking on “ASI6585/ASI6685” in the left-hand pane of ASIControl and selecting the mode in the Adapter_Mode box in the right-hand pane. A reboot is then required after changing from one mode to another.. The mode setting is saved on the adapter’s EEPROM.

The following table shows the number of record, input, and output streams, as well as the mixing capabilities of4-, 8-, 12-, 16-Play, and Mono adapter modes:

	4-Play mode (Default)	**8- Play Mode**	**12-Play Mode**	**16-Play Mode**	**Mono Mode**	**Low Latency**
Record Streams	4	8	4	16	16	
Stereo Livewire Inputs	4	8	4	8	16	
Stereo Livewire Outputs	4	8	4	8	16	
Mixing Capability	Full	Full	Full	Full	Full	
ASI6585 Driver required	3.08.00 and later	3.08.00 and later	3.08.00 and later	3.08.00 and later	4.02.00 and later	4.06.00 and later
ASI6685 Driver required	4.08.00 and later	4.08.00 and later	4.08.00 and later	4.08.00 and later	4.08.00 and later	4.06.00 and later
Notes		Use this mode to utilize all Livewire Sources and Destinations		It is recommend ed that play and records formats be constrained to 48kHz mono PCM if all 16 devices are to be used.		This more supports a single multichannel audio stream enabling live sound processing in ASIO and Core Audio applications. See the Low Latency Mode Datasheet for further information.

11.2 SSX2 Mode

The AudioScience Surround Sound eXtension v2 (SSX2) mode control changes the players/recorders of an adapter to be able to play/record multichannel files of 2, 4, 6, or 8 channels. Implementing SSX2 mode is slightly different depending on what driver version is installed with the AudioScience adapter' see below.

SSX2 Mode and Adapter Mode can be used in conjunction with each other. Set the required Adapter Mode (Mono mode cannot be used with SSX2 Mode), set SSX2 Mode to On and then reboot. For example, an ASI6518 set to "16-Play" in Adapter Mode and "On" in SSX2 Mode will show 4 multichannel players after reboot. An ASI6518 set to "8-Play" in Adapter Mode and "On" in SSX2 to on will show 2 multichannel players after reboot.

Note that in ASIControl, the Player volumes cannot be unlocked to move the left and right channels independently when an adapter is in SSX2 mode. For further information on SSX2, see its datasheet under the Technology section at www.audioscience.com.

11.2.1 Enabling SSX2

11.2.1.1 Interface

Figure 1. SSX2 Mode seen in right side of ASIControl.

Selecting "On" using the dropdown arrow changes the SSX2 Mode setting. A reboot is necessary after changing the mode setting. The mode setting is saved to the adapter's EEPROM. After rebooting, one multichannel play or record stream will be created for each 4 play or record streams on the adapter.

11.3 Player

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

11.3.1 Interface

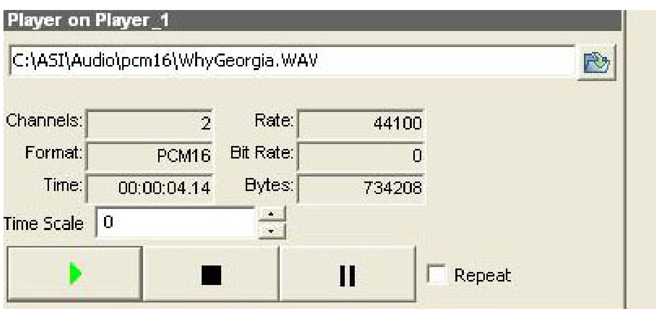


Figure 2. A player in ASIControl.

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.

11.3.2 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. The file information is also filled in and contains the following fields: **Channels**, **Rate**, **Format**, and **Bit Rate**. Most of these 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.

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.

11.3.3 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.

