

# Radio Guide

Radio Technology for Engineers and Managers

January 2005

## IP-Audio Distribution Moves Into the Studio



**Inside**

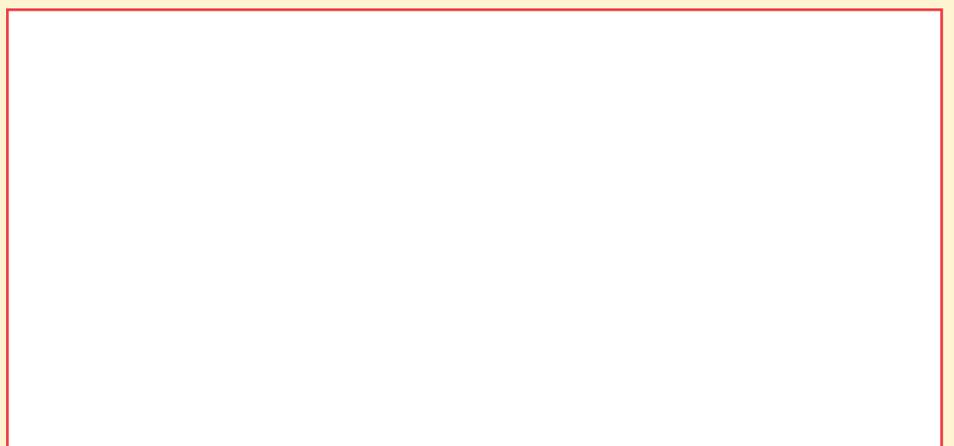
**IP-Audio Routing**

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# IP-Audio Routing

by Clark Novak, Axia Audio



## The Technology Underlying Axia

As more and more stations transition to or plan for digital audio plants, it is important to understand the technologies that make it work. This month Clark Novak, from Axia Audio, explains how the IP-Audio Routing brings benefits to users.

[CLEVELAND, Ohio - January 2005] As long as there has been information, people have searched for efficient ways of sharing and distributing it. History credits the first "information system" to the French, who in 1791 developed an "Optical Telegraph Network" that used a system of cross-arms and pulleys to transmit messages at the blinding speed of 20 characters per minute.

### A WHOLE DIFFERENT NETWORK

When we say "network" today, most folks immediately think of Ethernet. Devised in the 1970s at Xerox's PARC, this use of distributed packet switching to connect local computer networks has evolved into a high-speed universal connection method for sharing all sorts of digital information.

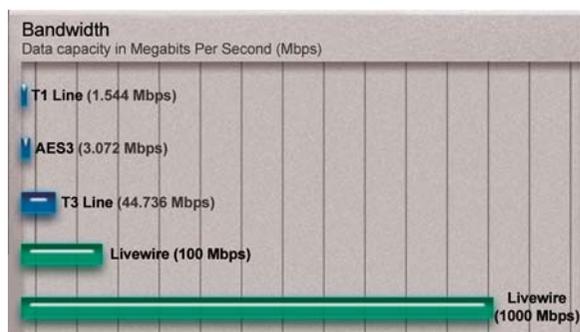
You may or may not have thought about it, but Ethernet is actually the most common digital audio transmission method in today's broadcast facilities, connecting audio delivery servers with studio computers. A natural extension of the technology would be to use Ethernet as a low-cost, universal way to connect audio and data for *everything* – including real-time audio – in broadcast studios.

Of course, it is not just the connection that is needed, but a connection with a pipe big enough and reliable enough to handle low-delay, high-reliability uncompressed audio over switched Ethernet. Axia Audio's solution for professional audio delivery is called "Livewire™."

Livewire is the core technology in all Axia products. It allows transport of real-time, "live" audio, plus Program Associated Data (PAD) and machine remote control over a network of switched Ethernet, a technique called "IP-Audio." This network can also carry file transfers, messaging and other routine traffic, resulting in a true "converged" network for the broadcast plant.

### NO MORE PATCH BAYS

"Ethernet connects everything," says Axia President Michael "Catfish" Dosch. "Two devices connect with a length of CAT-6 cable; multiple devices connect through an Ethernet switch. The same wire conveys audio, logic and control messaging, program-associated data – even general IP traffic. Very clean, simple and elegant."



Ethernet has more than enough bandwidth for uncompressed digital audio.

The Axia Livewire network replaces the need for a cross-point audio switcher, since every source is avail-

able to every destination on the network. The network is designed to be naturally scalable, and can effectively cross-connect either a few studios, or a few dozen. This way a single Ethernet switch will support hundreds of cross-points.

