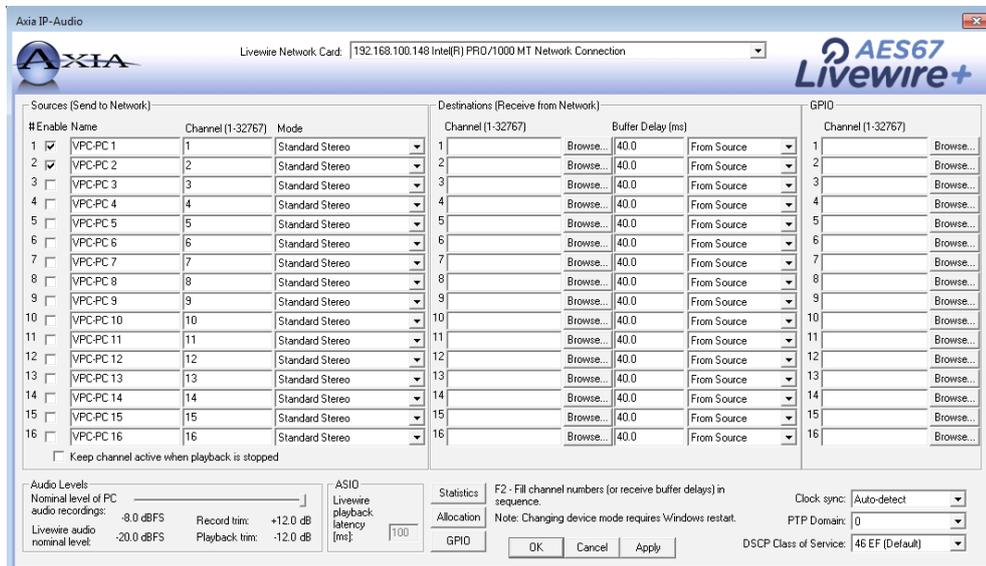


iP Audio Driver

Installation & User's Guide



Manual Rev 2.10 March 2017
p/n 1490-00042-011

iP Audio Driver Manual

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VIA WORLD WIDE WEB:

The Axia Audio web site has a variety of information which may be useful for product selection and support. The url is telosalliance.com.

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War of the Waves

Dear Valued Customer,

It's with great pride and a tip of the hat to an incredible team that I congratulate you on your new Telos Alliance product. Everything we do here at the radio division of the Telos Alliance is with one end goal in mind: To help broadcasters declare victory in extremely competitive environments. By purchasing this product from us, in essence, you have declared war on your competition.

After all, the majority of Telos Alliance employees were broadcasters themselves once, and the products we've developed over the years have been designed as solutions to specific issues faced on the front lines of our industry. We're right there in the trenches with you and have the weapons you need in your arsenal.

Telos Systems is a catalyst to out-of-this-world sound, with the most powerful and popular broadcast telephone systems in the industry; IP/ISDN codecs and transceivers; plus processing/encoding for streaming audio. We built an industry on the back of these amazing telephony systems, and they are still going strong.

While we at the Telos Alliance never forget our roots, we are also blazing trails in terms of new technologies like stream-encoding and AoIP, so that all types of broadcasters can excel in this ever-evolving digital world. Omnia Audio not only lets you stand out on the dial with your unique signature sound via legendary audio processors, audio codecs, and microphone processing, it lets you give your listeners a better streaming experience across devices with innovative stream encoding/processing software and hardware.

Axia Audio is a driving force behind the AES67 AoIP standard, and its networked AoIP radio consoles, audio interfaces, networked intercom, and software products continue to move AoIP adoption forward and help broadcasters streamline operations with cohesive, smart, and feature-rich AoIP ecosystems.

Last, but certainly not least, 25-Seven has traditionally been known for its audio delays, but its Voltair watermark monitor/processor has made a name for itself more recently as the disruptive product that helped broadcasters take back their ratings and harness the true power of their listening audiences.

