The networked studio has arrived.

Professional Networked Audio

AXIA
A TELOS COMPANY

Ethernet for studio audio systems.

Broadcast audio studios have really changed in the last decade. PC editors and delivery systems have made tape obsolete, even replacing CD's and other digital media in many facilities. Increasingly, mixing, routing, and processing devices are digital too.

But there's still cost and confusion involved in connecting all the pieces of a broadcast studio. We have to deal with analog (professional and consumer), digital (AES3 and MADI), and audio file transfer over data networks. And all the different connectors: XLR, RCA, DB-9, DB-15, 1/4" and 1/8" phone (stereo and balanced versions), RJ-45, fiber, copper... there's got to be a better way.

Ethernet is already the most common digital audio transmission method in radio facilities today, connecting audio delivery servers with studio computers. So why not use Ethernet as a low-cost, universal way to connect audio and data for everything – including real-time audio – in our studio facilities?

Introducing Axia, a completely new approach to studio audio. Using patented Livewire™ technology, Axia networks employ a framework of standard Ethernet hardware to transport high-performance audio throughout your entire facility.

Axia architecture is lightning-fast. Total input-to-output latency is less than 1ms. per network hop, enabling transmission of live audio without discernible delay. PC-based audio applications can send and receive digital audio using Ethernet — no sound cards needed.

The Axia network can carry all your other routine network traffic as well, like RDS information, messaging and file transfers, at the same time as your audio. By exploiting the capabilities of modern Ethernet switching hubs, audio takes the highest priority and never misses a beat.

And Axia's modular approach gives you significant advantages over traditional wiring and routing systems. Not only is installation time slashed from weeks to days, the small amount of cable makes it easy and cost-efficient to move your Axia system to a new location should the need arise.

Welcome to a new way of thinking about studio audio systems. Welcome to the networked broadcast facility of the future. Welcome to Axia.

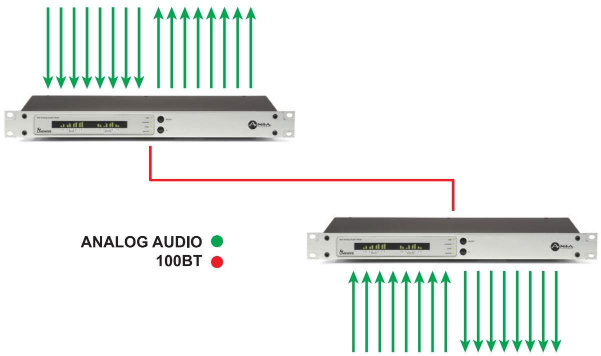
Simple, elegant and low cost.

Replace all that discrete wiring with simple, flexible and standard Ethernet.

What can you do with Axia?

You can connect virtually any audio devices! This unparalleled flexibility is made possible because Axia networks are constructed with building blocks we call “Audio Nodes.” As a result, Axia networks are easily scalable, so you can fabricate an audio network as large (or small) as you wish. Send just a few audio signals from one room to another — or construct a full-featured high-capacity routing switcher that serves your entire campus. For instance:

Make a snake.
Need to get a group of signals from one room to another without the cost and hassle of running discrete wires? One Ethernet link can carry it all!



Building a snake with Axia is easy: just plug audio sources into Axia Audio Nodes, and connect them with a single Ethernet cable — no cable trays or expensive, bulky multi-pair bundles needed.

And unlike hardwired cables, Ethernet’s inherent modularity lets you “grow as you go.” When you run out of pairs on a regular snake, you’re out of luck — but with Axia, channel capacity is nearly unlimited.

Send audio across the hall... or across town.
Today’s consolidated media facilities sometimes occupy multiple buildings, making audio transport even more challenging. Even if you can squeeze a 100-conductor cable into the conduit, there’s the possibility of performance degradation due to induced noise, line loss, *et cetera*.

With Axia, there’s an easy solution. Just add fiber optic media converters and cable to connect Axia audio nodes; now your audio can travel many kilometers — with absolutely none of the line loss inherent in analog methods. Channel capacity increases, too: you can send hundreds of channels of stereo audio using a single optical fiber link.

Snapshot: Husker Sports Network

“Husker Sports Network feeds University of Nebraska games to affiliates nationwide — baseball, basketball, and of course, Huskers football.

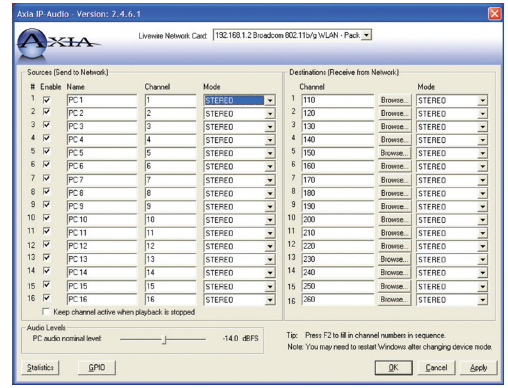
“We have a large system that covers the entire University campus. Many times, we’re live from several different locations at once, and we need to quickly change and re-route signals at a moment’s notice.

