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Issue 13 / May 2020

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Telos Infinity®

The AoIP Solution that Changes Everything

OVERVIEW

Telos Infinity IP Intercom

From the company that invented AoIP for broadcast.

Telos Infinity is the AoIP solution that delivers a quantum leap in scalability, ease of integration, efficiency, and total cost of ownership. As the first offering in the Infinity line up, Telos Infinity IP Intercom is a comprehensive next-generation communications solution, featuring:

- Seamless interoperability through standards-based Livewire+ AES67
- Infinite scalability with plug and play device integration
- Provides both communication and contribution audio
- Intuitive UI for simplified operation

- Matrix-free design
- Distributed DSP architecture
- Lower TCO than traditional intercom systems
- Optimization for broadcast communications

Telos Infinity IP Intercom is a complete reimagining of broadcast communications technology developed by the Telos Alliance engineering team that invented AoIP for broadcast in 2003. More than just a talkback system, Telos Infinity IP Intercom converges voice communication and contribution audio on a single IT backbone employing the latest standards-based VoIP and AoIP transport to provide dedicated features and functionality without compromise or limitations.

Break the Matrix

Telos Infinity IP Intercom replaces outmoded matrix technology with an advanced, distributed IP network solution that provides superior functionality in a simplified, more elegant form. Being matrix-free allows plug-and-play networked hardware and software devices to be added to the system as part of a planned or ad-hoc change, without ever worrying that you might exceed the number of available ports on a matrix.

As Part of an Integrated System...

Telos Infinity IP Intercom unleashes the full potential of a distributed IP audio infrastructure, allowing access to any networked audio endpoint through the intuitive Telos Infinity Dashboard application. Mix-minuses, program busses, mixer auxes, monitor feeds, remote contribution audio, and presenter mics are all available anywhere you need them - for communication or on-air use. And since Telos Infinity IP Intercom natively supports Livewire+ AES67, it can be used with our own AoIP products or those from other supporting manufacturers.

... Or as Part of a Legacy Installation

Realizing the benefits of Telos Infinity IP Intercom doesn't mean abandoning existing audio infrastructure. Instead, Telos Infinity seamlessly integrates into analog, AES, SDI, and MADI systems using Telos Alliance xNode baseband-to-IP interfaces and other AES67 partner devices. A transition to AoIP reduces the amount of cabling and its associated costs, system design time and installation expense - all while establishing a pathway to a complete AoIP solution in the future.

Infinite Possibilities

Telos Infinity IP Intercom marks the next generation of Audio over IP solutions destined to revolutionize broadcast communication through the creation of disruptive and innovative technologies - and it's just the start. More functionality, features and new products are just around the corner.

Award Winning

We're honored that Telos Infinity IP Intercom has been the recipient of several awards since its introduction, including TVB Europe Best of Show (IBC 2017), Radio World Best of Show (NAB 2018), TV Technology Best of Show (NAB 2018), and AV Technology Best of Show (InfoComm 2018).

IN DEPTH

Telos Infinity IP Intercom Panels

By combining familiar hardware Intercom facilities with the flexibility and workflow advantages of matrix-free IP communications, the INF-MP-16, INF-MP-16B, and INF-MPX-20 set a new benchmark in broadcast talkback technology.

The INF-MP-16 1RU Intercom Master Panel features 16 multicolor talk/listen lever display keys that can show up to four lines of text plus bitmap images and icons (coming soon). Complemented by optimized voice channels and loudspeaker reproduction, the INF-MP-16 features the excellent speech intelligibility required for mission critical broadcast operations. Each INF-MXP-20 Expansion Panel provides 20 additional display keys.