You work so hard on your programming day-in and day-out, it deserves technology that will optimize sound and performance at every point in the airchain and online. Armed with Telos products, you have what you need to set your competition squarely in your crosshairs.

With that, I'll leave you to prep your armaments. I hope that you will enjoy your Telos Alliance products for many years to come!

Sincerely,
Frank Foti
CEO, The Telos Alliance



Chapter 1:

Introducing the Axia IP Audio Driver

The Axia IP-Audio Driver is the software interface between your PC's audio applications and the Livewire network. It provides the following functions:

- Sends audio “sources” to the Livewire+ network from PC/Windows audio applications such as multichannel delivery systems and other audio players.
- Receives audio from the Livewire+ network to “destinations” on the PC/Windows system such as audio recording applications.
- GPIO' function conveys “button-press” data from the Livewire network to “destination” applications; i.e., a console fader start button can command a PC/Windows-based audio player to start playback

Versions of the IP-Audio driver appear to Windows as standard audio input/out devices. There are wide variety of applications where Axia Livewire+ hardware and the Windows software are used together, and the Driver provides all software components needed to record and play Livewire+ streams using standard recording software for Windows.

The Axia IP driver package also contains a GPIO module that allows controlling many Windows audio applications from Axia consoles. This feature is commonly used for “device start” commands initiated by pressing the Channel ON button on the Axia control surface.

Axia IP-Audio Driver Retail vs OEM

The Axia IP-Audio Driver is available in different configurations to satisfy different functional and financial needs of clients. The two classes are Retail and OEM. OEM versions are available through software partner companies who can provide support on interfacing with their application.

Retail

1 Input, 1 Output

A basic version which is useful for general purpose applications.

4 Input, 4 Output

A version to be used where the audio application needs more than a single i/o to the computer.

OEM

8 Inputs, 8 Outputs

Lower cost option with OEM partners where the application isn't more than 8 stereo channels.

24 Inputs, 24 Outputs

Full implementation of the Axia IP-Audio Driver. Some performance variance due to hardware used.

Other than the number of audio channel support and GPIO functionality, the drivers behave the same. The following information in this manual applies to all versions.

The Axia IP-Audio Driver supports the WDM (Windows Driver Model) and ASIO interfaces.

Chapter 2:

Installation and Configuration

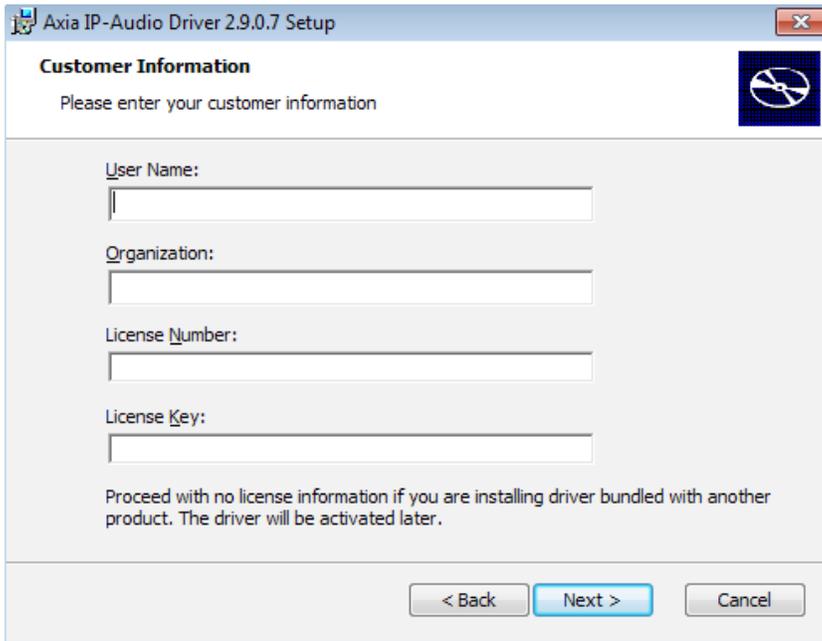
Starting with version 2.9 of the driver, legacy Microsoft operating systems are excluded from support. With 2.9 and above, the following versions are supported:

- Microsoft Windows 10 all Editions (32-bit and 64-bit)
- Microsoft Windows 8.1 all Editions (32-bit and 64-bit)
- Microsoft Windows 8 all Editions (32-bit and 64-bit)
- Microsoft Windows Embedded 8 (32-bit and 64-bit)
- Microsoft Windows 7 all Editions (32-bit and 64-bit)
- Microsoft Windows Vista all Editions (32-bit and 64-bit)

Anytime you are upgrading from a previous version of the IP-Audio Driver, it is *strongly recommended* that you uninstall the older version and reboot the machine prior to installing any new version.

Installation Steps:

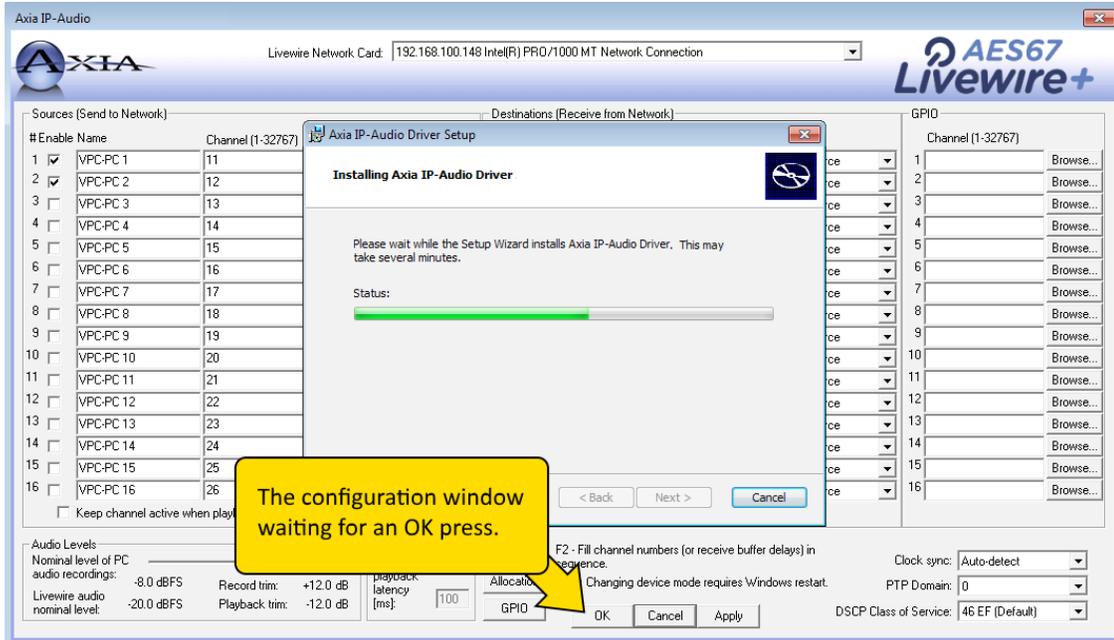
1. Download the installation package from the AxiaAudio.com site or FTP server. The file will be identified with axiaipdrv-{version} as an msi installer package or a compressed zip file. It may also be identified as 3001-00017-{revision} which is a Telos Alliance part number for the installer. If the file is compressed, please extract the msi file. Starting in Windows 10, there is a need to use the Windows 10 special installer. Earlier versions should use the non-Windows 10 installer.
2. Execute the installer package, msi file. If this is an upgrade from a previous version, you must uninstall the previous version and reboot the machine before doing this step.
3. The installation process will take you through a few windows in order to install the driver. Enter the appropriate information when requested.