11.3.4 Developer

11.3.4.1 Windows APIs

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

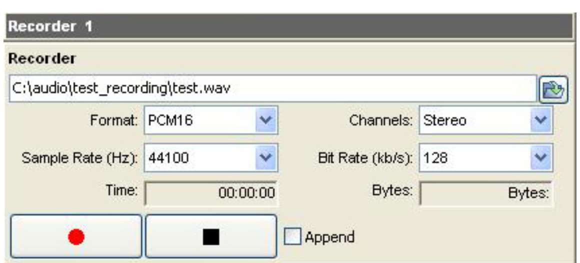
11.3.4.2 Linux APIs

HPI – Output stream functions documented here.
ASX – ASX Player control functions documented here.
ALSA – https://www.alsa-project.org/alsa-doc/alsa-lib/

11.4 Recorder

The Recorder control supports recording of an audio file.

11.4.1 Interface



A recorder in ASIControl.

The first box contains the name given to the recorded file and the location where it is to be saved. Below that is the file information, the record time and record bytes, the recorder control buttons and the file Append option.

11.4.2 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.

11.4.3 Developer

11.4.3.1 Windows APIs

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

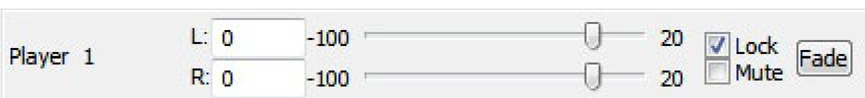
11.4.3.2 Linux APIs

HPI – use HPI_InStreamxxx() functions.
ASX – use ASX_Recorder_xxx() functions.
ALSA – https://www.alsa-project.org/alsa-doc/alsa-lib/

11.4.4 Volume

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

11.4.4.1 Interface



A Player volume 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.

11.4.4.2 Developer

11.4.4.2.1 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.

11.4.4.2.2 Linux APIs

HPI –HPI_Volume APIs.
ASX –ASX_Volume APIs.
ALSA – https://www.alsa-project.org/alsa-doc/alsa-lib/

11.5 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.

11.5.1 Interface

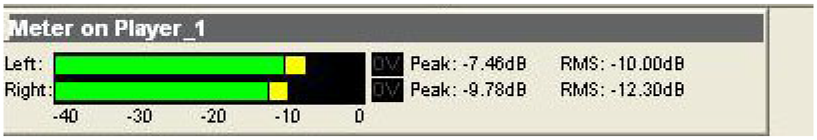


Figure 3. A stereo peak meter display. The RMS is the green bar and the peak is the yellow bar.

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.

11.5.2 Developer

11.5.2.1 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.

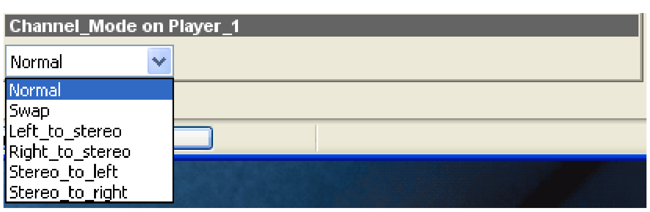
11.5.2.2 Linux APIs

HPI – Meters are read using the HPI_Meterxxx() API.
ASX – Meters are read using the ASX_Meter_xxx() API.
ALSA – https://www.alsa-project.org/alsa-doc/alsa-lib/

11.6 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.

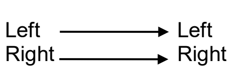
11.6.1 Interface



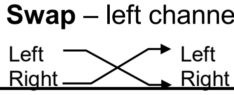
ASIControl view of a player’s channel mode control.

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:

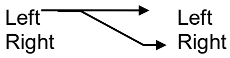
Normal – left channel out left channel, right channel out right channel



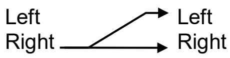
Swap – left channel out right channel and right channel out left channel



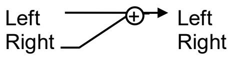
Left_to_stereo – left channel out to both left and right channels



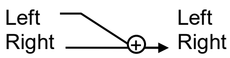
Right_to_stereo – right channel out to both left and right channels



Stereo_to_left – left and right channels out to left channel



Stereo_to_right – left and right channels out to right channel



11.7 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.

11.7.1 Interface

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
- Blu link

Adapter Rate: Displays the current adapter operating rate

12 LIVEWIRE SETUP

1. The ASI6585/ASI6685 contains a fully functional Livewire interface. By default the static IP address of this interface is set to 192.168.1.182. To access the Livewire interface, run your favorite web browser and enter http://192.168.1.182. The default IP address of the ASI6585/ASI6685 can be changed using the web browser.