Auxiliary, rear panel stereo analog I/O including a studio quality mic pre-amplifier, means that the INF-MP-16 can be used as a contribution or monitoring endpoint. The addition of GPIO as standard enables remote control of signalization or complex logic interfacing across the network infrastructure. Features of the INF-MP-16 Intercom Master Panel include:

- 16 multi-color display lever keys with additional shift levels at the touch of a button
- Multi-channel duplex voice and contribution audio endpoint
- 24 bit/48 kHz IP audio
- Natively Livewire+ AES67 compliant
- Network connection via standard Gigabit Ethernet
- PoE+ or external DC power (or both for redundancy)
- Aux rear panel I/O including studio-grade mic input for on-air contribution
- GPIO via rear panel D-Sub as standard
- Headset connector supporting industry standard XLR-4M
- Configurable dynamics processing for improved speech intelligibility
- Full configuration via Telos Infinity Dashboard
- Device management via built-in Web UI

The INF-MP-16B 1RU Intercom Master Panel Base Version offers all of the same features as the INF-MP-16 but without the rear panel audio inputs and outputs, making it a more cost-effective option in applications where local I/O is not required.

The INF-MPX-20 1RU Expansion Panel adds 20 additional display keys when paired with the INF-MP-16 or IN-MP-16B Intercom Master Panels.

Telos Infinity IP Intercom Desktop Station

The Telos Infinity INF-DS-16 16 Key Intercom Master Desktop Station is ideal for locations where a rackmount panel isn't practical but where the same functionality and features from the INF-MP-16 are required in a compact and attractive format. By combining familiar hardware Intercom facilities with the flexibility and workflow advantages of matrix-free IP communications, the INF-MP-16 sets a new benchmark in broadcast talkback technology. The INF-MP-16 1RU Intercom Master Panel features 16 multicolor talk/listen lever display keys that can show up to four lines of text plus bitmap images and icons (coming soon). Complemented by optimized voice channels and loudspeaker reproduction, the INF-MP-16 features the excellent speech intelligibility required for mission-critical broadcast operations.

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Telos Infinity IP Intercom Beltpacks

The Telos Infinity BP-2 is a next-generation wired Dual Channel Digital Beltpack that is powered by PoE (Power over Ethernet) and is natively AES67 compliant. Whether via Partyline, Group, IFB, Peer to Peer, or connection to any other IP audio networked element within the wider Telos Infinity IP Intercom, the Infinity BP-2 communicates across the network seamlessly while maintaining incredibly low latency.

Multiple INF-BP-2s can be daisy-chained together, powered from the same physical PoE port, providing a functionally superior alternative to traditional 2-wire or digital-audio-based partyline systems.

- Dual channel duplex communication plus auxiliary program source
- 24 bit/48 kHz IP audio
- Natively Livewire+ AES67 compliant
- Rugged, lightweight design with interchangeable belt clip/mounting accessory
- Network connection via standard 100mbit Ethernet
- PoE or PoE+ with managed daisy-chain connectivity
- Headset connector supporting industry-standard XLR-4M
- Internal mic/speaker for handsfree walkie-talkie, desktop or wall-mount use
- Configurable dynamics processing for improved speech intelligibility
- Full configuration via Telos Infinity Dashboard
- Device management via built-in Web UI

Telos Infinity IP Intercom Lite Single Ear Wired Headset

Professional single-ear lightweight headset specifically designed for exceptional comfort, flexibility, and durability as needed in the most demanding professional environments.

Features:

- Single-ear lightweight design with enhanced acoustic isolation
- Closed back supra-aural (on ear) design
- Flexible, ambidextrous swiveling mic boom
- Flip-up microphone mute
- Dynamic noise-cancelling cardioid microphone optimized for voice communications

- Comfort fit adjustable headband
- Replaceable ear cushion pads
- Non-reflective rubberized matte black finish

Includes:

- Fixed 5 ft. (1.52 m) cable with 4-pin XLR Female connector termination
- Ear Sock
- Magnetic Cable Clip

