AXIA xNODES
ADVANCED IP-AUDIO INTERFACES

ONE DAY, ALL BROADCAST GEAR WILL BE NETWORKED. UNTIL THEN, THERE'S xNODES.

Broadcast audio studios have really changed in the last decade. Axia's introduction of cost-efficient, high-performance audio routing systems that transport audio using standard Ethernet changed everyone's way of thinking – and opened up a whole new world of operational possibilities.

Dozens of hardware and software partners now offer broadcast products that connect to Axia networks using nothing more than a CAT-5 cable for audio, control and PAD, and more companies join every day. Everyone agrees that one day, all of the audio devices and peripherals that make up the modern broadcast studio will be network-connected. Until then, you need a way to add and control non-networked audio devices to your studio network. The solution: Axia xNodes.

ABOUT AXIA

Axia is the AoIP division of Telos Systems, a technology leader in professional audio equipment for radio broadcasters since 1984. In 2003, we introduced the world's first Ethernet-based console system for broadcasting. At the time this was a new idea, but VoIP showed the telecom industry how powerful, flexible and cost-efficient it was to move audio via IP, and the idea caught on fast with broadcasters, too. AoIP (sometimes called IP-Audio) is

mainstream technology now, with more than 2,500 radio studios around the world equipped with Axia networks.

THE POWER OF THE NETWORK.

Axia helps you build networked studio facilities to meet today's most demanding broadcasting applications. With Axia, you can quickly and easily connect a few rooms, or an entire facility. Axia networks have a total system capacity of more than 10,000 audio streams, and can carry hundreds of digital stereo channels (plus machine logic and PAD) over a single CAT-6 cable, eliminating much of the cost normally associated with wiring labor and infrastructure.

For example: a couple of Axia interface nodes, connected together, can move a group of audio signals over an Ethernet cable from one room to another. Connect with fiber and you can go across campus. Attach a few more nodes and a switch and you have a distributed multi-room routing switcher. Plug in a mixing surface and console engine to add a powerful networked broadcast console. Add intercom stations for broadcast-quality plant communications that can be taken to air. Plug in your delivery system PC and you can transfer files, live audio, and associated data all over the same network. And since Axia audio is networked, Analog

THE POWER OF THE NETWORK

and digital signals are merged seamlessly; cross-point switching from any source to any destination is fast and easy.

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. Audio, of course, takes the highest priority and never misses a beat.

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.

But there's much more to Axia than just the network. Once all of your consoles, peripheral devices and computer workstations are connected together for unlimited sharing, it's easy to add phone systems, audio processors, codecs, satellite receivers, program delay units, or any audio device from the ever-growing list of Axia Partners. All of these devices work together in tight integration, which leads to more intuitive and intelligent operation. By taking advantage of the efficiencies of computer networking, Axia simplifies, saves you money, gives you choices, and prepares you for the computer and data-centric studios of today — and tomorrow.

xNODES: COMPACT, MOUNT-ANYWHERE NETWORKED AUDIO INTERFACES.

Until the day that all professional audio equipment comes with an Ethernet jack, a way to interface conventional audio equipment with Livewire networks is needed. That's where the Axia family of xNode audio adapters comes in. There are five versions:

» Analog XNode
» AES/EBU XNode
» Microphone xNode
» GPIO xNode
» Mixed Signal xNode with a blend of Mic, Analog, AES and GPIO.