2. A dialog will pop-up requesting user name and password. The default username is “user” and the password is blank (i.e. there is no password).

3. Livewire Sources:

Source				Streams	Inputs
#	Name:	Channel:	Shareable:	Mode:	Gain [dB]:
1	SRC 1	18201	Yes	Fast Stereo	0.0
2	SRC 2	18202	Yes	Fast Stereo	0.0
3	SRC 3	18203	Yes	Fast Stereo	0.0
4	SRC 4	18204	Yes	Fast Stereo	0.0
5	SRC 5	18205	Yes	Fast Stereo	0.0
6	SRC 6	18206	Yes	Fast Stereo	0.0
7	SRC 7	18207	Yes	Fast Stereo	0.0
8	SRC 8	18208	Yes	Fast Stereo	0.0

Apply

4. Livewire Destinations:

Destinations				Outputs	
#	Name:	Channel:	Type:	Load:	Gain [dB]:
1	DST 1	18401 <SRC 1@ASI6585Test>	From source	hi-Z (default)	0.0
2	DST 2	18402 <SRC 2@ASI6585Test>	From source	hi-Z (default)	0.0
3	DST 3	18403 <SRC 3@ASI6585>	From source	hi-Z (default)	0.0
4	DST 4	18404 <SRC 4@ASI6585>	From source	hi-Z (default)	0.0
5	DST 5	18405 <SRC 5@ASI6585>	From source	hi-Z (default)	0.0
6	DST 6	18405 <SRC 5@ASI6585>	From source	hi-Z (default)	0.0
7	DST 7	18406 <SRC 6@ASI6585>	From source	hi-Z (default)	0.0
8	DST 8	18407 <SRC 7@ASI6585>	From source	hi-Z (default)	0.0

Apply

NOTES:

1. The meters on the Meters page are functional with firmware version 2.7.1d_r2 or later.

12.1 IP address recovery

In the event that the IP address of an ASI6585/ASI6685 is unknown, a combination of the ASI6585/ASI6685 MAC address and bootps.exe can be used to set the IP address back to a known state. The following steps detail setting the IP address back to the default shipping IP address and netmask: (this utility should be run from a network connected system, not necessarily the physical system where the card is installed unless it too is connected to the same IP network)

1. Look on the back of the orange faceplate (the daughterboard) of the ASI6585/ASI6685 and record the MAC address printed on the white, AudioScience part number label. This example uses 00:60:2B:06:D1:DA.
2. Download bootps.exe software from here: http://www.audioscience.com/internet/products/livewire/bootps.exe
3. To set the ASI6585/ASI6685 adapter IP address, from a Cmd prompt go
    - `bootps –m 255.255.255.0 –set 00-60-2b-06-d1-da/192.168.1.182`

Typically, the ASI6585/ASI6685 does not need to be restarted for the new IP address to be "active". In the event that the PC cannot connect to the new IP address, try rebooting the PC that has the ASI6585/ASI6685 in it. If the new IP address still does not work, run bootps.exe a second time.

13 GPIO

Note:
– GPIO functionality requires firmware version 2.5.2d_r2 or later. See Section 14.2 for a link to current firmware.
– GPIO handling on an automation network requires either an extra NIC in the PC (see Section 14.1) **or Axia's PathfinderPC Router Control Software** (see Section 14.4)**.**

The ASI6585/ASI6685 supports 8 channels of "virtual" GPIO. It similar to the Axia IP audio driver and behaves the same. The GPIO is comprised of 8 GPIO inputs and 8 GPIO outputs. The GPIO states can be viewed from the device's web interface and controlled via telnet. The following is from the Axia documentation describing the command set used to control GPIO.

