The networked studio has arrived.

Professional Networked Audio

A TELOS COMPANY

Ethernet for studio audio systems.

Those of us involved with broadcast audio studios have seen tremendous evolution during the last decade. Reel-to-reels have been pushed aside in favor of PC editors. PC delivery systems have replaced CD players and cart machines, and now we are smack inside the center of the conversion to digital for mixing, routing, and processing.

But we are still using limited, old-fashioned schemes for connecting all of these new pieces together. With technology in flux, we find ourselves lashing-up a mishmash of analog, both professional and consumer; digital, AES3 and MADI, and audio file transfer over data networks. To do this we use XLR, RCA, DB-9, DB-15, 1/4” phone, mini phone, RJ-45, fiber, copper... Our industry is clearly ready for a new way to interconnect studio components.

Ethernet is probably already the most widely used digital audio transmission method in radio facilities today. Computer audio delivery is usually a client-server system with Ethernet connecting the server and the studio computer. Why not take this model one step further? Why not find a way to ensure reliability and Quality of Service that would allow the use of Ethernet as a low-cost, universal way to connect audio and data for everything in our studio facilities?

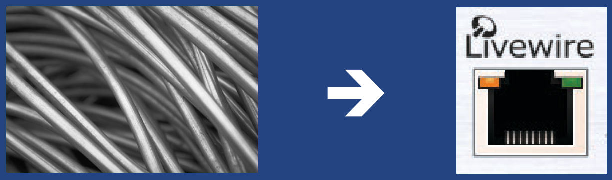
Welcome to the networked broadcast studio. Welcome to the world of Axia. Axia is a completely new approach to studio audio. Using patent pending Livewire™ technology, Axia networks utilize a framework of standard Ethernet hardware to convey high-performance audio throughout an entire facility.

The Axia architecture is lightning-fast, with total input-to-output latency of less than 1ms per network hop, making it possible for operators to easily monitor microphone and other live audio with virtually no discernable delay. PC-based audio applications can send and receive digital audio via Ethernet, eliminating sound cards completely.

The Axia network carries other forms of routine network traffic such as program associated data (PAD) and communications at the same time as the audio. By exploiting the capabilities of modern Ethernet switching hubs, audio data is always first, without ever missing a beat.

We invite you to a new way of thinking about studio audio systems. Networking has changed the computer world forever. The time is right for broadcast equipment to enjoy the many benefits of networking.

Simple, elegant and low cost.



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

- Livewire is the core networking technology employed in all Axia products. Livewire is the first audio network designed specifically for broadcast studios, conveying audio, control and data over standard Ethernet hardware.

- Use ubiquitous and low-cost Ethernet as a universal studio interconnect. A single CAT-6 or fiber reliably conveys multiple audio channels, control, program-associated-data, VoIP phone, and general computer data.

- Multiple studios are easily joined. All audio sources are made available throughout the entire plant. You don't need an expensive cross-point routing switcher to have all the benefits of one.

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

- PCs are directly connected to the network. An IP-Audio driver emulates a sound card, allowing PC audio applications to pass audio directly to and from the network.

- Axia reduces costs by simplifying wiring complexity and replacing sound cards, cross point switchers, patch bays, distribution amplifiers, line selection switches, etc. Taking advantage of computer industry scale, Ethernet networked audio provides flexibility at a surprisingly low cost.

- Systems are based on switching Ethernet hubs to guarantee audio Quality of Service (QoS). Audio is prioritized and takes precedence over all other data types.

- A 100Base-T segment can carry 50+ stereo channels of 48 kHz, 24-bit linear PCM audio; a 1000Base-T link or Gigabit fiber can carry 500+.

- A Studio Engine, based on an Intel P4 running real-time Linux, provides mixing console functions at low cost. All connections into and out of the Studio Engine are via a single Ethernet port.

- Just as with traditional analog, Livewire is naturally scalable. It can be used to connect a pair of devices, or as a sophisticated infrastructure for an audio plant with thousands of signals.

The switch makes it possible.

Switched Ethernet is an entirely different architecture than previous network approaches.