All xNodes feature a high-resolution OLED front panel display and two "soft" buttons to provide status information and assist with initial setup, and a multi-color LED that gives at-a-glance information about the xNode's power configuration. To ensure ultra-reliable network operations and extremely low delay, xNodes run 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.

xNODES ARE DESIGNED FOR PEACE OF MIND.
Each xNode is equipped with two 100Base-T Ethernet connections, allowing two connections to the network switch fabric. Should the active connection fail – due to an unplugged cable or other problem – output is automatically, seamlessly switched to the second Ethernet connection. Likewise, xNodes can draw power from two sources: the fanless internal, auto-switching power supply, or via PoE (Power over Ethernet) from the network switch. Should mains power be interrupted, the xNode powers itself using PoE; again, automatically and seamlessly. (Each xNode model is also available in a PoE-only version suitable for deployment in hard-to-access areas where mains power is not available.) There's even SNMP support to keep you informed of what's going on.

xNODES ARE CONVENIENT.
For example, a microphone xNode placed in a studio can collect audio from microphones and also provide outputs to associated studio monitors and headphones. An xNode in the rack room can collect audio from network feeds, codecs and other shared sources for system-wide use while providing handy outputs for audio processors and other terminal-room gear.

xNODES ARE EASY TO DEPLOY.
When you connect an xNode to your network, it automatically prompts you to give it an ID. Then, derives a unique static IP address, and even gives names to its sources and outputs (which you can edit later, from the comfort of your computer). All you have to do is connect devices to the inputs, and it advertises that its audio sources are available for use, allowing any users access to them. Installing xNodes is easy — just place them near your audio source and destination devices, and distribute them throughout your facility wherever it's convenient. Place them in cabinets or under desks; rack-mount them singly, or in pairs, with an optional rackmount kit (two xNodes fit side-by-side in a single RU) — even attach them flat to a wall or under a countertop, using the optional surface-mount kit.

xNODES PROVIDE SUPERIOR AUDIO QUALITY.
Not only are xNodes capable of operating at a network sampling rate of 48 kHz, they also employ high-resolution 32-bit floating-point SRC chips. xNodes produce a "sweeter," more natural audio quality — clients routinely tell us of noticeable sonic improvements after the installation of their Axia network.

TECHTALK BLOG
TELOSALLIANCE.COM/BLOG

BROCHURES
AXIAAUDIO.COM/BROCHURES

SOFTWARE UPDATES
AXIAAUDIO.COM/DOWNLOADS

FIND A DEALER
AXIAAUDIO.COM/BUY

THERE'S AN xNODE FOR EVERY NEED.

Audio comes in different flavors – Mic-level, AES/EBU, analog – so xNodes do, too. There's an xNode for each type of audio in your studio, plus one for GPIO machine-control logic, too.

MICROPHONE xNODE

Microphone xNode has four professional-grade microphone preamps with selectable Phantom power and software-adjustable gain. There are also four balanced analog line outputs to conveniently deliver headphone and studio monitor feeds back to your talent. Inputs and outputs are presented both on easy-to-install RJ-45s and high-density DB-25s, to suit your connection preference.

ANALOG xNODE

Analog xNode has 8 mono or 4 stereo balanced line-level inputs and 8 mono or 4 stereo balanced line-level outputs, on RJ-45 and DB-25 connectors. Each input is switchable to accommodate either consumer-level -10dBv or professional level +4dBu sources. The short-circuit protected outputs can deliver up to +24dBu before clipping. Axia uses only studio-grade A/D/A converters and low-noise components, so that each Analog node provides superior audio performance for high-end studio use.

AES/EBU xNODE

AES/EBU xNode provides 4 stereo AES inputs and 4 stereo AES outputs, which may also be treated as 8 mono inputs and outputs. Stunning performance specs include a 48 kHz sampling rate, 126dB of dynamic range, and <0.0003% THD.

MIXED SIGNAL xNODE

Mixed-Signal xNode is your utility player; perfect for places that require a mix of different audio I/O types. It provides 1 selectable Mic/Line analog input, 2 dedicated analog line inputs, 3 analog line outputs, 1 digital AES3 input and 1 AES3 output, and 2 GPIO ports – truly a "jack of all trades."

GPIO xNODE

GPIO xNode provides 6 general-purpose logic ports for machine control of studio peripherals – audio devices, loudspeaker muting relays, signal lamps, etc. – each with 5 opto-isolated inputs and 5 outputs. A logic port can be associated with any audio input or output and routes control data transparently along with the audio.