13.1 GPIO Commands

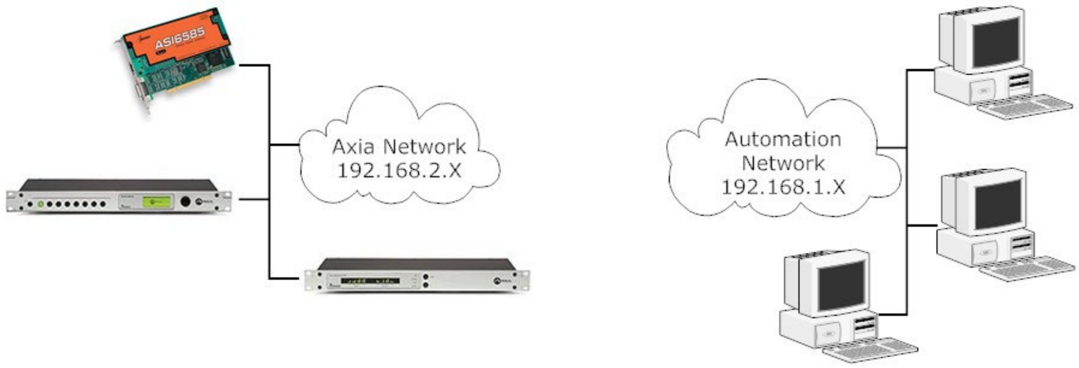
ADD GPO - Enables indications of GPO pin changes. Those are control messages for the auto-mation system.
ADD GPI - Enables indications of GPI pin changes. Those are control messages sent from the automation system. This is useful when multiple clients control one GPI port.
GPI <state> - Automation system emulates input pin state changes. A GPIO event will be sent to Axia network. Note: client must send LOGIN command before issuing GPI.
LOGIN - Enables commands that change state of the device from client. Required before issuing active commands.

13.2 GPIO Responses

State of GPIO port will be presented in simple text. The ADD command will generate a response of all pin states and initiate a subscription for future state changes. The response of pin changes with follow the following syntax:GP{I/O} {#} {$$$$$}

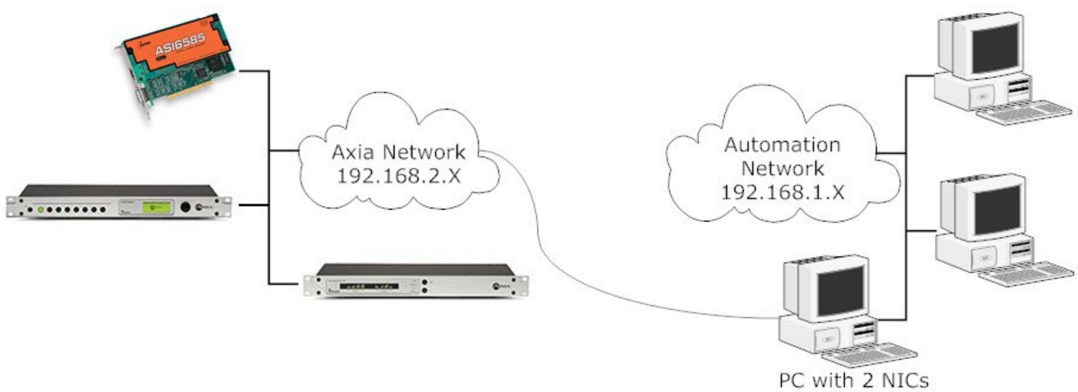
Examples
GPO 1 hhhhh - all 5 pins of device 1 are state high
GPO 2 Lhhhh - pin 1 of device 2 just changed to a Low state
GPO 3 hlhhh - pin 2 is low and all others are high. No new states.
GPO 4 hlhhH - device 4 has a previous low state on pin 2 and a new High state on pin 5.