Telos Infinity IP Intercom Dashboard Software

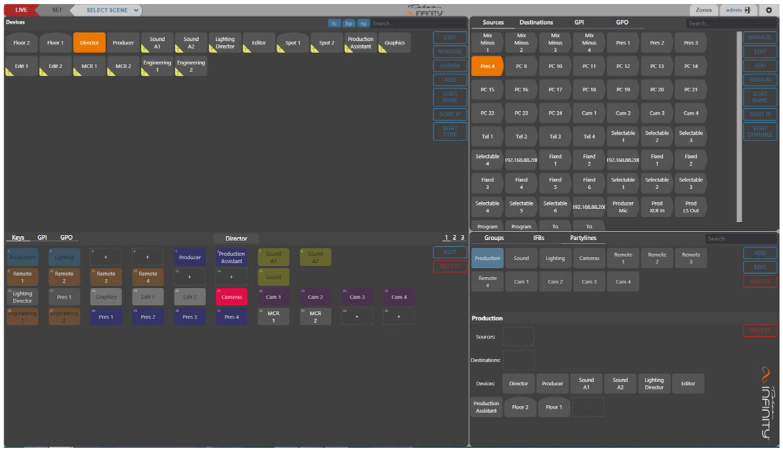
Telos Infinity Dashboard is unlike other Intercom system management tool. By displaying the entire Intercom system as a single page view without the clutter of multiple tabs and embedded menus, the operator can navigate quickly, and within minutes take control to create or manage a working IP Intercom. Much more than a simple configuration tool, Dashboard also manages all hardware and software elements within the system, including software and firmware updates, and monitors network performance and fault logging.

Dashboard Basic

A single user application of the Dashboard Basic license is supplied with all Telos Infinity systems. For many users with smaller systems, its basic feature set and ability to provide direct online control will provide all of the necessary tools to set up, configure, and customize their Telos Infinity system.

Dashboad Basic allows users to create unlimited Groups, Partylines and IFBs, drag-and-drop functions directly onto Panel and Beltpack keys in real-time, and manage AoIP sources and destinations as though they are physical ports on a virtualized Intercom matrix.

Dashboard Advanced



For installations requiring more features and flexibility, the optional Telos Infinity Dashboard Advanced software represents an optimized configuration and system management tool with an extended feature set to unlock the full potential of the multi award-winning Telos Infinity matrix-free IP Intercom System.

The Advanced Agent can be installed on either single or dual servers (coming soon) and constantly monitors the Telos Infinity Intercom network as well as any other connected AES67 devices, stream sources and destinations. User Clients connect to the Server Host via any HTML5 compatible web browser using a suitable connected device including PCs, Tablets and Mobile devices.

Dashboard Advanced can be used for all system configuration, from initial set up to 'on the fly' changes during live operation. However, Dashboard Advanced does not need to run constantly for Infinity to function normally (although it is highly recommended).

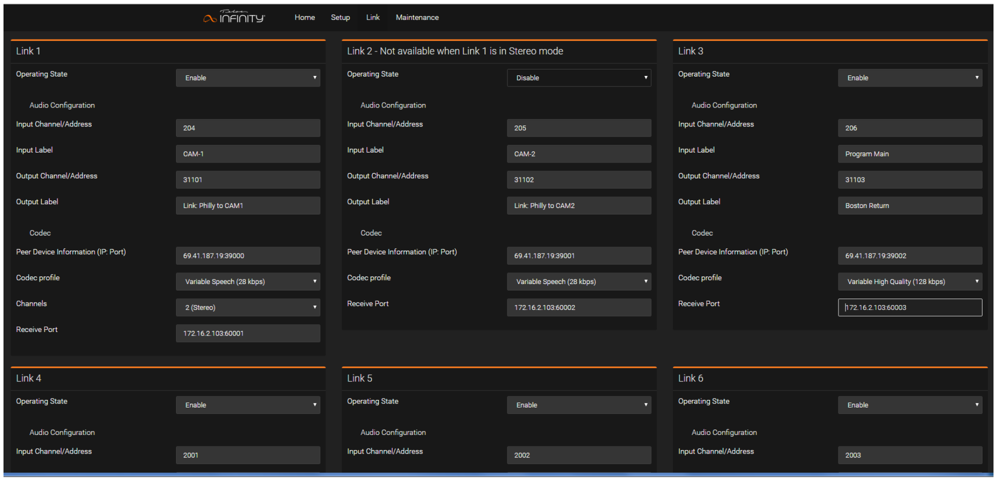
Additional features like System Zone and Scene support enable the user to define dedicated system views of specific components and create snapshots (Scenes) for different productions and applications. Tailored access to specific Zones and Scenes, as well as offline configuration modes are all protected by centrally managed user rights administration, ensuring that operators at all levels have restricted access to the resources that they need. Zones and Scenes can be edited, cloned and recalled both in 'Live' and Offline modes.