FEATURES AND BENEFITS

STUDIO FRIENDLY.
xNodes' fanless design uses heavy cast-aluminum heat-sinks; they are completely silent in-studio. There are no temperature worries when xNodes are placed into racks, since the front-panel heat sinks are cooled by ambient air, not "rack air."

INFORMATION AT THE READY.
High-resolution OLED displays speed setup, give software and status information. Multi-color LED glows green when AC mains power is present, red when running on PoE, and orange when both AC and PoE are connected.

PEACE OF MIND, BUILT IN.
Redundant power plan lets xNodes run on AC or Power over Ethernet (IEEE 802.3af). xNode will run on either power supply; connect both for redundant power with automatic switchover. Network connection has backup, too: dual NICs allow you to connect xNode to separate network branches for full audio pathway redundancy. xNodes are power-efficient, too: they use just 14 Watts each.

KNOWLEDGE IS POWER.
Built-in Syslog server with configurable event filter and SNMP (Simple Network Management Protocol) support help you stay fully informed, should an xNode's power or connection status change.

SYNCHRONICITY.
Want to sync your Axia Livewire network to a house master clock? No problem; simply feed your house clock to a designated xNode AES/EBU input to keep all of your AES streams running like. . . well, clockwork.

TOP-SHELF COMPONENTS.
Like all Axia product, xNodes are designed to keep running, and running, and running. xNodes use premium components: rugged cast aluminum faceplates and heat sinks, high-resolution OLED displays, bulletproof power supplies designed for high-availability telecom applications, studio-quality SRCs with specs a recording engineer would envy.

TECHTALK BLOG
TELOSALLIANCE.COM/BLOG

BROCHURES
AXIAAUDIO.COM/BROCHURES

SOFTWARE UPDATES
AXIAAUDIO.COM/DOWNLOADS

FIND A DEALER
AXIAAUDIO.COM/BUY

EASY INSTALLATION.

xNodes' I/O lets you have it your way: use industry-standard RJ-45 audio connectors, or high-density DB-25 connections, both available prefabricated and ready to attach in seconds. You'll never have to pick up a soldering iron again (unless you're feeling nostalgic).

STACK 'EM OR RACK 'EM.

xNodes are small – just 9.5" x 11" (25 x 28 cm.) and 1RU tall. You can use them freestanding, rack them side-by-side, or even wall-mount them using the accessory surface-mount kit.

MONO, OR STEREO? THE CHOICE IS YOURS.

xNodes can be configured to create 4 stereo audio channels for your network to use, or 8 mono channels. It can also receive 4 stereo or 8 mono channels, and send them to its outputs. No muss, no fuss – it's as easy as flicking a software switch.

FAQs

WHAT'S THE TECHNOLOGY BEHIND AXIA NETWORKING?

Axia systems use high-reliability switched Ethernet using patented Livewire networking technology developed by Telos. This sophisticated switching architecture eliminates the need for 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 this standardized transport backbone, Axia eliminates purpose-built hardware, which translates into dramatically reduced system cost.

HOW DO I KNOW THAT AUDIO OVER IP WILL BE RELIABLE?

Axia uses the same technology that underlies VoIP telephony. Did you know that nearly 80 of the Fortune 100 companies now use VoIP? Or that VoIP PBX systems now outsell the old kind by a wide margin? With these systems, telephones plug into a standard Ethernet/IP network. Contrast this with traditional PBX phone gear — proprietary devices which required you to purchase phone sets and parts exclusively from the company that built the mainframe. You were locked into a single vendor, because the technology that ran the mainframe was owned by the company that made the gear.

IP is now accepted as a universal transport for almost any kind of signal. You see it in television studios, business teleconferencing, government communications, banking, etc. And it's hardly unproven, especially for applications specific to radio studio infrastructure. As of 2012, over 3,000 studios around the world - many in mission-critical, 24/7 broadcast applications in major markets like New York City, Chicago, Paris, Rome and Bangkok - have been built using Axia IP-Audio infrastructure.