14 AXIA NETWORK TO AUTOMATION NETWORK ROUTING OPTIONS



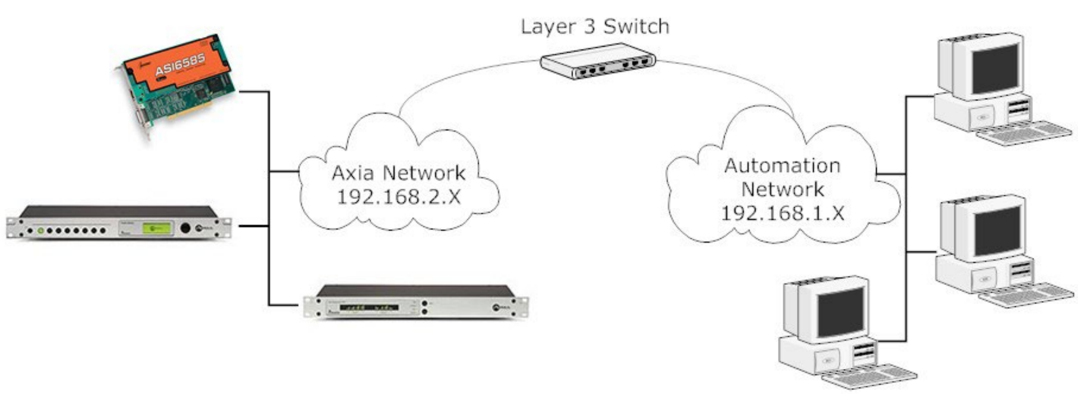
Since the ASI6585/ASI6685's network interface is not directly accessible from the automation network PC, a network path from the automation network PC to the Axia network containing the ASI6585/ASI6685 should be implemented. The following sections outline the possible approaches. In all the descriptions that follow the Axia network is assumed to support IP addresses in the 192.168.2.xxx range, while the automation network supports IP addresses in the 192.168.1.xxx range. One of the 4 options listed below may be used.

14.1 Using an extra network card in the PC



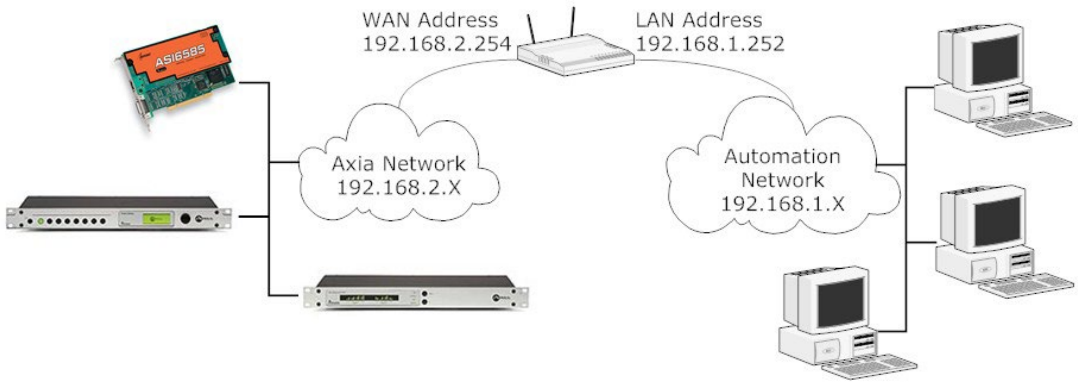
The simplest option is to add a second network card to each automation PC and assign it an IP in the Axia device IP range and then connect it to the Axia switch. The automation PC will now be able to Telnet to any Axia device in the Axia network. The automation PC does not need to have the Livewire IP driver installed to telnet to the ASI6585/ASI6685. The downside of this approach is that every PC that wishes to access the Axia network must have an additional network card installed and an extra port is consumed on the Axia router.

14.2 Using the virtual network and routing capabilities of a Layer 3 switch



A single port of a Layer 3 managed switch can be use to connect between the Automation and Axia networks. On the switch the port should be assigned to its own VLAN and then non-multicast routing enabled between the Automation VLAN and the Axia LAN.