Critical for use within the wider Audio over IP ecosystem, Dashboard Advanced adds a tool that enables direct connection of third-party AES67 streams (including SAP for Dante and future NMOS modes).

Telos Infinity users have the option to purchase either single or dual server versions of Dashboard Advanced, each including Systems Administrator login licenses. Further logins for additional clients can be purchased. There is no limit to the number of simultaneous users who can be connected to Dashboard Advanced.

- Optimized user experience
- Infinity Dashboard Basic application for smaller systems (single license, online control only)
- Infinity Dashboard Advanced total system control and management (multi-user server application)
- Online and offline configuration (Dashboard Advanced)
- Create unlimited Groups, Partylines and IFBs
- Drag-and-drop system configuration
- Manages all Telos Infinity and Telos Alliance AoIP devices
- Livewire+ AES67 source and destination management (virtual matrix ports)
- GPIO / Logic configuration and management
- Constantly updating system database backup
- Native AES67 input/output support for all Livewire+ and non-Livewire+ streams (including SAP discovery and advertisement)
- Drag-and-drop Intercom Device copy and cloning of Panel and Beltpack configurations
- System Zones enable user-restricted or bespoke system viewports
- System Scenes enable multiple 'snapshots' for offline creation, save and live recall of show specific configurations

Telos Infinity Link



Telos Infinity Link allows Infinity systems in remote locations to connect by using embedded, versatile, speech-optimized OPUS codecs to convert WAN-side VoIP (including the Internet) to LAN-side Livewire+ AES67.

Telos Infinity Link is available in the form of software licenses that can be added to any host Infinity Panel or Beltpack hardware, or by way of a dedicated higher-density 1RU hardware gateway.

Infinity Link for Panels is available for Infinity Intercom Master Panel, Infinity Expansion Panel, and Infinity Desktop Panel by way of either the Link 4 or Link 8 options, offering four or eight bi-directional codecs respectively.

Infinity Link for Beltpack adds two bi-directional codecs via the Link 2 option.

Infinity Link Gateway supports either eight or sixteen bi-directional codecs running on a dedicated 1RU hardware platform, making it ideal for higher-density applications. Infinity Link Gateway features dual redundant internal power supplies and dual Gigabit Ethernet ports.

Once connected, the resulting Livewire+ AES67 streams from the remote locations appear as sources and destinations within the Telos Infinity Dashboard software and behave just like local networked audio sources, providing the ultimate in flexibility.

Telos® Z/IP ONE IP Codec

The IP Codec that Drops Jaws. Not Audio.

OVERVIEW

Z/IP ONE is a 1 RU rack-mount IP codec for remote broadcasting. It's a single-space rack unit perfect for studios, TOCs and remote kits. Z/IP ONE Includes a full range of codecs including AAC-ELD, AAC-HE, AAC-LD, MPEG 4 AAC, MPEG 2 AAC, MPEG Layer 2, G.711, G.722 codecs, plus linear audio and optional aptX® Enhanced coding. Z/IP ONE supports SIP 2.0 protocol and conforms to N/ACIP standards; it also works with VoIP devices and connects to compatible SIP PBXs. A full complement of I/O, including Livewire® AoIP, analog and AES/EBU, is standard.