DOES YOUR SYSTEM ROUTE LOGIC WITH AUDIO, TOO?

Of course. IP is great for data, no? GPIO xNodes let you bring logic commands from external devices like CD players, DAT machines, etc., into the network. The logic data is then "bound" to the audio stream, and is routed with it to whatever console the source is loaded on.

FAQs

Devices equipped with Livewire interfaces (like the latest Telos Zephyrs and phone hybrids, Omnia audio processors, AudioScience cards and IDC satellite receivers, for example) supply audio and control logic directly from the device to the Ethernet switch over a single CAT-5e connection, further simplifying in-studio wiring and making Livewire's audio+logic routing even more convenient.

DO I HAVE TO BE AN IT EXPERT TO RUN AN AXIA SYSTEM?

An IP-Audio network is like a car: you don't have to understand how the engine works in order to drive it. Just connect two pieces of gear together with CAT-5e and they will talk to each other — like plugging a mic into a mixer. The Livewire protocol takes care of routing the audio without any need for intervention from you. And the equipment interface is all web-based with GUI control. It works intuitively, and you don't have to know anything about the tech inside to make it work. Axia consoles even include built-in, zero-configuration network switches — no network switch setup needed.

That having been said, another of the advantages of Ethernet and IP is that it's well-documented. Telos founder Steve Church and technologist Skip Pizzi have even collaborated on a book published by Focal Press, Audio Over IP: Building Pro AoIP Systems With Livewire. It's available from all the usual booksellers, and as an eBook as well.

MOST COMPANIES RECOMMEND THAT I BRING THEM ON-SITE TO HELP INSTALL AND CONFIGURE THEIR SYSTEMS. DO I NEED YOUR HELP TO INSTALL AN AXIA SYSTEM?

With those other guys, you'd better hire their systems engineers. With us, it's much easier! If you know how to use a Web browser and plug a telephone into the wall, you've got all the skills needed to install and configure your new Axia network. And Axia Technical Support is there to help if you need it, too. If you still decide you'd like on-site installation services, we'll be happy to talk with you about it.

WHAT ABOUT PROGRAM ASSOCIATED DATA? IS YOUR SYSTEM COMPATIBLE?

Yes. Devices that generate PAD plug into the Axia network; the information they supply is sent along with its associated audio, and any devices that need it can also plug into the network and retrieve it. This means that you can send audio and PAD together, without incurring extra costs for separate audio and data networks.

IS LIVEWIRE AUDIO COMPRESSED, OR LINEAR?

Livewire is not compressed. Axia xNodes produce linear 48 kHz, 24-bit studio-grade audio, and there are switches that have enough bandwidth to carry 10,000+ channels of uncompressed, real-time stereo audio simultaneously.

IF I USE xNODES TO BUILD AN AXIA ROUTING NETWORK, DO I HAVE TO USE AXIA CONSOLES?

Not necessarily. Axia networks can work with your existing consoles — just plug the inputs and outputs into the xNodes and add our XY Controllers for route switching. For sophisticated systems, use our PathfinderPC router control software package. You can do everything any other router can do – and much more. Of course, if you do decide to use Axia consoles, you'll have the advantage of features like automatic mix-minus on every channel, the ability to control Telos phone systems and codecs right from the mixing board, the option to use integrated Intercom systems with broadcast-quality audio, and even the ability to remotely-control your console from another room – or even off-site. And setup is simple — the built-in network switch inside Axia integrated console engines lets you daisy-chain up to 4 consoles together without the need for an external switch.

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.

I'VE GOT A LARGE FACILITY. HOW MANY STUDIOS CAN I INTER-CONNECT?

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.

I HAVE A LOT OF MONO SOURCES AT MY FACILITY. CAN xNODES HANDLE MONO SOURCES, OR ARE THEY STRICTLY STEREO?