Ethernet's enormous data capacity makes it possible for a 100Base-T segment to carry 25 stereo channels of 48 kHz, 24-bit linear PCM audio in both directions. A 1000Base-T or Gigabit fiber link can handle ten times that amount – up to 32,000 stereo channels per system.

### HOW IT ALL WORKS

Livewire networks have three parts: the **Ethernet backbone**, the **software layer**, and the **user layer**.

Why Ethernet? According to Telos founder Steve Church, "Low-cost mass-market Ethernet switches offer us something very interesting. Since their function is to direct packets from port-to-port, we can use them to move our audio signals from whatever source to whatever destinations we want. This means we get a simple, flexible, facility-wide audio routing system for almost free."

Livewire has an audio advertising system based on the familiar concept of IP addressing, Church continues. "Every source has a text name and numeric ID. These are transmitted from source devices to the network. Receivers can build lists of all available sources from which users can select. With hardware nodes, you enter the names, numbers, and other configuration information via an attached PC with a web browser. With PC nodes, you open a configuration window."

In this manner, Livewire networks are constructed using a "building block" approach: determine the location and number of your audio inputs, then place Axia Audio Nodes next to them. There is an Analog Node for line-level sources, an AES/EBU Node to handle digital streams, a Microphone Node (with internal preamps) for microphone inputs, a General Purpose I/O (GPIO) Node for logic-follows-source machine control, and a Router Selector Node, a unique hybrid X-Y controller with its own AES and analog I/O ports.

Users like the fact that routing systems built using Livewire typically cost about half that of a hardwired TDM router. "With a traditional router, you spend something in the neighborhood of \$100,000 just to put one console on the air, plus you need the special links to get audio into the frame, and adding another room means you have to invest in even more proprietary gear," explains Jim Hibbard of Pacific Mobile Recorders.

Hibbard, who installed Axia for Univision's Austin, Texas cluster notes, "Adding another studio with Axia is real simple, not a break-the-bank proposition. All we have to do is plug it in."

### AVOIDING TRAFFIC DELAYS

In the "software layer," audio is sent in two ways: **livestreams**, which use small, fast packets to send uncompressed 48 kHz/24-bit PCM audio to network destinations in real time, and **standard streams**, whose larger, slower packets use the Internet's RTP/IP protocol to transfer pre-recorded audio. All Axia hardware transmits and receives both stream types; path selection happens transparently, without user intervention.

Whichever path is used, Livewire audio overcomes typical network issues such as the delay that

plagues Internet audio. Such delay is often multiple seconds because of the long buffers needed to ride out network problems and the delays inherent in multiple-hop router paths.

On the other hand, without the limitations of the public Internet, and with 100% control over all parts of the system, live audio can be transported without delay or dropouts at full quality. Livewire's specifications show live audio delay to be *less than 1 millisecond* per network hop – comparable to any professional A/D or D/A converter.

The Livewire network capacity is tremendous (up to 32,000 stereo channels) – well above the needs of a typical broadcast facility. The capacity is enhanced by the fact that all audio streams stop at the local Ethernet switch, consuming no network bandwidth unless and until a receiver subscribes to them. Each receiver takes only the stream it needs, eliminating the problem of unrequested data flooding the network.



A small (16x16 stereo) routing switcher configuration.

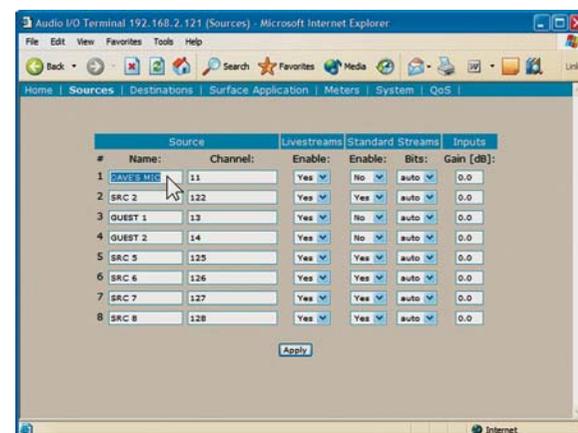
### EXTERNAL CONNECTIONS

Since Livewire networks are Ethernet-based, the time and effort needed to install and configure components is significantly less than that needed for a comparable hard-wired router.

Axia's "user layer" consists of the devices used to translate analog and digital audio into network data packets. These "Audio Nodes" come in several different versions, each providing eight inputs to feed audio into the network, and eight programmable outputs to send audio back to studio devices or monitors.