FEATURES

- Works with wired and wireless IP connections including WiFi, WLAN and UMTS/EVDO networks.
- Exclusive Agile Connection Technology (ACT) automatically senses network conditions and adapts codec performance to provide the best possible audio.
- Largest choice of high-performance codecs: AAC-ELD, AAC-HE, AAC-LD, MPEG Layer-2, MPEG-4 AAC-LC, MPEG-2 AAC-LC, G.711, G.722, and linear PCM. Enhanced aptX coding optional.
- Dual Ethernet ports for separate streaming and control, LAN for local control with Livewire audio and GPIO; separate WAN for secure connection to wide area networks.
- Livewire, analog and AES/EBU I/O standard.
- Easy browser setup via built-in Web server.
- “Push Mode” for one-way network transmission.
- “Multiple Push Mode” for audio distribution to multiple destinations.
- Distributed Z/IP Server directory service, with multiple geolocations, lets you easily connect to other Z/IP ONE devices without the need for an IP address and also provides sophisticated NAT traversal support.
- Transparent, time-aligned RS-232 channel for remote control or metadata, e.g., RDS.
- Time-aligned 8-bit parallel GPIO port for signaling and control.
- Slim 1RU form factor is perfect for studio racks, remote kits or road cases.

IN DEPTH

Z/IP ONE: It's The Zephyr® for IP

These days, you can get broadband Internet just about everywhere, which makes it ideal for live remotes. But public Internet can also be erratic. You could be lucky enough to get a good connection, but it might deteriorate during your broadcast. What to do? Cross your fingers and hope for the best? Or reduce your bit rate, sacrificing audio quality in hopes of making it through your show?

With Z/IP ONE (the "Z/IP" stands for "Zephyr IP"), you don't have to compromise audio quality for a solid connection. Z/IP ONE helps you get the best possible quality from public IP networks and mobile data services — even from connections behind NATs and firewalls. Telos® collaborated with Fraunhofer (the developers of MP3 and many AAC breakthroughs) to develop a unique coding control algorithm that adapts to changing Internet conditions on the fly, helping you maintain quality and stability.

We call it ACT, short for Agile Connection Technology, and only Telos has it. Using ACT to sense and adapt to the condition of your IP link, Z/IP ONE delivers superb performance on real-world networks. ACT adapts dynamically to minimize the effects of packet loss and jitter. When the bits are flowing smoothly, you'll benefit from the lowest possible delay and the highest possible fidelity. If congestion starts to occur, Z/IP ONE automatically lowers bit rate and increases buffer length to keep audio flowing at maximum quality. You'll get reliable audio even when network conditions are unpredictable — and you won't need to fiddle with settings or codecs to do it.

To make certain your remote broadcast has excellent audio quality even when IP connections are not-so-excellent, Telos engineers employed AAC-ELD (Advanced Audio Coding-Enhanced Low Delay) to produce excellent fidelity at low bitrates, with nearly inaudible loss concealment and very little delay. Standard high-performance codecs are a part of the Z/IP ONE toolkit as well, such as AAC-HE, AAC-LD, MPEG4 AAC-LC, MPEG2 AAC-LC, G.711, G.722 and even linear PCM. And if apt-X is part of your codec cache, you can add it to your Z/IP ONE as a small extra-cost option.

It's from Telos, so of course you expect that Z/IP ONE will be easy to set up and easy to use. And it is — the front panel controls are intuitive and friendly, and the built-in Web server makes short work of configuration or remote control via any PC with a Web browser. And our exclusive worldwide Z/IP Server service, free to Z/IP owners, lets you easily get around NATs and network firewalls for fast connections to your favorite locations. For even more flexibility, Z/IP ONE can connect to third-party apps such as LUCI LIVE and LUCI LIVE Lite to receive on-the-go reports from smartphones and tablets.

Around back, you'll find analog and AES3 XLR ins and outs, a Livewire LAN port for quick connection to Axia® networks, and a separate WAN port for safe connection to "the outside world."

