Axia
Element and Livewire

By Chris Karb, PE

The Axia Audio system is a versatile and powerful studio audio system. Axia uses its Livewire protocol to encode audio into Ethernet packets, then mixes and routes them with Linux-based mix engines and standard Ethernet switches.

The system uses hardware interfaces called nodes to encode/decode audio to the IP stream. These nodes accept and deliver analog and AES3 audio inputs and outputs. Audio is encoded at 48kHz, 24-bit, giving a signal-to-noise ratio better than 102dB analog in to analog out and 138dB digital in to digital out. Contact closures (inputs and outputs) are handled by GPIO (general purpose input/output) nodes. Computer audio is encoded using a special Axia Audio driver application that encodes the audio into the Livewire IP stream. No sound cards are needed on audio servers or PCs.

The nodes connect to an Ethernet switch (edge switch), to which the control surface and mix engine also connect. CAT-6 is needed for the gigabit connection to the mix engines and the uplink to the core switch, which comprise the central part of the system. Nodes connect to the edge switches with CAT-5 cable. Each of the eight input and output channels in each node create a 4MB Livewire stream so 100Mb ports are more than enough for the nodes.

Our system pairs each studio with a production room and an edge switch, and the five edge

Performance at a glance

- Digital mixing and routing system
- Uses standard Ethernet switches for routing
- Web-browser configuration
- Cabling via CAT-5 or CAT-6
- Requires no sound cards
- More than 10,000 audio channels

switches (we have 11 studios) are connected to the core. We use another switch in the rack room to handle the nodes for our 50 satellite channels, codecs and such. Each PC driver also connects to its respective switch for the control room in which it is located.

A Windows server runs the Pathfinder PC audio/GPIO router that controls audio and contact closure routing for the entire facility. We run a hot standby with automatic fail over. Pathfinder is configured to accept closures from our five stations' automation systems, two digital recorders, 13 satellite receivers, EAS decoders/encoders and so forth. The closures can be configured to route audio from to a live board, satellite channel, backup studio or any source directly to air, or anything else likely to be needed, such as on-air lights. A clever software bridge designed by Software Authority interprets contact closures in and out of the automation server via IP to the Pathfinder server, replicating the analog hardware source switch normally associated with automation systems. It works flawlessly, and saves a tremendous amount of wiring and equipment. The router is completely configurable, can send e-mail notifications, execute stacked events on time or contact closure, just about anything one would want a router to do.

Stay connected

Connection to the outside world is easy using the Radio Systems Studio Hub+ RJ-45 dongles. The nodes use RJ-45 connectors for audio. The mic node has XLR connectors as well. We picked the connector we needed (XLR, $1/4$", $1/8$" mini, RCA, etc.) and the proper length of shielded jumper and connected it. No punchblocks, no cross-connects needed. No soldering, either. The equipment design is rugged, and we have experienced no mechanical failures in 18 months of operation in any of our 11 studios.

From the operator standpoint, the control surfaces work just like a traditional mixing console, except that the metering, clock and timer are viewed on a monitor. It took no time for the staff to get used to that.

Metering is provided for four stereo program channels. We assign one to the on-air program feed, another to the record bus, the third to codecs and the fourth to the phone recorder. Every channel generates its own mix-minus, which makes multiple phone and codec hookups a snap. There are an additional 48 mix channels in each mix engine that do not appear on the control surface but can be used to sum automation channels, and switch satellite and other sources. We use automation control closures to turn those channels on and off, and

the associated contact closures are routed to the audio server to start breaks and play liners.

The complete configurability of the system is dizzying at first, but the Web-browser interfaces are easy to understand. Because every source is available to every channel, one has to use care in configuring what is accessible in each channel to just those that the operator is likely to need. Our facility has almost 1,000 audio sources in use. The system is capable of handling more than 10,000 channels.

Changing channel assignments involves the

Axia Audio

P 216-241-7225

W www.axiaaudio.com

E inquiry@axiaaudio.com

push of a button on the channel strip and then selecting the new source with a rotary control on the monitoring panel. The push of a second button calls compression and equalization if enabled on the source for that particular channel. All standard compression, expansion, gating and equalization parameters are included, and software buttons allow for quick dial-in. The sound is much better than any of our traditional mic processors.

Determining the overall system organization at the very beginning is extremely important. All networked equipment is assigned a fixed IP address, so it is important to group studios, analog nodes, GPIO nodes, PC drivers and switches into address sections with plenty of spare addresses so that expanding the system does not result in an address that's out of order. That leads to confusion in a large system in a hurry. A spreadsheet is provided to help with the organization and the Axia folks are pleased to assist and recommend.

Something special

Our system used several innovations that had not been tried before, such as mixing and switching audio sources with the virtual mix busses, and the IP bridge to the PathfinderPC router from the Automation servers. This required the commitment of Axia and RCS Automation to help get everything working properly.

Factory personnel were on-site to perform the initial control set-up and switch configuration, and get the Pathfinder router running. It is always a good idea to have the factory people get the system going for you, especially if your

system is large. They can do it faster and show you what you need to know in a hurry. Any difficulties will be dealt with in a flash, and you'll have contact people who know your setup. It really helped us.

It took several weeks of steady clacking to configure our system, assigning source names, choosing channel source availability and source parameters, setting up contact closure assignments from the satellite channels and so forth. It is different to be sure, but our engineers quickly became familiar with it. The laptop is the new "greenie".

Since installation we have had two memory chips in the control surfaces fail, (none resulting in an off-air emergency) and one case of overloaded mix engine due to the busyness of our 24-channel talk studio with six processed mic channels running at once. The memory chips were quickly replaced and the overloaded processor issue was solved by switching the talk mix engines to dual-core processors, which are now standard on all engines. Axia's tech support is extremely good, which is vital when dealing with new technology.

Our system has been very reliable and versatile. It is easy to troubleshoot and quick to reconfigure for special needs. Upon launching the first sta-

The Livewire audio system includes a wide range of nodes and interfaces.

tion to air, we noticed an immediate extension in frequency response on both low and high end as well as an improvement in depth and clarity, with the same audio chain we had been using before. The board ops and DJs love it. We would not build another cluster without an Axia system.

Karb is director of engineering and IT for Clear Channel Asheville, NC.

**2101 Superior Ave.
Cleveland, Ohio, USA 44114
216-241-7255
inquiry@AxiaAudio.com**