Unlike a standard hub, a switch does not flood all packets to all ports. Rather, it only sends the appropriate packets to specific ports. In this way, only necessary traffic is allowed onto each network segment. This dramatically increases network payload and eliminates congestion.

Further, the switch allows for the prioritization of message traffic. When the switch encounters messages tagged as high priority, it processes them before any other messages. In this way, it is possible for urgent/important traffic such as audio to be conveyed 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 those ports wishing to subscribe, and prioritizes the audio packets so they take precedence over all other traffic.

Axia Audio Nodes

With all these benefits, it's easy to imagine future broadcast gear providing Ethernet jacks instead of— or in addition to— analog and AES-3 connections. Meanwhile, we are going to need a way to convert conventional audio sources and destinations to and from the Livewire network. Meet the Livewire family of audio adapter nodes.

There are now five versions, each with a specific purpose. Each of these 1RU units is equipped with a 100BT Ethernet connection. When a node is connected to the Livewire network, it advertises the availailability and the attributes of its signals to the rest of the network, allowing interested receivers to subscribe to the signals originating from this node.

Nodes can be placed physically near the audio and may be distributed throughout a facility according to convenience. A unit placed within a studio can collect audio from microphones and deliver audio to monitors and headphones, while another in the central equipment area can enter network feeds, codecs, Telco remotes, etc., into the system and provide convenient outputs for processors and other terminal-room gear.

Each node contains a built-in auto-sensing, fanless power supply, and the highest quality components to deliver superior audio performance with the reliability broadcasters demand.

The microphone, line and AES nodes, each offers a multi-character LED front panel that can display text labels to make configuration and identification of devices on a network easy. In its normal operating mode, the LED's display confidence metering to show the audio activity on each of the inputs and outputs.

The nodes run a version of Linux on an embedded processsor for ultra-reliable network operations and extremely low delay. A built-in webserver enables remote configuration and control— all in an intuitive plain language manner— via standard browser interface.

Utilizing the very best quality A/D/A converters, low-noise components, each of the analog nodes offers superior audio specifications for high-end studio use.

The Microphone Node has eight professional-grade microphone preamplifiers with phantom power and software-adjustable gain. It also contains eight balanced analog line outputs for convenient delivery of associated headphone and studio monitor feeds to the room in which the mics are located. The inputs are on XLR connectors while the 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. The inputs are switchable to accommodate consumer level -10dBv or professional level +4dBu. The short-circuit protected outputs are capable of delivering up to +24dBu before clipping.

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

The AES/EBU Node provides eight stereo digital AES-3 inputs and eight AES-3 outputs. Sample-rate conversion is available on all of the inputs or the unit can sync to a house clock. Like the mic and analog line nodes, this device displays confidence metering for each of its inputs and outputs on the front panel which also provides an easy window to identify and configure the system.

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

The Router Selector Node resembles the X-Y controllers used with those expensive cross-point audio switchers. The LCD lists available sources which can be scrolled and selected with the wheel. Eight radio buttons provide instant access to programmed sources. Unlike the aforementioned controllers, the Router Selector Node also has audio output direct to headphones and analog and AES-3 outputs. It even provides a convenient analog and AES-3 input, making this an ideal device for a production or news studio desiring to consume and create network streams.

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

A GPIO Node offers 8 logic ports, 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 for various control interface functions.

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

All sources and destinations are accessible by the Livewire network, making it possible to facilitate cross-point switching from any source to any destination.

This deceptively simple architecture opens the door to new ways of mixing and processing audio signals. Specifically, we are now able to build a mixing/processing engine that subscribes to networked audio streams, modifies them and presents the resulting streams back to the network for interested destinations.

The Axia StudioEngine is an extremely powerful mixing and processing device, based on an Intel P4 processor. This is a blazingly-fast and powerful processor, capable of surpassing even the largest dedicated-DSP embedded designs. Further, the hardware is ideally suited for a network-based architecture since all input and output streams are routed through a single Ethernet connection.

In order to deliver the reliability and ultra-low latency required by the Axia network, we built our Studio Engine around a very fast and 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.

Generally, one StudioEngine is required for each radio studio. Each engine is actually capable of outperforming the very largest traditional consoles, with multiple simultaneous inputs, outputs, mix-minus feeds, monitor signals, etc. Additionally, the Studio Engine can provide EQ for multiple channels and has the headroom to support future features.