Z/IP ONE is also wireless-capable and connects natively to IP networks via Wi-Fi. A parallel port is provided for end-to-end, time-aligned GPIO contact closures; Z/IP ONE can also transport RS-232 serial data (using an inexpensive USB-to-Serial adaptor cable), synchronized with audio delivery — useful for RDS/RBDS data, as well as other serial data, at up to 9600 bps.

SPECIFICATIONS

Conformance and Compatibility

- Conforms to N/ACIP (Open) Standards. Fully supports Session Initiation Protocol 2.0 (SIP). Compatible with TCP, UDP, DNS, Zephyr Xstream®, Uncompressed PCM and other Internet Protocols.

Codecs

- SIP: G.711, G.722, MPEG Layer2, MPEG AAC, MPEG 4 AAC LC, MPEG 2 AAC LC, Linear PCM, MPEG AAC-Enhanced Low Delay (ELD), High Efficiency AAC.
- Optional: apt-X Enhanced ® from CSR.

Connections

Analog

- 1x Stereo input, presented on two XLR-F connections
- 1x Stereo output, presented on two XLR-M connections

Livewire

- 1x 100BASE-T connections, presented on RJ-45

AES/EBU

- 1x Stereo Input, presented on one XLR-F connection
- 1x Stereo Output, presented on one XLR-M connection

Network

- 2x 100BASE-T connections, presented on RJ-45 (1x LAN, 1x WAN)

USB

- 2x A-Type, Female

Parallel (GPIO)

- 1x DB25, Male

Audio:

Analog Line Inputs:

- Input Impedance: 6K Ohm differential
- Input Range: Selectable, Line (+4 dBu nominal), Microphone (-50dBu nominal)
- Selectable Phantom power

Analog Line Outputs:

- Output Impedance: 50 Ohm differential
- Output Clipping: +22dBu

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: 48, 44.1 or 32 kHz, or "sync to input" (auto-matches rate and clock from AES/EBU input)
- 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: +/- 1dB 25– 20 kHz

Headroom

- 18 dB

Dynamic Range

- 87dB Unweighted
- 90 dB "A" Weighted

Total Harmonic Distortion + Noise

- < 0.03% @ +12dBu, 1 kHz Sine

Crosstalk Isolation

- > 80 dB

Power Supply AC Input

- Auto-ranging supply, 90VAC to 240VAC, 50 Hz to 60 Hz, IEC receptacle, internal fuse
- Power consumption: 14.2 Watts

Operating Temperatures

- 0-40 degrees C (32-104 degrees F), stirred air

Dimensions

- 19" (48.3 cm) standard rack mounting front panel
- 1.75" (4.5 cm) height, 6.5" (16.51 cm) depth
- Shipping Weight: 8 lbs. (3.62 kg)
- Shipping Dimensions: 24" x 14" x 6" (61 cm x 35.6 cm x 15.25 cm)

Telos® Zephyr Xstream® ISDN Codec

The Best Way To Hear From There®

OVERVIEW

Zephyr Xstream is the world's leading ISDN codec, compatible with the widest variety of third party codecs. Coding choices include MPEG4-AAC and AAC-LD, Layer 2, Layer 3 & G.722 coding for full-duplex stereo operation of up to 20 kHz audio on a single ISDN line; broadcast quality mono audio at 15 kHz or 20 kHz is possible on a single ISDN "B" channel or other 56/64 kbps channel. All Xstream models feature professional balanced analog inputs/outputs, as well as Livewire® I/O for quick connection to Axia® networks; AES/EBU I/O is standard on rackmount model. An ISDN TA with integral NT1 provides worldwide ISDN compatibility without software changes. Remote Control is possible over RS-232 or Ethernet. Available in rack-mount version and portable version which incorporates a 4-channel stereo DSP mixer with selectable digital processing by Omnia®.