The screenshot shows a Windows installer window titled "Axia IP-Audio Driver 2.9.0.7 Setup". The window has a standard Windows title bar with minimize, maximize, and close buttons. The main content area is titled "Customer Information" and contains the instruction "Please enter your customer information". There are four text input fields: "User Name:", "Organization:", "License Number:", and "License Key:". Below these fields is a note: "Proceed with no license information if you are installing driver bundled with another product. The driver will be activated later." At the bottom of the window, there are three buttons: "< Back", "Next >", and "Cancel".

NOTE: Licensing information is case sensitive. Insure you have entered the information correctly!

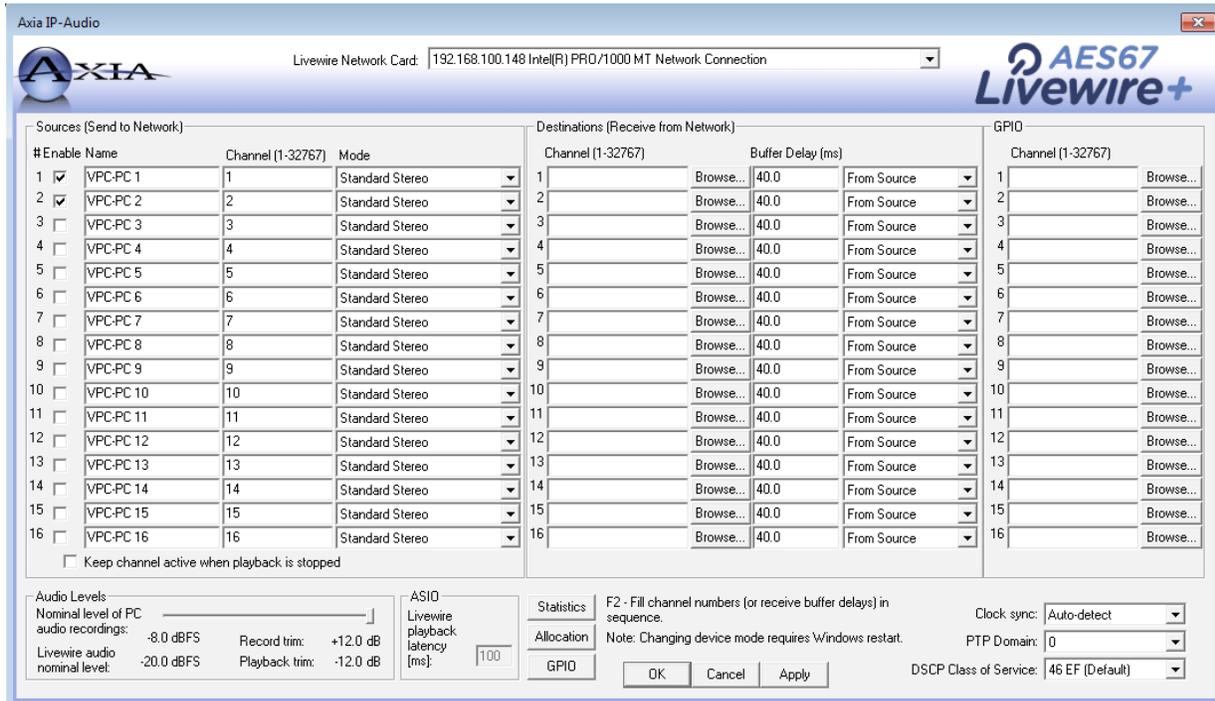
4. A progress bar window will present itself to provide feedback on the progress. In some cases it may appear stalled because a window has opened waiting for user input. Before assuming the process has failed, make sure all active windows have been acknowledged. Closing the installation process premature can lead to a corrupted install.



5. Once the install is completed successfully, there will be a notification window providing the option to press the Finish button. The IP-Audio Driver is installed. The next step is the configuration.