14.3 Adding a hardware router



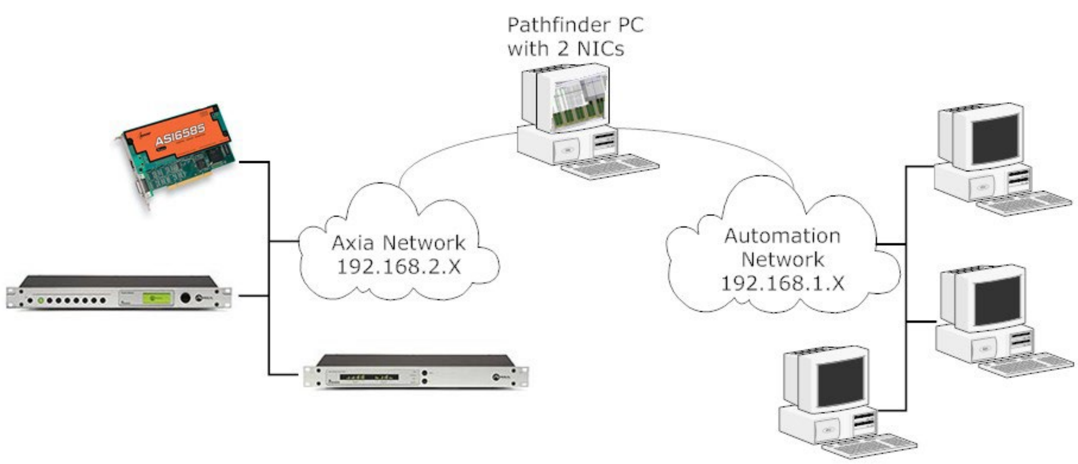
In the case where the Axia LAN switch does not support Layer 3 protocols, an external router can be added between the Axia and Automation networks. Axia outlines approaches for doing this here and here.

We have had success performing the following steps on a LinkSys WRT54G router.

1. Disable wireless.
2. Set the internet (WAN) to static IP of 192.168.2.254.
3. Set the Local IP (LAN) to 192.168.1.252.
4. Disable DHCP
5. In Advanced Route set the operating mode to "Router"

Plug the Automation LAN into one of the 4 LAN ports and the Axia LAN in to the WAN port. On the automation PC use
C:\>route add 192.168.2.0 mask 255.255.255.0 192.168.1.252
to set up route to the Axia network.
You should now be able to ping and or telnet to the Axia devices as if they were on your local network.

14.4 Using PathFinder



Run Axia's PathfinderPC Router Control Software (see http://www.axiaaudio.com/components/default.htm#Pathfinder for more information) on a PC somewhere on the network so that GPIO information from the Livewire network can be handled on the automation network. Axia can provide more information of the exact components and configuration required to form the required connection.

15 LIVEWIRE FIRMWARE UPDATES

NOTE Please note that any attempt to load the Axia AES, Analog, GPIO, Microphone, or Router Selector Node firmware (found on Axia's website) onto an ASI6585/ASI6685 will render the ASI6585/ASI6685 inoperable and it will need to be sent back to AudioScience for repair. Use only the AudioScience recommended firmware.

The latest current firmware version for the ASI6585/ASI6685 is v2.7.1d_r2.

AudioScience is aware that the firmware version for Axia nodes found on the Axia website may be higher.
If the firmware on the Axia website is higher, we have been assured by Axia that the improvements in the Livewire firmware found on their website does not affect the ASI6585/ASI6685.

NOTE New products should ship with version v2.7.1d_r2 already installed.

The ASI6585/ASI6685's Livewire interface has two internal memory "banks" for storing the Livewire firmware. Each bank contains room for a complete version of the firmware. This approach allows a software update download to be completed and checked without danger of making the unit inoperable if the download were to be incomplete or corrupted. It also provides an easy way to try a new software version and still return to the old version.

The software version in each bank is displayed from the System web page. The lower half of this screen, shown below, shows the current firmware versions and allows you to select what version will be used at startup. To change banks simply click in the "radio button" for the desired bank and then click on Apply.

IMPORTANT! The node will reboot after you click Apply if you change the software version. This will result in loss of audio locally, and at any unit using the local sources.

15.1 Saving Bank 1 Software

Software is always downloaded to bank 1 (the secondary bank). Downloading software (see below) will overwrite the software currently in this bank, if any. If you wish to save the software currently residing in bank 1 you can save it by moving it to bank 0 as follows:

1. Click on "Commit this version to Bank 0" box (see picture below). (If the Commit checkbox is not showing, select Bank 1 and click Apply to bring it up.)
2. Click on Apply. The node will now reboot.