“We looked at a number of systems, but I didn’t want a central core-type system that must process all audio, as this would involve running additional wiring extended distances. With Axia, we could simply place Audio Nodes throughout the University’s athletic complex and Memorial Stadium and transfer all the audio via CAT-6.

“With our Axia system, we can quickly change the routing, enabling all of the talent to hear one another, hear the right mix minus, be able to take phone calls, and have the correct mixes to callers. The ease of use is terrific, and Axia’s PathfinderPC software makes it very quick and easy to recall routing configurations, or program the system to switch settings automatically.

“Axia’s networked audio system solves a lot of common audio headaches. There’s no doubt we’ll continue to grow our Axia system.”

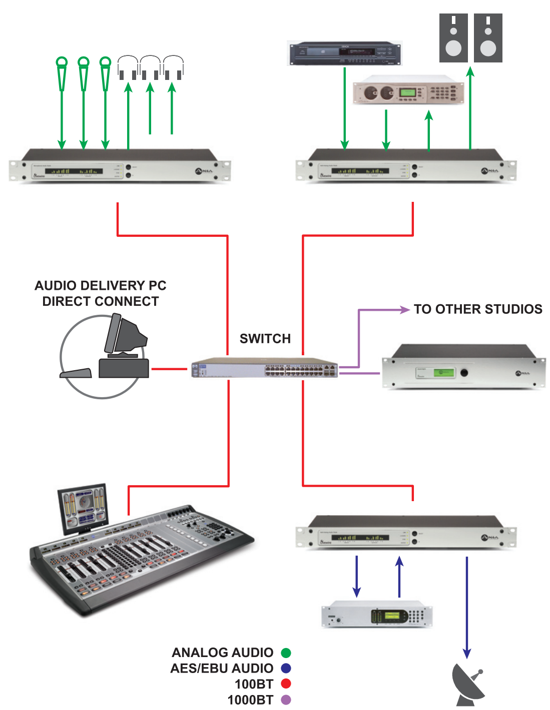
— Mike Elliott, General Manager



Get premium quality digital audio out of your PC — without sound cards!

Even the best sound cards can be compromised by PC noise, inconvenient output connectors, poor headroom, and other gremlins. With Axia, you can eliminate sound cards by connecting your PC directly to the Axia audio network using ordinary Ethernet ports. Your PC audio stays pristine, and not only do you eliminate the expense of sound cards, but also the audio inputs (router or console) normally fed by the sound cards. For highly-computerized environments, the cost savings can be significant.

Using a special Windows driver, your PC becomes an Axia "node" and its Ethernet port connects directly to the Livewire network, and the Axia driver handles all the necessary audio conversions. The PC can then send its audio to any network destination, and can record or audition any other network source as well.



Share audio devices between two studios... or twenty.

Axia networked audio devices allow you to assemble many different applications. The illustration at left shows how the various equipment in a typical studio might interconnect. From a simple air studio to a huge clustered-station facility, the network scales logically and-cost effectively to meet any need.

For instance, we told you how a few audio nodes can be used to move audio signals over a CAT-6 cable between studios. Attach a few more nodes and an Ethernet switch, and you have a distributed multi-room routing switcher. Plug in a control surface and a mix engine and you have a powerful networked broadcast console. Now, plug your delivery system PC into that same Ethernet switch and you can transfer files, live audio, and associated data all over that same net. Think of the time and money you'll save constructing a network operations center or consolidated broadcast facility.

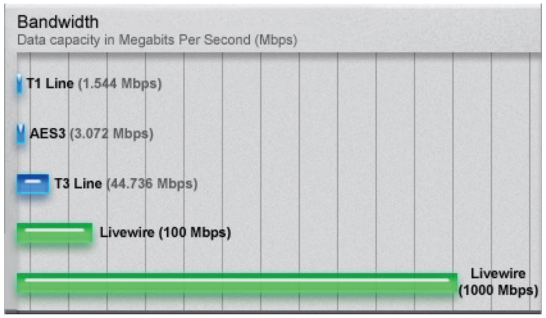
In fact, an Axia network completely replaces the need for a cross-point audio switcher. Every source is available to every destination on the network. And because the network is scalable, it can cost-effectively cross-connect a few studios, or a few dozen. A single Ethernet switch will support hundreds of crosspoints; more switches can be cascaded for nearly unlimited capacity.

So how does it work?

All Axia products are based on **Livewire**, a pioneering technology invented by Telos Systems to convey low-delay, high-reliability audio over switched Ethernet.

With Livewire, a single CAT-6 Ethernet cable or fiber link carries multiple channels of real-time, uncompressed digital audio, device control messages, program associated data, routine network traffic, even VoIP telephone data. An entire facility can be wired in hours, instead of weeks.

Expanding or modifying your system is simple thanks to Axia's inherent scalability and modularity. It can be used to connect a pair of devices, or as a sophisticated infrastructure for an audio plant with thousands of signals.