Configuration

The configuration window appears as part of the installation process. If you didn't use that opportunity to configure the IP-Audio Driver and instead selected OK to move on with the install, you should locate the Axia IP-Driver configuration in your program listing of the computer and run it. The configuration window is as follows:



Livewire Network Card

At the top of the window is a drop down selector providing the option to select which NIC (Network Interface Card) is to be used with the driver. It is typical to see computers with multiple NICs, one NIC for Livewire+ and another for the general office network (internet or playout file access).

Sources (Send to Network)

Sources are audio streams generated into the network. The computer playout will become these sources into the network, instead of playing out of a sound card into your analog equipment. The sources section of the window has 5 items of interest.

#	Enable	Name	Channel (1-32767)	Mode
1	<input checked="" type="checkbox"/>	VPC-PC 1	1	Standard Stereo
2	<input type="checkbox"/>	VPC-PC 2	2	Standard Stereo
3	<input type="checkbox"/>	VPC-PC 3	3	Standard Stereo
4	<input type="checkbox"/>	VPC-PC 4	4	Standard Stereo
5	<input type="checkbox"/>	VPC-PC 5	5	Standard Stereo
6	<input type="checkbox"/>	VPC-PC 6	6	Standard Stereo
7	<input type="checkbox"/>	VPC-PC 7	7	Standard Stereo
8	<input type="checkbox"/>	VPC-PC 8	8	Standard Stereo
9	<input type="checkbox"/>	VPC-PC 9	9	Standard Stereo
10	<input type="checkbox"/>	VPC-PC 10	10	Standard Stereo
11	<input type="checkbox"/>	VPC-PC 11	11	Standard Stereo
12	<input type="checkbox"/>	VPC-PC 12	12	Standard Stereo
13	<input type="checkbox"/>	VPC-PC 13	13	Standard Stereo
14	<input type="checkbox"/>	VPC-PC 14	14	Standard Stereo
15	<input type="checkbox"/>	VPC-PC 15	15	Standard Stereo
16	<input type="checkbox"/>	VPC-PC 16	16	Standard Stereo

Keep channel active when playback is stopped

1. **Enable** check box – used to enable/disable the source stream. For each audio output from the computer you desire, enable.
2. Stream **Name** – used to associate a name with the stream. Each audio device participates in advertisement of sources and the name here will be part of that process. The advertisement of sources will be in the format of {Name}@{device name} where device name is the host machine.

3. Stream **Channel** – each source needs a unique numerical identifier. The number will translate into a multicast stream, thus needs to be unique to the network. A good practice is to use some derivative of the IP address. For example if the IP address of the NIC is 192.168.2.100, the first stream could be 10,001 (100 + 01). Pressing F2 will numerically add the next instance.

The channel number of 10001 will equate to a MC address of 239.192.39.17.

4. Stream **Mode** – is the option to define if the stream is “Standard Stereo” (5ms of audio per packet), “Low-Latency Stereo (AES67)” (1ms of audio per packet), or “Surround (8-channel)” (1.25ms of audio per packet). Livewire incorporated support for surround much earlier than market demand. Most installs use the default Standard Stereo.
5. **Keep channel active** check box – is a utility function which disables streams if there is no audio present. We have found that some prefer to keep the stream active even if there is no audio. If this is the case for you, make sure this box is checked.

Destinations (Receive from Network)

Network audio that would be inserted into a computer application would be selected in the Destinations field. Either the computer is used for recording audio content from the network or the computer is used as a codec through the internet and using a communication application which needs a mic source or mix-minus source. There are 3 items of interest.