FEATURES

- Ethernet ports for remote control via LAN or WAN, and for connection to your Livewire AoIP networks. Bring audio from any codec anywhere in the world directly to your Axia network.
- Auto Receive mode quickly determines the correct coding algorithm for incoming audio streams.
- MPEG AAC (Advanced Audio Coding) permits true CD-quality stereo transmission with a connection speed of just 128 kbps.
- Low-Delay MPEG AAC-LD coding delivers crystal-clear audio quality and greatly reduced encoding delay for smooth, natural bi-directional remotes.
- MPEG Layer-3 coding for compatibility with the largest number of third-party codecs. When using MPEG Layer-3, a unique Dual Receive mode allows reception of independent audio streams arriving from two distant ISDN lines – great for bilingual broadcasts.
- Exclusive Error Concealment technology prevents occasional network glitches from being heard.
- RS-232 and 8-input, 8-output parallel ports provide ancillary data and bi-directional contact closures.
- Hand-in-glove operation with companion Zephyr Xport® portable codec for reception of 15kHz audio using a POTS field connection.
- V.35/X21 option allows connection to serial synchronous data equipment, for use with dedicated lines, Switched 56 circuits or satellite services.
- N/ACIP compliant for compatibility with the widest range of ISDN codecs.
- Convenient ISDN Voice Telephone Mode allows placement of standard G.711 phone calls from your Zephyr Xstream.

IN DEPTH

Advanced caller management and superior sound

The Telos Zephyr is the best-loved broadcast codec in the world, and for good reason: Zephyr saves you time and money. A Zephyr Xstream at your studio becomes a "universal codec," connecting with every popular ISDN codec for full-duplex, 20kHz stereo audio. And in the field, Zephyr Xstream is a powerful remote tool, with intuitive step-by-step operation, context sensitive help, and a simple user interface that eases operation for non-technical personnel. Zephyr pioneered the concept of the ISDN codec — which is why you'll find more Zephyrs in studios and remote kits around the world than any other codec.

Zephyr Xstream has a huge range of standard MPEG coding options, which include MPEG Layer-3 and MPEG AAC for indistinguishable source-from-input audio at only 128 kbps. Zephyr Xstream can also be used for LAN and WAN IP streaming of MP3 or AAC over properly managed networks. Zephyr's AAC coding includes error concealment to inaudibly recover from a lost packet or two, and an adjustable packet jitter buffer allows you to easily accommodate different networks.

There are two Zephyr Xstream models tailored to fit your needs: The standard rack-mount Xstream, and the portable Xstream MXP, a ruggedized portable version with built-in DSP mixer and Phantom microphone power to help reduce field equipment inventory and setup times. All have standard Analog and Livewire I/O, and a built-in terminal adapter with integral NT1 for worldwide compatibility without software changes. The studio version also has standard AES/EBU I/O, and the portable Xstream features a DSP-based AGC/limiter with Omnia audio processing and selectable presets for music & voice.

Ready for your rack

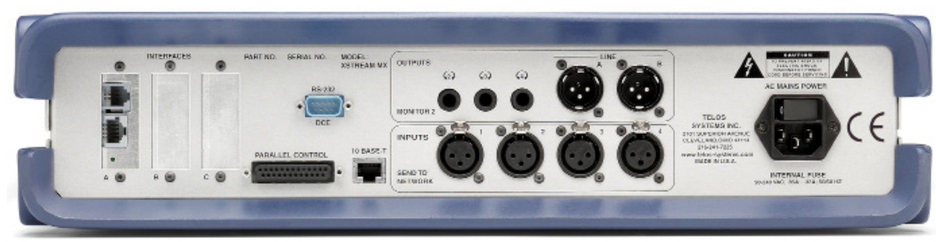
The Zephyr Xstream rackmount version is a full-featured ISDN transceiver that's become the "gold standard" for ISDN codecs around the world. In fact, Zephyr may be the most popular digital broadcast product ever, with tens of thousands in service at radio and TV stations, recording and voice-over studios everywhere. The front panel has a backlit display screen with a friendly, logical control structure and Fast Access Menu Keys to quickly call up system information and settings. Other controls include meters for send-and receive-audio levels, a dialing keypad, and a front-panel headphone jack with level control for convenient direct monitoring. Zephyr Xstream also includes an Auto-Dial function with storage for up to 100 stored Preset Numbers — each with its own bitrate and transmit/receive settings. 30 Location settings permit quick recall of ISDN line and audio settings for your most commonly visited remote sites.