This is made possible by Ethernet's enormous data capacity. A 100Base-T segment can carry 25 stereo channels of 48 kHz, 24-bit linear PCM audio in both directions; a 1000Base-T link or Gigabit fiber can carry ten times that amount, with *tens of thousands of **stereo channels*** per system! All Axia networks are based on switching Ethernet hubs to guarantee audio Quality of Service (QoS); audio is prioritized and takes precedence over all other data types.

And Livewire features extremely low latency, enabling real-time monitoring of live audio sources. Per-link delay is less than 1ms. for high-priority live audio signals.

Here's the best part: Axia saves you money. Taking advantage of computer industry scale, Ethernet networked audio provides flexibility at a surprisingly low cost. How low? Our hardware typically costs less than half the cost of those big mainframe routers. That's right... ***half***. Axia saves money by eliminating distribution amps, line selectors, sound cards, patch bays, multi-pair cables, and tons of discrete wiring — not to mention the installation and maintenance time you'll recover. Once you experience the benefits of networked audio, you will never go back.

The switch makes it possible.

Switched Ethernet is an entirely different architecture than previous network approaches. Unlike a standard hub, a switch doesn't flood all packets to all ports — it only sends appropriate packets to specific ports, so that the traffic on each network segment is only that which belongs there. This eliminates congestion and dramatically increases network payload.

The switch also allows prioritization of message traffic. When it encounters messages tagged as high priority, it processes them before any other messages, so that urgent, important traffic (such as live audio) can be sent over the same segments as more routine traffic.

Livewire takes full advantage of the traffic management and prioritization capabilities of modern Ethernet switches. The switch delivers Livewire packets only to the target ports, and prioritizes the audio packets so they take precedence over all other traffic (such as when a PC, using a single Ethernet port, is sending both live audio and generic data).

With all these benefits, it's easy to imagine future broadcast gear providing Ethernet jacks instead of (or in addition to) analog and AES3 connections.

Meanwhile, we're going to need a way to interface conventional audio equipment with the Livewire network. That's where the Axia family of audio adapter nodes comes in. There are five versions:

- an Analog Node,
- an AES/EBU node,
- a Microphone node,
- a Router Selector node,
- and a GPIO node.

Each of these 1RU units is equipped with a 100Base-T Ethernet connection. When a node is connected to the Livewire network, advertises that its audio sources are available for use, allowing any users access to them. Installing Axia audio nodes is easy — just place them near your audio source and destination devices, and distribute them throughout your facility wherever it's convenient.

For example, a microphone node placed in a studio can collect audio from microphones and also provide outputs to associated studio monitors and headphones. Another node in the central equipment room can collect audio from network feeds, codecs and other shared sources for system-wide use while providing convenient outputs for audio processors and other terminal-room gear.

The microphone, line and AES nodes feature a multi-character LED front panel with confidence metering that shows the audio activity on each of the inputs and outputs, and can also display text labels to ease configuration and identification of audio sources during installation.

To ensure ultra-reliable network operations and extremely low delay, Axia audio nodes run a version of Linux on an embedded processor, and a built-in web server in each node gives you remote configuration and control — in an intuitive easy-to-understand manner — using any standard Web browser interface.

Snapshot: KWMU-FM

"KWMU is an NPR affiliate located on the campus at the University of Missouri St. Louis. Our facility consists of four studios. In Air Control, we have a 20-position Element control surface with 12 faders; Production A has 16 faders in a 24-position frame, and both Production B and our news booth utilize 12-position Elements with 4 faders each.

"Axia represents great value. Constructing the entire system with standard Ethernet cable is a huge cost savings compared to conventional wiring. Ethernet is already the most common method of transferring digital audio in a broadcast environment, so it makes perfect sense to run the entire system this way. We're not yet broadcasting in 5.1 surround, but we're looking forward to it, and it's really good to know that our Elements can go from stereo to surround with a single command. This translates to a huge cost savings because there's no additional console/router hardware required.

"The on-air talent has been extremely receptive to the new system. It's been a big change going from an analog board to a digital setup like the Element, but the system's ability to provide custom configurations has made everyone very eager to learn more."

— Terence Dupuis, Chief of Broadcast Operations

The Microphone node has eight professional-grade microphone preamps with selectable Phantom power and software-adjustable gain. There are also eight balanced analog line outputs to conveniently deliver headphone and studio monitor feeds back to the talent. Inputs use XLR connectors; outputs are on easy-to-install RJ-45's.

Microphone Node, 8 preamp inputs and 8 stereo outs.

The Analog Line Node has eight balanced stereo inputs and eight balanced stereo outputs, all on RJ-45 connectors. Each input is switchable to accommodate either consumer-level -10dBv or professional level +4dBu. The short-circuit protected outputs can deliver up to +24dBu before clipping. We make use of the very best quality A/D/A converters and low-noise components, so that each Analog node provides superior audio performance for high-end studio use.