– Destinations (Receive from Network)

	Channel (1-32767)	Browse...	Buffer Delay (ms)	
1	1	Browse...	40.0	From Source
2	2	Browse...	40.0	From Source
3	3	Browse...	40.0	From Source
4	4	Browse...	40.0	From Source
5	5	Browse...	40.0	From Source
6	6	Browse...	40.0	From Source
7	7	Browse...	40.0	From Source
8	8	Browse...	40.0	From Source
9	9	Browse...	40.0	From Source
10	10	Browse...	40.0	From Source
11	11	Browse...	40.0	From Source
12	12	Browse...	40.0	From Source
13	13	Browse...	40.0	From Source
14	14	Browse...	40.0	From Source
15	15	Browse...	40.0	From Source
16	16	Browse...	40.0	From Source

1. Stream **Channel** – is the identifier of the audio stream from the network that you wish to receive. Next to the text field is a Browse button which will present a list of advertised sources to select. You may also manually type the channel number if known.
2. **Buffer Delay** – is an option to adjust the receive buffer for the streams discreetly (new in version 2.10). This is needed if dealing with poor jitter or if the stream has little jitter the delay in reception can be reduced.
3. Stream **Type** – is a new option in version 2.9.x allowing for the support of console back-feeds. The Type setting will be automatically set when selecting from the browse view. If manually entering a channel number, you may also need to manually set the type if the stream is something other than a normal stereo stream.

GPIO

New in version 2.9.x is direct access to GPIO settings. Earlier versions of the IP-Audio Driver assumed one type of application usage and thus automatically configured the GPIO based on this assumption. Due to market demand, the need for specialized configuration was added. This requires the manual configuration of the GPIO section from now on. The configuration behaves the same as the GPIO configuration of an Axia xNode. If you wish to associate the IP-Audio Driver Source with GPIO logic of a console, typically you would apply the same channel number. As example, if Source #1 is channel 10,001, your GPIO #1 would also be 10,001. If the application you are using is a GPIO route with an Axia GPIO port, you would enter in the IP address of the xNode followed by a port number in the following syntax {ip}/{port}:

192.168.2.46/2 - xNode at IP address 192.168.2.46, port 2 of that device.

Audio Levels

The slider permits global calibration of audio levels between the PC and the audio network. Livewire networks are typically run at a nominal level of -20dBFS while computer content can run significantly higher. Without calibration, audio content played into the network can be near digital clipping levels. The reverse is also true that content from the network which is inserted into computer application would appear very low in level. The slider calibrates the two domains. To achieve unity level (the same nominal levels) between the PC and the network, place the slider to the far left. This will result in no gain adjustments on input or output. The default setting is to the far right which attenuates audio ployout by 12 dB and amplifies audio input by 12 dB. Adjust the slider to what fits your facility needs.



ASIO

The field is only informative.

Button - Statistics

Pressing this button will bring up IP-Audio Driver stream diagnostic information. Refer to troubleshooting for more information.

Button - Allocation

Source allocation is a function of most Axia consoles. Pressing this button will present a pop-up showing where computer sources are assigned. This is specifically if the Source is assigned to a source profile and that profile is loaded to a channel strip of a console in the network. This functionality utilizes the Advertisement channel.

Button – GPIO

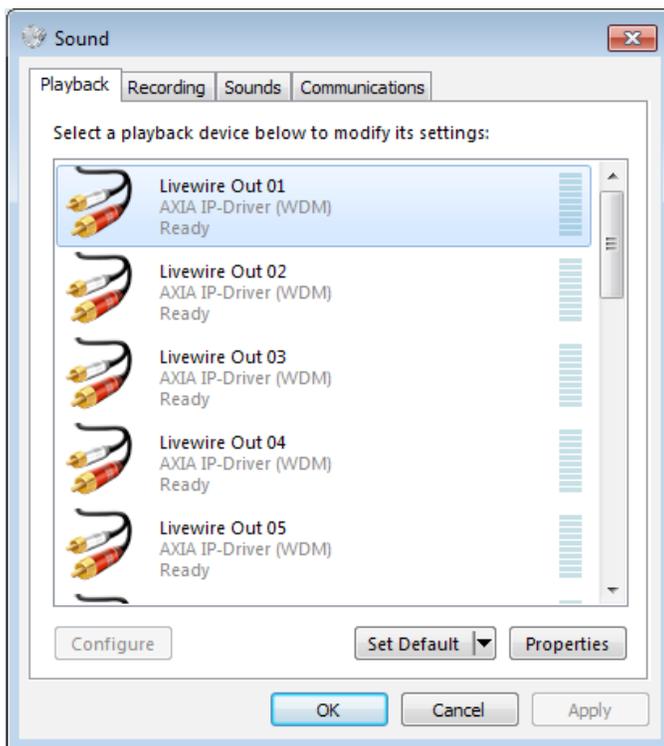
The GPIO button opens a pop-up for use of a deprecated GPIO configuration method. The window has the option to load a gpiovk file which defines an API between the IP-Audio Driver and another application running on the computer. The 5 button below allow for testing the API with said application. More information is available in Appendix B.