Like all Axia gear, xNodes give you plenty of choices. You can run an Analog or AES xNode in stereo mode, generating 4 channels of stereo input and 4 channels of stereo output, or mono mode, with 8 channels in and out.

THIS SOUNDS PRETTY SOPHISTICATED. WHAT ABOUT FOR SMALLER STATIONS?

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.

And, should you choose to use Axia consoles, it gets even simpler, thanks to the onboard, zero-configuration network switch

TECHTALK BLOG 🌐
TELOSALLIANCE.COM/BLOG

BROCHURES 📄
AXIAAUDIO.COM/BROCHURES

SOFTWARE UPDATES ⬇
AXIAAUDIO.COM/DOWNLOADS

FIND A DEALER 🔍
AXIAAUDIO.COM/BUY

that's built into every Axia integrated console engine. These engines include I/O for mics, analog and AES inputs and outputs, GPIO ports and Livewire inputs for networked audio devices, along with a zero-configuration Gigabit network switch that lets you daisy-chain up to 4 studios without a core switch. Thanks to the "all in one" nature of these mixing engines, Axia consoles are as well-suited for standalone operation as they are for networked installations.

THIS SEEMS LIKE A LOT OF IP TO KEEP TRACK OF. ARE THERE ANY ADMINISTRATION TOOLS?

All Axia devices have a web browser control and monitoring capability. You can address them individually, or use our iProbe network management software, which documents keeps track of every device connected to your network.

HOW DO I ADD AUDIO SOURCES FROM NON-NETWORKED DEVICES TO THE NETWORK?

With xNodes. These come in variants for line and microphone applications.

There are also a growing number of Livewire partners, offering dozens of software and hardware products that connect directly to Livewire via CAT-5. Audio, control and data all travel down that one cable, making it even easier and more efficient to add new audio peripherals to your network.

WHAT HAPPENS IF SOMEONE ACCIDENTALLY UNPLUGS A CABLE? WHAT THEN?

Axia networks utilize network switches with Spanning Tree Protocol, a link management protocol that provides path redundancy while preventing network loops. Axia xNodes take full advantage of STP, with dual NICs that allow simultaneous, redundant connections to your audio network. In the event that one cable is unplugged, audio streams are automatically, seamlessly routed to the second, redundant connection.

I READ AN ARTICLE ABOUT BROADCAST AUDIO NETWORKS AND HOW THEY RELATE TO THE OPEN SYSTEMS INTERCONNECTION (OSI) REFERENCE MODEL. WHERE DOES AXIA FIT IN THAT MODEL?

The OSI model is a seven-layer framework that defines network functions. Axia operates on Levels 1 through 5 of the OSI model. The IP Network Layer provides routing capabilities. The UDP Layer provides transport service using port numbers. The RTP Layer provides additional information about streams, helping to identify media format and look for dropped or duplicate packets, as well as other useful functionality using the RTP standard for best compatibility with other media streaming solutions.

xNODE SPECIFICATIONS

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 inputMeta

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

➤ Signal Format: AES3 (AES/EBU)

➤ AES3 Input Compliance: 24-bit with sample rate conversion

➤ AES3 Output Compliance: 24-bit

➤ Digital Reference: Internal (network timebase) or external reference 48 kHz, +/- 2 ppm

➤ Internal Sampling Rate: 48 kHz

➤ Input Sample Rate: 32 kHz to 192 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)