The front panel display on the StudioEngine provides confidence feedback on system operations at a glance, while the selector knob makes it possible to set the engine IP address right from the front.

Studio Control Surface

The StudioEngine performs all the mixing and signal processing functions that would have been performed in the past by an audio console. Therefore, the traditional console is not needed in an Axia networked audio system.

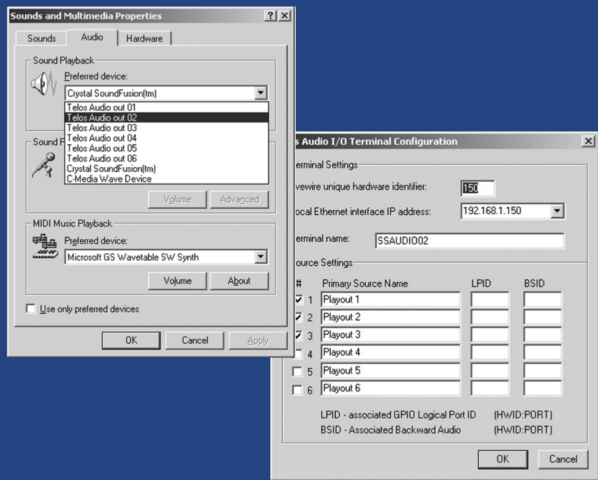
However, operators still need to have control interfaces so they can interact with the studio elements. The Axia SmartSurface shown above is a Livewire-compatible control surface. Designed for the needs of fast-paced live programming, SmartSurface provides users with the functions they need in an uncluttered and intuitive format.

SmartSurface can save unique configurations (we call them profiles) for each user, allowing different preferences, layouts and defaults for a variety of shows and talent. Loading a new profile can completely reconfigure a studio setup instantly, allowing seamless transitions from show to show.

One of the most popular features is the way SmartSurface handles mix-minus for phones and codecs. Every channel has the ability to provide a mix-minus output automatically, without any intervention from the operator. Operators simply select a phone or codec source and the backfeed is automatically generated based on preferences established when the engineer configured the user profile. Finally, we can eliminate a source of confusion and error for many board operators.

In addition to console functions, SmartSurface provides controls and displays that interact with phone systems, codecs, editors, PC playout systems, etc. In short, SmartSurface was designed to be the key control interface for radio operators. A detailed SmartSurface brochure is available at www.AxiaAudio.com.

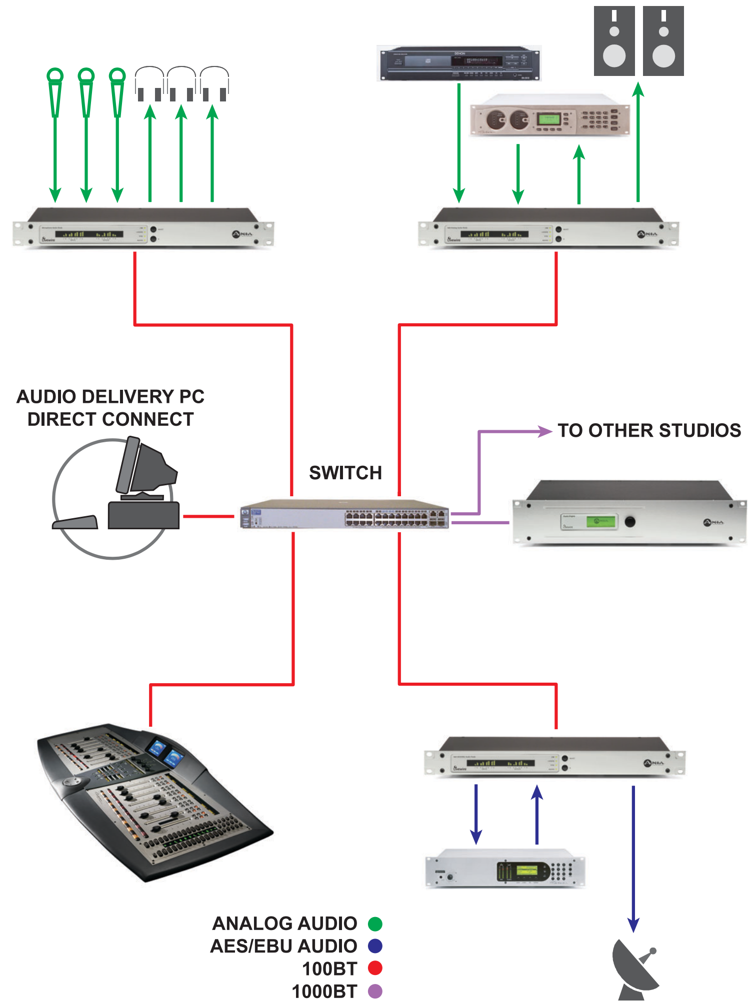
Get audio out of your PC. Really.