Network Sync

The bottom right of the configuration window has network sync options. Its default is auto detect which will sync to the first clock packet received. The manual two other options are for the standard Livewire Slave or PTP slave (used in AES67 networks). The other items are used for non-standard PTP networks which may be required as AES67 becomes more prominent.

The remainder

The IP-Audio Driver provides a virtual sound card which is able to deliver and request streams from the network. The other side of the application is how it interfaces with an application running on the computer. The application can be a basic application that uses the windows default sound card or is a more advance application that has its own configuration. In the latter case, it is the responsibility of the partner company or the customer to locate support for the remainder of the configuration. In the case of a basic application which uses the default windows sound card, you will need to enter into the windows sound card properties and select the IP-Audio Driver channel as the default sound card.



In the case of more advanced applications, our support department is not responsible for knowing all iterations of third party applications and may not know how to configure the application to select the ASIO or WDM instances provided to the OS by the IP-Audio Driver.

Appendix A:

Livewire Ports

Computers are notorious for having various protection security software installed which can block or limit network traffic. The following is a list of common ports and protocols used within a Livewire network that may be helpful for a network administrator who needs to configure security applications to allow this traffic to function.

93	TCP	Livewire Routing Protocol
2055	UDP	Multicast based GPIO (CMsg2 protocol) (mcast on 239.192.255.4) - GPIO commands from GPIO node to console-type endpoint
2060	UDP	Multicast based GPIO (CMsg2 protocol) (mcast on 239.192.255.4) - GPIO commands from console-type endpoint to GPIO node
4000	UDP	Livewire Advertisement and Source Allocation Protocol - Full info advertisement requests and source allocation requests
4001	UDP	Livewire Advertisement and Source Allocation Protocol (mcast on 239.192.255.3) - Periodic announcements and full info advertisements - Source allocation state announcements and responses
4010	TCP	Livewire Control Protocol
5004	UDP	RTP Livewire Audio
7000	UDP	Livewire Clock

Appendix B:

GPIO Interfaces

The Axia IP-Audio Driver includes Livewire GPIO support per audio source stream. The intention here is for use with radio delivery systems. The typical application is for remote control between the On-Air console and the computer. Customers have found other ways to use the GPIO and this manual does not dictate the only functionality possible.

GPIO – Control Surface Bridge

The Control Surface Bridge is a deprecated feature as it isn't widely used nor understood. It is documented here for completeness. Future versions of the driver may exclude this feature. The Control Surface Bridge is an API bridge between Axia control surface generated logic and a computer application. The bridge is not bi-directional and is only provides support from control surface to the designated application.

The Control Surface Bridge is accessible from the GPIO button from the main configuration window of the IP-Audio Driver. Pressing the button will present a new window. The window is split into two sections, the top *Configuration* portion and the lower *Test* portion.

Configuration

A Profile which defines the API bridge can be selected in the configuration section and the ability to Enable said bridge. Once a profile is selected, close the window so that the profile is able to load. More on the profile definitions below.