Analog Line Node, 8 stereo ins and 8 stereo outs.

The AES/EBU node provides eight stereo AES3 inputs and eight AES3 outputs. Sample-rate conversion is available on all inputs; the unit can also sync to a house clock. Like the mic and analog line nodes, the AES node displays confidence metering for each of its inputs and outputs on the front panel, which doubles as a system configuration display.

AES/EBU Node, 8 stereo AES3 inputs and 8 AES3 outs.

The Router Selector node resembles the X-Y controllers used with expensive cross-point audio switchers. The LCD screen lists available sources, which can be browsed and selected with the scroll wheel; eight "radio buttons" provide instant access for your frequently-used sources. Unlike an X-Y controller, however, the Router Selector node has audio output direct to headphones, and analog and AES3 outputs. It even provides a convenient analog and AES3 input, making it ideal for production or news studios where operators typically both create and play audio streams.

Router Selector Node, 1 input 1 output, mixed signal.

Finally, the GPIO node provides 8 logic ports for machine control, each with 5 opto-isolated inputs and 5 isolated outputs. A logic port can be associated with any input or output and routed along with the audio.

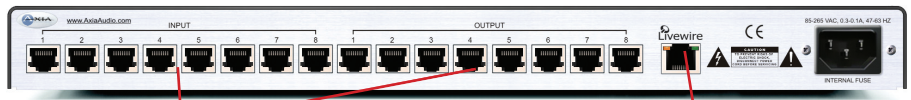
GPIO Node, 8 logic ports, each with 5 ins and 5 outs.

A look behind the scenes

Auto-switching internal power supply on all Nodes.

The Microphone node's 8 balanced XLR connectors feed eight studio-quality mic preamps and 24-bit A/D convertors.

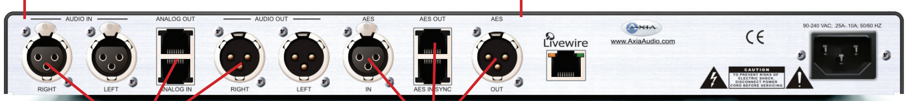
Outputs use RJ45s for quick installation. Use StudioHub+ pre-wired connectors, or crimp your own.



RJ45 connectors on Analog and AES/EBU line nodes make installation fast and easy.

A single CAT-6 cable from this Livewire port to the Ethernet switch instantly connects your sources to the network.

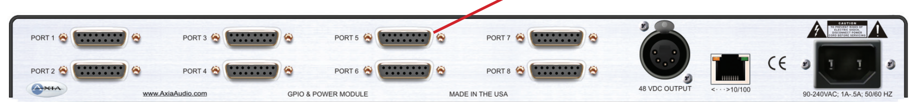
The Router Selector node has something no other router controller has — audio inputs and outputs, both analog and digital. Perfect for news or production rooms.



Send and receive analog audio directly using either balanced XLRs or RJ45s.

AES/EBU I/O gives you your choice of XLR or RJ45 as well. "AES In" RJ port doubles as a sync input for your house clock.

The GPIO node provides eight opto-isolated general-purpose I/O ports (each with five inputs and five outputs) for machine control of your playout system, CD players... whatever. Connections are on common DB-15 connectors.



If you're using an Axia Element for studio control, a special version of the GPIO node contains a high-current power supply for the console as well..

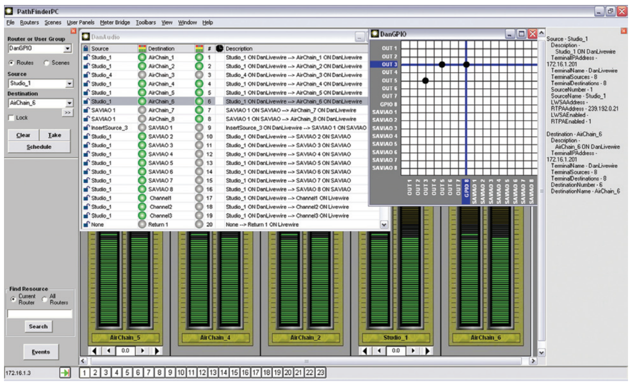
Wiring made fast and easy.

You've probably noticed our liberal use of RJ45s for audio I/O on Axia nodes. Not only is the RJ45 the perfect connector for data networks, it turns out to be an excellent choice for line-level analog and AES/EBU digital signals as well. That's why we've partnered with Radio Systems, makers of the popular StudioHub+ pre-wired modular connection system; Axia nodes are pin-compatible with StudioHub+ components to help reduce the time, frustration and expense associated with on-site wiring. Conversion cables from traditional audio connectors to RJ45 are available as accessories.

Matrix management: PathfinderPC

If you're constructing an Axia network to serve multiple studios, chances are you'll want to use Livewire's sophisticated cross-point routing capabilities. For this, we offer a powerful software program to design and administer your cross-point network: PathfinderPC.