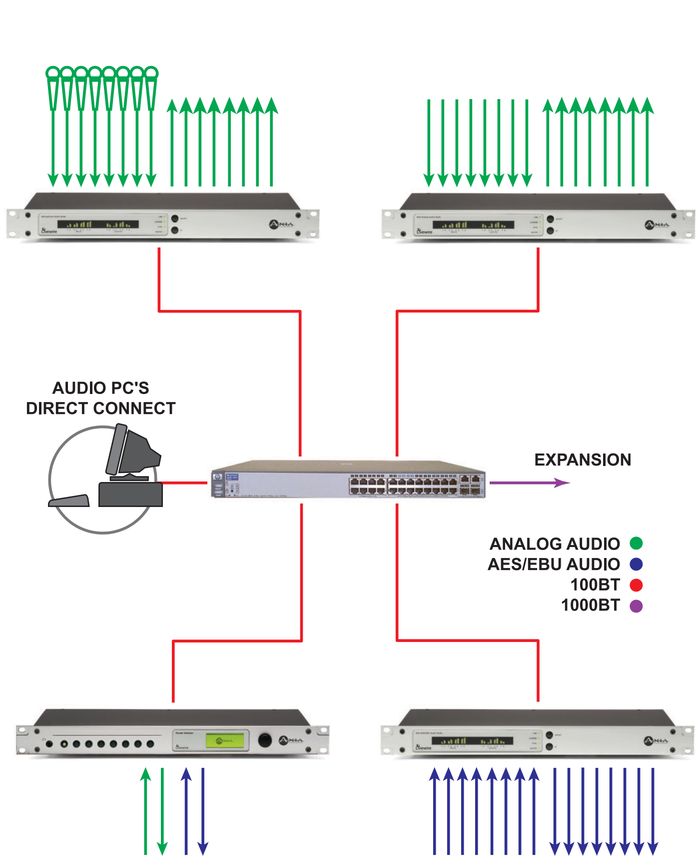
The Axia IP-Audio driver causes the network to "appear" like a sound card to connected Windows PC's, facilitating easy interchange of audio to and from delivery systems and audio editors via Ethernet.

Multiple channels are supported so that, for example, delivery systems are able to send an independent output from each 'player' to a channel on the network and control surfaces can have a fader for each.

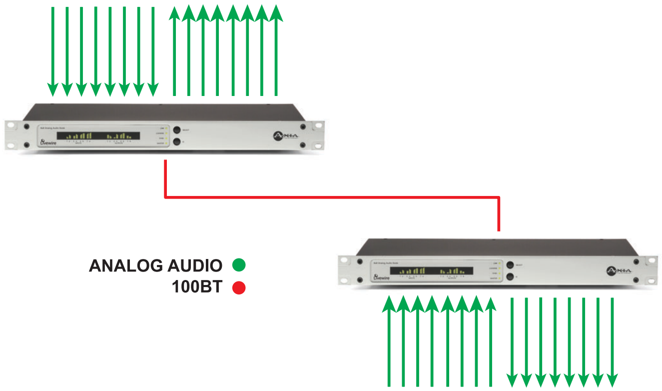
The IP-Audio driver also provides a simple API (Applications Programming Interface) that delivery software developers can use to access the stop-start functions from control surfaces that have traditionally been provided by GPIO connections.