Test

The lower *Test* section of the window provides the option to verify the API bridge is controlling the computer application. The section offers ability to select one of several “Devices”, where a device is one of the source stream – GPIO associated entities. As example, if the version of IP-Audio Driver is a 4x4 version, the supported devices would be 4. The remaining items are the 5 buttons labeled per the Computer PC logic source profile available on Axia consoles. Pressing the ON button would simulate the ON state being reported by the console surface for the device (GPO PIN 1 Low state) and if the API profile defines a command for this state, you would see the computer program respond accordingly. A possible use of this would be a player application which uses the “P” key for start of play could be triggered.

Profile.GPIOVK

The format of the file is ASCII text. If sending Windows™ messages to the computer application, the profile must define the Window Class of the application.

```
WINCLASS={application class}
```

The remainder of the profile defines device events. The format is:

```
{device}. {pin} {message}
```

A device is defined with DEV followed by its numerical value:

DEV0 is device 1 (the first source stream – GPIO association)

The pin value should be a value between 0 and 4 and refer to the GPO pin sent by the Axia control surface's defined Source Profile.

A poor man's example (that has no useful function) is the following profile which defines the key press of X in Microsoft's Notepad when Pin4 is received on device 1.

```
# Poor man's example with use of Notepad
```

```
WINCLASS=Notepad
```

```
DEV0.3 KEY=X
```

Sending UDP messages is possible by replacing the WINCLASS callout with a UDP_

DSTPORT={port #}. An example of this used by a radio automation system many moons ago is:

```
# Many moons example from an automation system
```

```
UDP_DSTPORT=2002
```

```
DEV0.3 COMMAND="<DADCMD><ID>99999</ID><COMMAND>play pbkl</COM-  
MAND></DADCMD>"
```

GPIO through TCP-IP Control protocol interface

More commonly used interface is through the LWRP (Livewire™ Routing Protocol) which is accessed through port 93 of the NIC hosting the IP-Audio Driver. Once a TCP connection is established, simple text is exchanged to command GPIO state changes or report state changes.

Commands

ADD GPO - Enables indications of GPO pin changes. Those are control messages for the automation 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.

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 will 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.

Appendix C:

Livewire+

The Axia IP-Audio Driver includes support for AES67, Livewire+, with version 2.10. This support adds the 1ms packet time required in AES67. The standard defines what receivers shall be able to receive, but does not dictate how a transmitter should be transmitting. As long as a transmitter is sending packets which fit into the many possible AES67 defined stream it should work, right? (snicker)

So the Axia IP-Audio Driver maintains the typical 2-channel transmit streams which is the most common need by the market. The legacy 8-channel surround mode, which was introduced as part of Livewire in 2003 still remains. Other channel count transmitted streams are not widely needed at the moment, but Telos Alliance is working on changes that would enable clients to use these more obscure transmission combinations.

To be compliant with AES67, a product must be able to receive various streams and the Axia IP-Audio Driver supports the more obscure styles through a back end and not through the configuration UI which is documented in this manual. The Windows configuration UI is provided to support the most common usage of the driver.

To access the backend, the end user would need to setup a TCP connection to the PC on port 93. This can easily be done with a windows telnet session or widely used IT applications like Putty. From a connection, the Destination (what is received from the network) can be configured to accept alternate streams. For example, to receive a unicast stream into the IP driver through the first Destination:

```
DST 1 ADDR:"sip:6@192.168.101.73"
```

Or a multicast address outside of the standard channel range used by the Livewire network:

```
DST 1 ADDR:"239.190.1.1"
```

The syntax is multicast/unicast value;fmt=A/B/C

where

A is L16 or L24 : either a 16bit or 24 bit depth stream

B is 44100 or 48000 : either a 44.1khz or 48khz sample stream

C is 1 - 8 : the number of channels in the stream. For example a stereo stream would be 2.

For example, a 16-bit, 48khz, 8 channel stream at multicast group 239.191.10.10 would be written as follows:

```
239.192.10.10;fmt=L16/48000/8
```

Defaults are L24, 48K, 2