PathfinderPC automatically scans your audio network for Axia audio nodes, and gathers information about the audio sources and destinations, then presents this information to you in an easy-to-use interface. Configuration is accomplished using helpful, intuitive "wizards"; make a route by simply clicking on the destination you want to route to, and then selecting the source. These "route points" can then be locked or unlocked to prevent other users of the system from inadvertently changing a route that's in use.



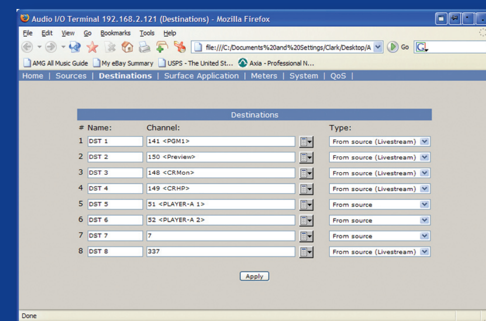
You can quickly create and recall Snapshots (or scenes) to make multiple routes with a single mouse click. These snapshots can make changes to *many* routes – for example, to return the entire system (all audio nodes) to a default routing setup – or it can make changes to only *a few* routes, such as routing audio from one studio device to another.

But that's not all. PathfinderPC can combine audio and machine logic into a single "virtual router" that allows you to route bi-directional audio and GPIO simultaneously. Built-in audio metering (something not found in most cross-point routers), combined with a configurable silence sense, lets you set up automated "watchdogs" on important audio sources, automatically switching to a backup source if audio is not present. Imagine: an audio system which can "heal itself" from a dead-air condition!

And PathfinderPC even gives you remote access to your routing system, via its integrated, platform-independent web server. Log in securely using any standard, Java-compliant browser on a Windows, Mac or Linux-based PC for instant off-site control of your entire networked facility.

Browser-based configuration.

Because Axia is a true audio network, you have the ability to configure and control its components using any PC with a standard Web browser. Simply enter the IP address of any audio node into your browser, and you'll see configuration screens similar to the one shown below. (And yes; all Nodes' configuration screens are password protected to prevent tampering.)



From here, you can name and configure input sources and output destinations, associate GPIO closures with audio channels, set input gain, and determine a host of other options.

Just think of the time and energy you'll save with the ability to configure your entire network from a single location.

There are plenty of ways to control your Axia network. Built-in web servers on all Axia equipment allow easy configuration using a standard Web browser. PathfinderPC® software for Windows allows central administration of every audio path in your plant. Router Selector nodes provide quick local selection of audio sources.

But what about the mix? Certainly, you can mix audio from an Axia network using a conventional analog or digital console. Or, you can install an intelligent Axia control surface (like the Element console pictured above) for tighter integration with the Axia network and powerful new features for your operators.

For instance, talent can instantly access and mix any source in your facility. You can save configuration snapshots (we call them profiles) for each user, allowing them to instantly reconfigure their console just they way they like it. Imagine: each user can have a custom surface configuration that loads voice processing, monitor settings, sources assignments and more with just one button press!

One of its most popular features is the way Element handles mix-minus for phones and codecs — **every channel** has the ability to provide a mix-minus output automatically, without any intervention from talent. Operators simply select a phone or codec for on-air use and the backfeed is automatically generated, eliminating a source of confusion and error that's troubled board ops for decades.

In addition to traditional console functions, Element provides controls and displays to interact with phone systems, codecs, editors, PC playout systems, etc. In short, Element was designed to be the key control interface for radio operators. And a wide variety of modules and frame sizes let you design Element consoles that are perfect for your needs.

Even though no audio passes through this surface, it can access and control any audio source or destination in your networked plant. This sophisticated "remote control" surface is always associated with a networked DSP device we call the Axia StudioEngine.

The Mixing StudioEngine.

Because the Axia system is a network at heart, all connected sources and destinations are accessible from anywhere in the networked facility. This makes cross-point switching possible from any source to any destination.

This deceptively simple networked architecture opens the door to a new way of mixing and processing audio, using a DSP engine that accesses audio streams, modifies them, and then presents the resulting streams back to the network as program output (or monitor output, or mix-minus output, *et cetera*).

The Axia StudioEngine is an extremely powerful mixing and processing device, based on a blazingly-fast Intel processor that can out-perform even the largest dedicated-DSP embedded designs. This approach is ideally suited to a network-based audio architecture since all input and output streams are routed through a Gigabit Ethernet port.

To deliver the reliability and ultra-low latency required, we powered the StudioEngine with a fast, robust version of the Linux real-time operating system. Then we optimized our engine processing program so that total input to output latency is just a few hundred microseconds. In fact, each StudioEngine has so much CPU power, it can outperform the very largest digital or router-based consoles, with multiple simultaneous inputs, outputs, mix-minus feeds, monitor signals, etc. It can even provide EQ for multiple channels, and has the headroom to support future features. One StudioEngine is required for every Axia control surface.

Along with all that power and flexibility, the Axia Element and StudioEngine also help you reduce studio cost and complexity. You can network all of your facilities' audio peripherals for greater efficiency. And you can finally enable communication between computer-based studio applications... playout and traffic systems, logging applications and much more.