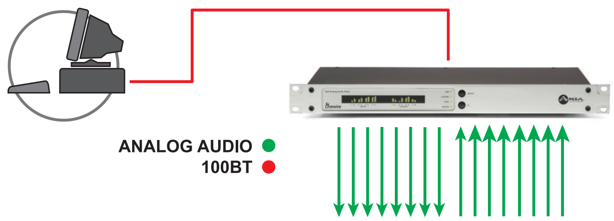
Here we see the networked studio taking shape. PC audio sources are fed to the network. Audio terminals in the studio capture live sources and feed monitors and headphones. Shared sources are in the terminal room as is the program feed. The StudioEngine provides the mixing and processing for the SmartSurface.



The 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 naturally scalable, it can cost-effectively cross-couple a few studios... or a few dozen. A single switch will support hundreds of crosspoints; more switches can be ganged for virtually unlimited capacity.



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!



An Axia audio node makes an excellent replacement for PC soundcards (see sidebar page 8). With its high performance A/D/A converters, onboard power supply and pro-grade specs, this approach helps eliminate the problems plaguing internal PC soundcards.

You say all Axia products use Livewire technology. What exactly is Livewire? How does it compare to other audio networking systems?

Livewire is an audio networking system that allows real-time uncompressed digital audio to be conveyed over standard Ethernet hardware. There are a few alternative systems now on the market, but Livewire is unique in several key ways: First, Livewire is extremely low latency, about 80% faster than any other audio-over-Ethernet approach. This is especially important for broadcast facility operation, where live monitoring and cascaded links are common. Second, Livewire includes all the technology needed for practical studio applications: Sources are ID'd and advertised to receivers, GPIO over the network is covered, etc. Third, Livewire connects directly to PCs — no soundcard or other hardware is required.

All Axia studio products are Livewire-enabled. We offer you all the pieces you need to build a modern broadcast studio. Audio adapter Nodes, Engines, Surfaces, PC drivers. We are experienced broadcasters, so we know how to support radio studio applications.

So, what about that 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 .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 email, 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 a 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 moxt 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 and more will be available soon.

Are optical audio links supported?

Livewire is fully compatible with 100BT and 1000BT copper and fiber connection types. We imagine a common configuration to be switches dedicated to studios with 100BT copper connecting terminals, engines, surfaces, etc. A fiber backbone connects the switches in order to share audio between the studios.

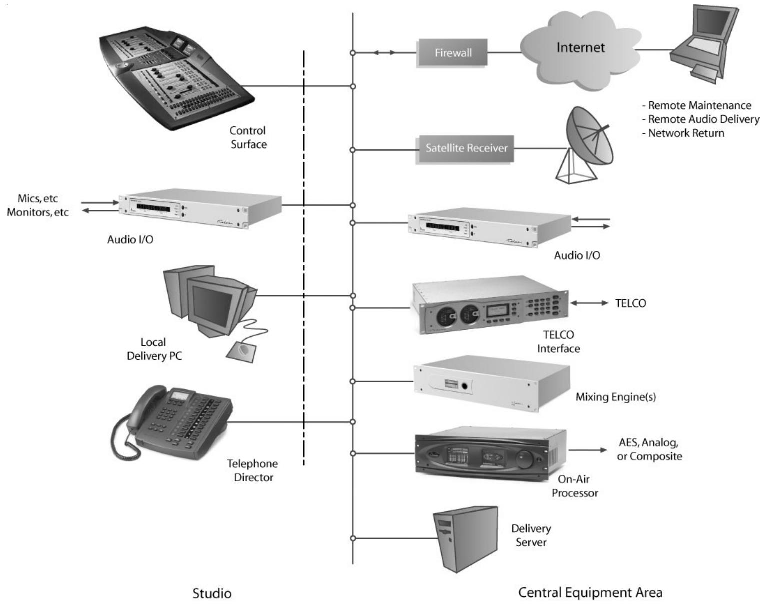
Is Livewire standards-based?

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 directly plugs to 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.



Now with Axia, broadcast plants can take full advantage of Ethernet for live audio, remote audio, associated control, data, telephony and remote maintenance.

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