xNODE SPECIFICATIONS

DYNAMIC RANGE

➤ Analog Inputs to Analog Outputs 108dB referenced to 0dBFs, 111dB A-weighted

➤ Analog Inputs to Digital Outputs 110dB referenced to 0dBFs, 113dB A-weighted

➤ Digital Inputs to Analog Outputs 112dB referenced to 0dBFs, 115dB A-weighted

➤ Digital Inputs to Digital Outputs 126dB

EQUIVALENT INPUT NOISE

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

TOTAL HARMONIC DISTORTION + NOISE

➤ Mic Pre Input to Analog Output: < 0.005%, 1 kHz, -36dBu input, +18dBu output

➤ Analog Input to Analog Output: < 0.005%, 1 kHz, +18dBu input, +18dBu output

➤ Analog Input to Digital Output: < 0.004%, 1 kHz, +18dBu input, -6dBFs output

➤ Digital Input to Analog Output: < 0.004%, 1 kHz, -6dBFs input, +18dBu output

➤ Digital Input to Digital Output: < 0.0003%, 1 kHz, -20dBFs

CROSSTALK ISOLATION, STEREO SEPARATION AND CMRR

➤ Analog Line channel to channel isolation: 90dB minimum, 20Hz to 20kHz

➤ Analog Line stereo separation: 85dB minimum, 20Hz to 20kHz

➤ Analog Line Input CMRR: 80dB minimum, 20Hz to 20kHz

➤ Microphone Input CMRR: >60 dB, 20 Hz to 20 kHz
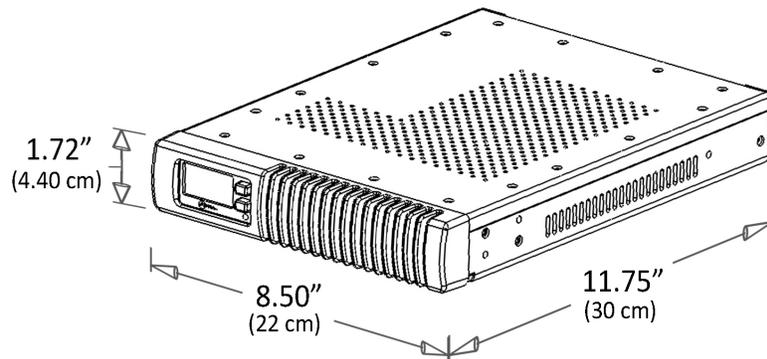
POWER SUPPLY AC INPUT

➤ Auto-ranging supply, 95VAC to 240VAC, 50 Hz to 60 Hz, IEC receptacle, internal fuse

➤ Power consumption: 14 Watts

OPERATING TEMPERATURES

➤ 0 degree C to +40 degree C, <90% humidity, no condensation

DIMENSIONS AND WEIGHT

➤ 8.5" (22 cm) wide; two may be mounted side-by-side in a standard 1RU rack space (with accessory mounting kit)

➤ 1.72" (4.4 cm) height, 11.75" (30 cm) depth

➤ Shipping Weight: 7 lbs. (3.2 kg.)

➤ Shipping Dimensions: 17" (43.2 cm) length, 13" (33 cm) width, 7" (17.8 cm) height

1.72" (4.40 cm)

8.50" (22 cm)

11.75" (30 cm)

TECHTALK BLOG 🌐
TELOSALLIANCE.COM/BLOG

BROCHURES 📄
AXIAAUDIO.COM/BROCHURES

SOFTWARE UPDATES ⬇
AXIAAUDIO.COM/DOWNLOADS

FIND A DEALER 🔍
AXIAAUDIO.COM/BUY

ARCHITECT'S SPECIFICATION

The following describes a series of five appliances which will provide a way to interface both audio and control signals from conventional audio equipment to Livewire networks. The series will include:

» Analog audio interface
» AES/EBU audio interface
» Microphone audio interface
» GPIO logic interface
» Mixed Signal interface including a mixture of analog, AES/EBU and Microphone audio, and GPIO logic interfaces

The signals received by these appliances shall be converted into network packets and delivered to a network supporting multi-cast (as defined in RFC-1112).

The appliance shall be of a professional design, suitable for use by broadcasters, pro audio technicians, and others requiring reliable, 24/7 unattended operation. The appliance shall feature professional audio inputs and outputs and/or support GPIO control logic up to 48 volts. The appliance shall have a front-panel display which enables audio monitoring and basic configuration. The appliance shall include remote monitoring and configuration accessible via standard Web browser.