Snapshot: Minnesota Public Radio

"Late in 2003, [we] began planning our new technical infrastructure with a thorough examination of the distributed routing/control surface model. Our goal was to determine if it would give us operational advantages. The result of our research was a resounding 'yes'...We chose the Axia IP-Audio system and Element modular control surfaces.

"Because of the large amount of content MPR produces, we wanted studios that could accommodate any show at a moment's notice. Since Element functions equally well as an on-air or production console, we sidestepped the need for different boards in different studios. Our operators can do any show in any studio and be instantly familiar with the console's controls and operation.

"Generating mix-minus with Element couldn't be easier...When a codec or a phone hybrid is placed on-air, Element automatically generates a mix-minus and sends it to that backfeed without any operator intervention. Element handles talkback very smoothly as well.

"Element is a very well-thought-out and well-executed control surface."

— Ethan Torrey, Chief of Research & Development

With a little help from our friends.

A networked audio system doesn't just replace a traditional console and router — it improves upon them, by providing complete integration with PC-based audio delivery systems. Leading companies in our industry have realized the advantages of tightly integrated systems, and are making new products that reap those benefits.

Check AxiaAudio.com/partners/ to find out who else is partnering with Axia. Don't see your system listed? Ask your favorite supplier about becoming an Axia partner.

Microphone Preamplifiers

- Source Impedance: 150 ohms
- Input Impedance: 4 k ohms minimum, balanced
- Nominal Level Range: Adjustable, -75 dBu to -20 dBu
- Input Headroom: >20 dB above nominal input
- Phantom power: +48VDC, switchable

Analog Line Inputs

- Input Impedance: >40 k ohms, balanced
- Nominal Input Range: Selectable, +4 dBu or -10dBv
- Input Headroom: 20 dB above nominal input

Analog Line Outputs

- Output Source Impedance: <50 ohms balanced
- Output Load Impedance: 600 ohms, minimum
- Nominal Output Level: +4 dBu
- Maximum Output Level: +24 dBu

Digital Audio Inputs and Outputs

- Reference Level: +4 dBu (-20 dB FSD)
- Impedance: 110 Ohm, balanced (XLR)
- Signal Format: AES3 (AES/EBU)
- AES3 Input Compliance: 24-bit with selectable sample rate conversion, 32 kHz to 96 kHz input sample rate capable.
- AES3 Output Compliance: 24-bit
- Digital Reference: Internal (network timebase) or external reference 48 kHz, +/- 2 ppm
- Internal Sampling Rate: 48 kHz
- Output Sample Rate: 44.1 kHz or 48 kHz
- A/D Conversions: 24-bit, Delta-Sigma, 256x oversampling
- D/A Conversions: 24-bit, Delta-Sigma, 256x oversampling

Frequency Response

- Any input to any output: +/- 0.5 dB, 20 Hz to 20 kHz

Latency

- Analog Input to Analog Output, 2.75ms including network, converters, and mixing process
- Digital Input to Digital Output, 1.75ms including network mixing engine (ASRC off)

Dynamic Range

- Analog Input to Analog Output: 102 dB referenced to 0 dBFS, 105 dB “A” weighted to 0 dBFS
- Analog Input to Digital Output: 105 dB referenced to 0 dBFS
- Digital Input to Analog Output: 103 dB referenced to 0 dBFS, 106 dB “A” weighted
- Digital Input to Digital Output: 138 dB

Equivalent Input Noise

- Microphone Preamp: -128 dBu, 150 ohm source, reference -50 dBu input level

Total Harmonic Distortion + Noise

- Mic Pre Input to Analog Line Output: <0.005%, 1 kHz, -38 dBu input, +18 dBu output
- Analog Input to Analog Output: <0.008%, 1 kHz, +18 dBu input, +18 dBu output
- Digital Input to Digital Output: <0.0003%, 1 kHz, -20 dBFS
- Digital Input to Analog Output: <0.005%, 1 kHz, -6 dBFS input, +18 dBu output

Crosstalk Isolation and Stereo Separation and CMRR

- Analog Line channel to channel isolation: 90 dB isolation minimum, 20 Hz to 20 kHz
- Microphone channel to channel isolation: 80 dB isolation minimum, 20 Hz to 20 kHz
- Analog Line Stereo separation: 85 dB isolation minimum, 20 Hz to 20 kHz
- Analog Line Input CMRR: >60 dB, 20 Hz to 20 kHz
- Microphone Input CMRR: >55 dB, 20 Hz to 20 kHz

Power Supply AC Input

- Auto-sensing supply, 90VAC to 240VAC, 50 Hz to 60 Hz, IEC receptacle, internal fuse
- Power consumption: 35 Watts

Operating Temperatures

- -10 degree C to +50 degree C, <90% humidity, no condensation

Dimensions and Weight

- Microphone node: 1.75 inches x 17 inches x 10 inches, 6 pounds
- Analog Line node: 1.75 inches x 17 inches x 10 inches, 6 pounds
- AES/EBU node: 1.75 inches x 17 inches x 10 inches, 6 pounds
- Router Selector node: 1.75 inches x 17 inches x 10 inches, 6 pounds
- GPIO node: 1.75 inches x 17 inches x 13 inches, 8 pounds
- Studio Mix Engine 3.5 inches x 17 inches x 15 inches, 10 pounds

Snapshot: WOR

“Buckley Radio decided to move WOR to a new location, leaving behind studios we’d called home for over 50 years.