15.2 Downloading new software

NOTE Firmware version 2.5.8a_r2 must be the current firmware version on the ASI6585/ASI6685 in order to upgrade to 2.7.1d_r2. If you have an older version of firmware you will need to run incremental updates to bring the firmware to the latest version. If you currently have a version older than 2.5.8a_r2 and need to update please see section below entitled Firmware version upgrade requirements or contact AudioScience technical support for assistance.

A new software version can be downloaded into Bank 1 as follows:

1. Visit https://www.audioscience.com/internet/products/livewire/asi6685.htm to download the firmware to your computer (this should be the computer that you will use to access the ASI6585/ASI6685's web page). Your local computer operating system should display a prompt to permit you to choose where you wish to locate the downloaded file. You can choose any convenient location, just be sure to note the drive and location where the file is to be saved.

2. Open a web browser and connect to the ASI6585/ASI6685 to be updated. Enter the complete path and file name for the firmware file or click on the Browse button to locate the file. Once the proper path and filename are displayed, click on Apply to download the file.

3. 3. A successful download will be indicated by the new version being displayed in the Bank 1 field. If the download were unsuccessful the field for Bank 1 would be blank.

15.3 Firmware Release Notes

ChangeLog (from Axia)

Version 2.7.1d_r2 (patch 2018-01-25)

Updated meter page to work without Java.

Version 2.5.8a_r2 (patch 2009-09-09)

Selector: Fixed output volume jump. When the volume knob was used, load and out gain settings were ignored.

Version 2.5.7a (patch 2009-02-24)

Advertisement daemon protected from crashes caused by invalid input messages.

Version 2.5.5a (patch 2009-01-08)

GPIO Node: Fixed memory leak in gpior. Memory leak was occurring every time connection to remote snake end point failed. Memory leak was increasing as long as the invalid endpoint was configured causing GPIO message processing to stop.
Optimized channel list processing: building advertisement database and updating Router Selector front panel scroll list.

Version 2.5.4a (patch 2008-12-22)

Optimized channel list processing: building advertisement database and updating Router Selector front panel scroll list.

Version 2.5.3a (patch 2008-08-07)

Fixed HEAD request processing in the WEB server. In previous versions, processing HEAD was causing server crash and audio glitch.
Added additional security checks to processing BOOTP responses to avoid accepting incorrect network mask and packets from DHCP servers. Some DHCP servers would respond to BOOTP requests sent by nodes leading to unusable network mask configuration.
Fixed: Accepting new IP parameters from BOOTPS was deleting the hostname.

Version 2.5.2g (release 2008-07-15)

Livewire master priority factory default changed from 0 (disabled) to 3. This change makes initial node configuration easier. Users who do not use the mode switch on the front panel and used the WEB interface to do the entire configuration tend to forget to enable the mastership on the QoS page resulting in having no clock in the network. The new default enables the clock automatically. It is no longer safe to plug in a new node to non-qualified network.
Selector: Case insensitive channel list sort.
Fixed: WEB page title was not showing hostname if its length was maximum (12 characters).
Hostname: letters, digits, hyphen only enforced. All characters entered by user, which are not allowed are converted to the hyphens (-).
Fixed HTTP redirection problem after changing the hostname. Automatic redirection did not work in previous version and user had to click the redirection link manually.
Critical bug found in release 2.5.2f. Livewire Routing Protocol server was crashing when changing properties of a single audio destination other than

DST 1. That includes audio route change command typically issued by Path Finder.

Version 2.5.2f (release 2008-05-01 canceled 2008-05-01)

Added Syslog messages notifying about changes in AES inputs synchronization state, as well as AES/Analog automatic switch.
Selector: Improved Analog/AES input automatic selection logic, so the switch to analog is delayed until after the end of reset/verify cycle. This was temporary glitch in AES signal does not cause switching to analog input and long audio gap.
Selector: Bugfix: Since 2.3.4a missing or unstable AES input was causing interruptions of the analog input signal.
AES input chip restart made quicker, so a glitch in AES signal caused by a PLL synchronization loss is short.
Destinations WEB page: In surround bundle name column is hidden for consistency with Sources WEB page.
Bugfix: Nodes use Standard Streams QoS settings for audio Livestreams after changing destination.
Selector: Screen update optimization preventing flicker when operating volume control.
Selector: Front panel System screen Stream Mode terminology change from SLOW/FAST to Standard/Live.
In November 2004 (1.2.25 software) users: "USER", "axia", "AXIA" and "Axia" were added. Setting password for those results in password change operation failure. Improved software to handle old units. Possible workarounds for old units are:
1. Copy default passwd file to add missing users. From telnet: 'cp /etc/default/passwd /etc/config/'. No restart is necessary.
2. Reset unit to factory defaults. On audio nodes press and hold SELECT and ID buttons on power up. On GPIO node, telnet in, and do the following 'cp /etc/default/* /etc/config/', then 'reboot'.