The appliance shall stream up to 4 stereo or 8 mono channels to a network. The appliance will receive up to 8 streams from a network. The appliance shall support power over Ethernet (PoE) as defined in IEEE 802.3af-2003 allowing the appliance to operate without direct connection to AC power mains. The appliance shall connect to the studio network via 100Base-T Ethernet for superior noise/RF immunity, and shall provide two Ethernet connectors for dual-redundant connections to the network switch fabric.

PHYSICAL

The appliance shall consist of a 1RU, 9.5" x 11" (25 x 28 cm.) enclosure. Appliances shall include provisions for side by side rack mounting, or wall mounting, with accessory rack-mount or surface-mount kit. The front panel shall be an attractive, yet functional design with a bright, clear OLED-based display, controls to invoke display of status information and setup mode, and a multi-color LED to convey power configuration information. Heat sinks are integral to the front panel, eliminating the need for fans or cooling via rack air. The appliance shall be quiet in operation, fanless, and connect to AC power via an IEC power entry module. AC power input shall be of universal design, accommodating worldwide standard AC power voltages and frequencies. Appliance shall further be capable of utilizing drawing PoE power from an attached Ethernet switch as defined in IEEE 802.3af-200.

Microphone, Analog and AES/EBU versions shall have audio in-puts and outputs presented both on RJ-45 and high-density DB-25 female connectors.

Mixed-Signal versions shall have two high-density DA-15 female connectors for GPIO logic. Appliance shall further include eight RJ-45 connectors for analog and AES/EBU audio, and one XLR-F bulkhead connector for microphone audio.

GPIO xNode shall have six high density DA-15 female connectors.

The rear panel of all appliances shall provide two Ethernet/IP (network) connections on standard RJ-45 bulkhead connectors.

AUDIO INPUT AND OUTPUT

The AoIP audio I/O functionality of descrbed appliances shall be fully compatible with the Livewire AoIP standard.

Microphone appliance shall have four professional-grade microphone preamps with selectable Phantom power and software-adjustable gain, and four balanced stereo analog line outputs.

Analog appliance shall have audio inputs which may be configured as either eight mono or four stereo balanced line-level inputs, and audio outputs which may be configured as either eight mono or four stereo balanced line-level outputs. Connections shall be presented on both RJ-45 and DB-25 connectors. Each input shall be switchable to accommodate either consumer-level –10dBv or professional level +4dBu sources.

AES/EBU appliance shall have audio inputs which may be configured as either eight mono or four stereo balanced line-level inputs, and audio outputs which may be configured as either eight mono or four stereo balanced line-level outputs. Connections shall be presented on both RJ-45 and DB-25 connectors. Outputs shall be short-circuit protected.

Mixed-Signal appliance shall have one selectable Mic/Line analog input, two dedicated stereo analog line inputs, three stereo analog line outputs, one stereo digital AES3 input and one stereo AES3 output. Microphone input shall have selectable Phantom power and software-adjustable gain. Outputs shall be short-circuit protected.

MACHINE CONTROL LOGIC

GPIO appliance shall have six general-purpose logic ports, presented on DA-15 connectors. Each port shall include five opto-isolated inputs and five outputs. These logic ports shall be able, by means of software configuration, to be associated with device(s) which provide audio input or output to Livewire AoIP networks.

NETWORK SUPPORT

The appliance shall support two self-contained Ethernet/IP network interfaces. Either connection may be used for remote control, monitoring, streaming, and AoIP (Livewire standard) audio I/O.

SYSTEM UPDATES

The appliance shall allow two versions (banks) of operating software to be stored internally. The appliance shall support the updating of its operating software via standard web browser access and file upload. The appliance shall further support web or file-based software updates with selection of booting to either software bank.

SUPPORT AND WARRANTY

The appliance shall be offered with a standard limited warranty period of two years. English-language factory support shall be available to users at no charge on a 24/7 basis.