“WOR needed a state-of-the-art facility that was digitally based. I looked at the systems available and settled on Axia. I’d heard that the Axia IP-Audio system could give us the high-end features we needed. The more I learned about Axia, the more impressed I became with their routing switcher and consoles, and how well their network topology was designed.

“So I decided to break new ground and order the Axia consoles and routing setup, nine studios worth. It’s been on the air for over a year now, and we love it.

“The decision to install Axia has proven to be a good one. The system worked out of the box. Installation time was cut way down. Connections are simple. The system, coupled with PathfinderPC routing software, is powerful. All the data switches used are top-line off-the-shelf items. Our operators keep raving about how easy things are to operate. Even our listeners tell us how good WOR sounds!”

— Thomas R. Ray III,
Vice President/Corporate Director of Engineering

How exactly does Axia differ from other router-based studio systems?

One big difference is the cost! All of those router-based systems share a similar architecture with everything built around a shared centralized switcher frame. Not only are these frames expensive, but studio-located sources and destinations require long audio cables or more expensive local hardware.

Instead, Axia systems use high-reliability switched Ethernet (using patented Livewire networking technology) to eliminate the need for those expensive proprietary TDM main frames, DSP farms and local acquisition frames. Also, Axia eliminates the need for PC sound cards. Not only does this save the cost of the sound cards, it also eliminates corresponding console or router input cards. And Axia eliminates miles of discrete wiring and labor used to install standalone routers, instead using CAT-6 to transport dozens of digital stereo channels on a single cable.

By using standard computer-industry devices Axia eliminate purpose-built hardware, which translates into dramatically reduced system cost. Most Axia clients save at least 50% when compared to any other digital router-based approach.

What about delay?

For live monitoring, such as when an air talent hears his own microphone in headphones, 10-15ms is the limit before noticeable problems. We've kept Livewire link delay to below 1ms, so a number of links can be successfully cascaded. To put this in perspective, a normal professional A-to-D or D-to-A converter has about 0.75ms delay.

How can you promise live audio over Ethernet? Won't it drop out?

No. We wouldn't be proposing any system that wasn't full broadcast quality. With Ethernet switching, each device owns all of the bandwidth on a link so there is no possibility of contention or audio loss. If a node needs both audio and data, such as a PC running an audio editor and a web browser, audio is prioritized and always takes precedence. Livewire has been proven rock-solid under extreme laboratory and real-world broadcast conditions for hundreds of hours while carefully logging packet transmission.

But the Internet is a packet network with poor quality for audio.

Right. Internet bandwidth is not guaranteed, so there can be problems when there is not enough. But you completely own and control all the pieces of a Livewire system, so performance is fully reliable.

Are you sure this is robust enough for 24/7 operation? My Windows networks sometimes have downtime.

Livewire equipment is based on tight, embedded hardware and software. The Ethernet switches we recommend are fully professional devices with high reliability.

Can the network be used for general data traffic as well as audio?

Most certainly, should you choose to do so. The Ethernet switch naturally isolates traffic. You may even use one link for both audio and data, since the audio is prioritized. This will probably be the case when a PC is connected to the network — you will sometimes want to download files, receive e-mail, etc., in addition to the audio stuff.

Do you use compression? I am concerned about codec cascading.

Livewire audio is uncompressed 48kHz/24-bit. It would be possible to have compressed streams sharing the Ethernet, but this is not a part of Livewire.

What about connections to the Internet?

Livewire is intended for use within a facility on a switched Ethernet. Normally, a gateway would be required to interface with the Internet or other general-purpose IP network. And since we are using uncompressed audio, the bandwidth requirements are probably too high. The gateway could provide the required compression.

I've got a large facility. How many studios can I interconnect?

There is no practical limit. You may have as many studios and audio channels as your Ethernet switch can support. Switches come in all sizes, some with hundreds of ports. And multiple switches may be cascaded to expand ports. We recommend that you use

(continued)

one switch per studio to isolate any problems to a defined area. These are then interconnected with a backbone. Switches may be physically associated with each studio or may be placed in a central location, as you prefer.

What about for smaller stations? This all sounds pretty sophisticated for a simple set-up.

Look at Ethernet for data applications. You have everything from a few PCs in a small office to huge campus networks with thousands of nodes. This is one of the reasons we went with Ethernet - you can use it for big and small facilities. The technology and economics naturally scale to suit the application size. We figure, in fact, that small stations may benefit the most as they gain routing capability at a very modest cost.