Each studio's equipment list includes an Ethernet switch such as HP's ProCurve 2626, which has the guaranteed bandwidth and IGMP support necessary for Livewire. Setup begins by connecting each studio's audio nodes to that room's local switch using CAT-6 cable; inter-studio connections are made using a single 1000-BaseT link between switches.

Mark Manolio, Chief Engineer of Cleveland's WCSB-FM (where he installed one of the first Livewire systems) and an Axia support technician, gives us an overview of how Livewire studios are configured: "First, you connect your studio devices to Axia Audio Nodes. Audio Node inputs use standard RJ-45 jacks – except for mic inputs, which take XLRs. Users can crimp their own cables or use prewired adaptors.



### Browsing the Audio I/O Node

"Next, you assign each Audio Node its own IP. This is done using the Node's front-panel controls. Then you can use any networked computer to access the Node's setup screens using your Web browser.

"Using these setup screens, sources and destinations can be defined by giving them descriptive names, choosing stream priority and input gain settings, and defining GPIO options if needed," Manolio says.

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# IP-Audio Routing

by Clark Novak

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Mixing and studio control is handled by the Axia SmartSurface™, a programmable 16-fader controller with all the functions of a traditional console – plus a bunch that were never possible on your BMX.

The SmartSurface has four program buses, extensive monitoring and talkback capabilities, and a GPIO interface to control connected devices. Users can save unique profiles with different layouts and defaults, any of which can be recalled on any networked SmartSurface.



WCSB-FM, Cleveland

## LINUX ENGINE UNDER THE HOOD

The actual audio mixing is handled by the Axia StudioEngine, a network-based DSP mixing engine running real-time Linux to ensure bullet-proof, 24/7 operation. Axia plans to make additional control surfaces available in 2005.

To eliminate any need for D-to-A conversions in the sound card, the IP-Audio Driver enables computers to exchange audio directly using their Ethernet ports on the Livewire network. A multi-channel version of the IP-Audio Driver for use with audio delivery systems is available from Axia development partners Enco Systems, Prophet Systems and Scott Studios.

To assign and control the audio sources iPlay, an audio monitoring application for Windows workstations, allows individuals to select and monitor any networked audio stream.



iPlay assignment screen.

Another powerful network configuration tool – PathfinderPC – lets authorized users control the Audio Nodes, allowing engineers to build and manage facility-wide routing applications.

PathfinderPC can change between presets manually, on a day-part schedule, or in response to an external trigger from an automation system or other source. Developed by longtime Telos' partner Software Authority (whose work will be known to many Zephyr users), it can even be programmed to sense problems such as silence at a particular audio port, and patch around it without user intervention.

## EARLY REPORTS

In early 2004, Auburn University's WEGL-FM, in Auburn, AL became the first station to build a studio using Livewire.

"We have been using this system for about 10 months. I could not be happier with the products, service, and dependability that Telos has assembled in their Axia equipment," says station Chief Engineer Marc Johnson. "The Axia equipment came straight out of the box and was on the air in a matter of hours. All we did was place the new gear, plug in the power, and connect it to the network."



Radio Skonto, Riga, Latvia

Ivo Bankavs, CE at Radio Skonto in Riga, Latvia – the first international installation of Livewire technology – is pleased with the reliability.

"We've had no failure or downtime of any kind since we installed Axia," says Bankavs. "The company

warned us that we would be among the first to use this tech, so we were ready for start-up bugs," he says, "but we've been happy – there have been none."

While early adopters have been smaller stations, the Livewire concept will get its first major-market workout when WOR installs it in a new ten-studio facility in New York City, as part of a move in early 2005.

Chief Engineer Tom Ray says it was an easy choice: "WOR was looking for a high tech digital solution that was also cost effective. The Axia system provides all the flexibility the WOR operation requires, allows a mixture of analog and digital I/O, and was extremely cost effective. The ability to access virtually any audio source anywhere in the facility is mind-boggling," says Ray.

Clark Novak is a marketing team leader for Axia products. He is happy to answer questions at: [cnovak@AxiaAudio.com](mailto:cnovak@AxiaAudio.com)



## AudioVAULT Big Bang for Small Bucks.

No, we didn't change our price list... AudioVAULT has always been an economical, modular solution for small- and mid-sized stations requiring the right balance to meet programming, operational and budget requirements. Support of multiple studios and stations, as well as true IP networking, are only some of the reasons AudioVAULT is also the first choice for major markets. The latest version of reliable, flexible AudioVAULT provides individualized user interfaces, and integrates with RDS and HD Radio data, including secondary audio services, such as Tomorrow Radio. AudioVAULT can improve your productivity and profit, backed 24/7 by a company you know you can trust. Contact BE today for a custom quotation... and be prepared to spend less for more.



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