Version 2.5.2e_r2

Add adapter type information to prevent loading incorrect firmware.

Version 2.5.2d_r2
Changes:
Dual mono implementation.
Stream mode description changed from Fast/Slow to Live/Standard.
Added “Virtual GPIO”, so automation system can use Livewire IP-only GPIO interface to communicate with consoles and GPIO nodes.
Added support for Livewire Sync LED.

Fixes:
Web based meters now work.
iProbe support: basic device information, audio and GPIO configuration.
802.1p and TOS audio packet marking. Since 2.3.x, values set for Standard Streams were also applied to Live Streams.

Version 2.3.3b
Initial release

15.4 Firmware version upgrade requirements

In most cases you cannot upgrade from a very old version of firmware to the most recent version. In those situations you need to update to an intermediate versions first. Below are 2 example of how to accomplish this.

15.4.1 Instructions on upgrading to 2.5.8a_r2

Firmware version 2.5.2e_r2 must be the current firmware version on the ASI6585 in order to upgrade to 2.5.8a_r2. If it is not, see section on installing firmware version 2.5.2e_r2.

15.4.2 Instructions on upgrading to 2.5.2e_r2

There are two steps to loading firmware version 2.5.2e_r2 from 2.5.2d_r2 or 2.3.3b:
Step 1 Setting the node type of the ASI6585
Step 2 Downloading the firmware into Bank 1 (found in the ASI6585's web interface)
The following describes how to do Step 1. Follow the steps in the "Downloading new software" section for Step 2.

Step 1
Firmware version 2.5.2e_r2 checks the Livewire node type before it loads. ASI6585 Livewire firmware versions earlier than 2.5.2e_r2 do not have the node type set correctly (2.5.2d_r2 and 2.3.3b). The following steps describe a manual method of setting the node type so that the upgrade to firmware version 2.5.2e_r2 from 2.5.2d_r2 or 2.3.3b is successful.

The node type information is stored in a file called product.id. This file must be edited so that it contains the correct node type information. This is done via a Telnet session. The steps below describe how to Telnet from Windows to the ASI6585 and edit the product.id file.

Information required before proceeding: the IP address of the ASI6585 and the username/password to access the ASI6585's web interface. The factory default settings are as follows: IP address – 192.168.1.182, username is 'user' and password is <blank> (i.e.: no password). Check to see if any of these settings for your ASI6585 have been changed.

How to Telnet from Windows to the ASI6585 and edit product.id (presumes factory default settings)

1. Click Start→Run. This opens the Run dialog box.
In the Open textbox, type **telnet 192.168.1.182** and click OK.

2. A command line window opens with a login prompt.
Type in 'user'. Press the Enter key on your keyboard.
(If you previously set a password for your ASI6585 web interface, you will be prompted for it now.)
The **/home>** prompt is displayed.

3. Type **vi product.id** and press the Enter key on your keyboard.
This opens the "product.id" document so it can be edited.
There is only one line in product.id - "liveio.r2."

4. Press <Shift><C> on your keyboard. The "2" in liveio.r2 will be replaced with a $.
This means that the text 'liveio.r2' can now be overwritten.

5. Type **asi6585.r2** then press the <Enter> key on your keyboard to take you to the next line.
Type **ASI6585** then press the <Esc> key on your keyboard.

6. Type **:x** then press the <Enter> key on your keyboard to save the changes and
get back to the **/home>** prompt.

7. Type **exit** and press the <Enter> key on your keyboard.
The command line window will close.

[end]