This seems like a lot of IP to keep track of. Are there any administration tools?

All devices have a web browser control and monitoring capability. Keep the IP numbers in a "favorites list" and you can easily check them.

How do analog sources become part of the network?

With Axia audio (adapter) nodes. These come in variants for line and microphone applications. Over time, you can expect that codecs, hybrids, processors, etc., will offer direct Livewire connection ports.

How does your StudioEngine compare to other "engines?"

Most other engines are central to an entire system. Our engine is a "network appliance," only attached to the audio network when mixing or other DSP functionality is required. Because there can be multiple distributed engines on an Axia network, there is no single point of failure for the entire system. A single StudioEngine has all the DSP capability to run the most complex studio console with ample headroom for future features without requiring new hardware.

How do mix-minuses get generated?

This is a software function within studio processing engines. We provide one for each channel automatically without user intervention.

Tell me about your IP-Audio driver for workstations.

It makes the Axia network look like a sound card to a PC Windows application. Many popular audio applications are already compatible, with more on the way.

Are optical audio links supported?

Livewire is fully compatible with 100Base-T and 1000Base-T copper and fiber connection types. We imagine a common configuration to be switches dedicated to studios with 100Base-T copper connecting nodes, computers and surfaces. A fiber backbone connects the switches in order to share audio between the studios. Engines are connected to 1000Base-T ports either in the studio or the terminal room.

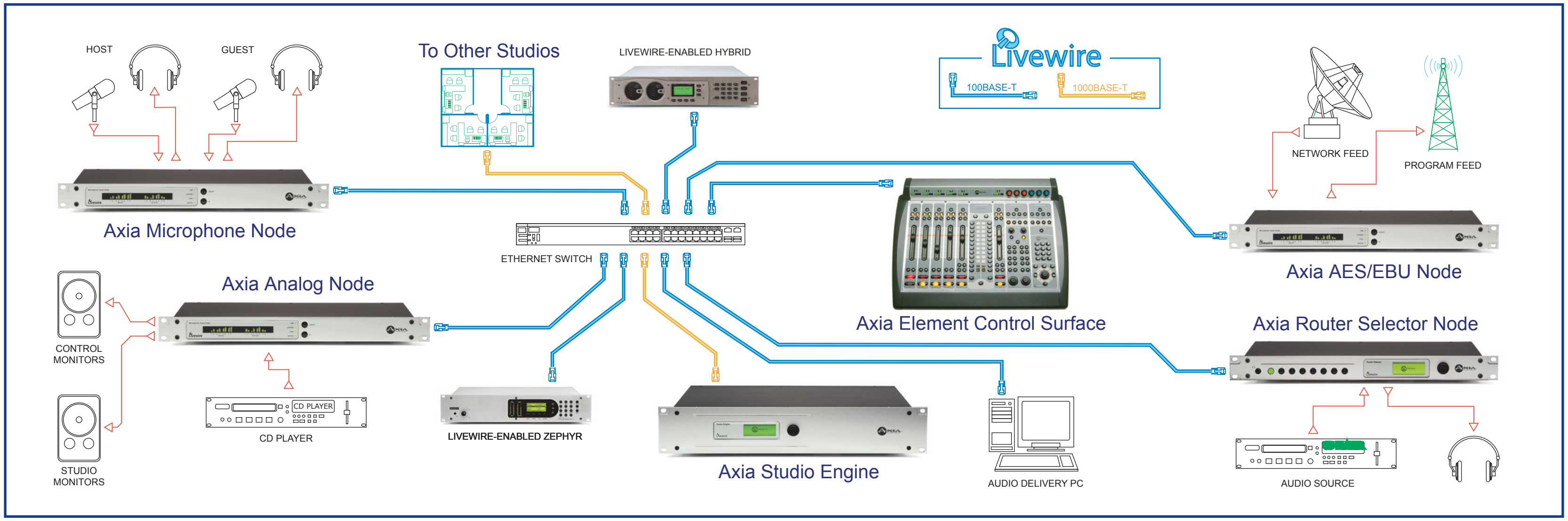
Is Livewire standards-based?

It runs on off-the-shelf Ethernet switches and components, but since there is no standardized way to convey low-delay full-fidelity audio over Ethernet, the audio protocol had to be developed by us. We do support a higher-delay (5ms.) mode for connecting to PCs, which uses the Internet standard for streaming media. (The RTP format defined in the IETF standards document RFC1889.)

Also, we needed to implement a protocol for tagging audio sources with names and advertising these to receivers. Nothing was available off-the-shelf, so we had to invent something. Same for the GPIO-emulation functions.

Are you planning to share information so that other vendors can make gear that plugs directly into Livewire?

Yes. Software vendors for PCs can use our driver to easily make their applications compatible. Makers of audio hardware would have to coordinate with us to be compatible. Of course, you can use whatever equipment you want via the analog and AES Nodes.



Intrigued? Find out more about how it all works at www.AxiaAudio.com/tech/ .

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