On the rear panel you'll find standard balanced analog I/O with auxiliary unbalanced inputs, presented on combination XLR/TRS connections. AES/EBU I/O is standard, with a separate AES sync input. Remote control is supported using either the 100BASE-T Ethernet port or the RS-232 port. There are also eight bi-directional inputs/outputs for end-to-end contact closure emulation. The built-in ISDN Terminal Adapter is compatible with telcos worldwide. And Zephyr Xstream also has native Livewire support, for one-cable connection to Axia AoIP networks.

Ready for the road

The portable Zephyr Xstream MXP has all the features found in the rackmount Zephyr Xstream, plus a digital four-channel stereo mixer with two local mixes, inside a road-ready case designed for the rigors of on-the-go broadcasting. The rugged shock-resistant case helps prevent bumps and bruises, and an included flip-up metal stand lets you tilt the unit up for the best viewing angle on desktops or whatever handy surface you're broadcasting from. The alpha-numeric dial pad also generates DTMF tones for navigation through voice menu systems.

Zephyr Xstream MXP's four-input stereo DSP mixer directly feeds its codec section; mic/line switchable inputs with pan also include a preset mic limiter & AGC processing by Omnia; inputs 1 & 2 have switchable 48-volt Phantom power. There's a front panel headphone jack for Local Mix 1 that monitors either Send or Receive audio, or a mixture of the two. Local Mix 2 has separate front-panel controls for the three rear-panel headphone jacks, plus a pair of balanced XLR line outputs to feed guest phones or monitors.



Around back, the Xstream MXP differs from its rackmount brother by its inclusion of 4 input connections, plus headphone/monitor outputs. Both Zephyr Xstream models are fan-free for silent operation.

SPECIFICATIONS

General

- Full duplex, high-fidelity codec using MPEG-2 AAC, MPEG-4 AAC-LD MPEG-2 Layer-3, MPEG-2 Layer-2, AACPlus, and G.722; fully compliant with international standards.

Frequency Response

- 20 - 20kHz @ 48kHz fs (+0/-1dB, swept sine tone procedure)
- AAC all modes except Stereo 64: 20-19,800Hz at 48kHz fs., 20-15,000Hz at 32kHz fs.
- AAC Stereo 64: 20-10,000Hz at 48kHz fs., 20-7,000Hz at 32kHz fs
- AACPlus mono (for use reception from the Xport): 20-15,000 Hz 48kHz fs
- Layer 3 all modes: 20-16,000Hz at 48kHz fs., 20-15,000Hz at 32kHz fs
- Layer 2 mono, dual mono: 20-7.8kHz/9.8kHz
- Layer 2 mono 20-8.6 kHz at 24 kHz fs.
- Layer 2 joint stereo: 20-20,000Hz at 48kHz fs. 20-15 kHz at 32kHz fs
- G.722: 20-7,500Hz.

THD+N

- Audio loopback, 48kHz fs, analog I/O, input at 1kHz +20dBu: 0.004%

Dynamic Range

- A Weighting, AAC, Layer-3 or 2 end-to-end: 101dB typical

Send Input

- Active balanced with RF protection.

Zephyr Xstream:

- LINE: Settable to -11 or +4dBu (or -15 to 0 dBu) nominal level
- Clip point: 18 dB above chosen nominal level.
- Impedance: > 10K Ohms (x2)
- Connector: XLR female/quarter-inch TRS combo connector.

Zephyr Xstream MX and MXP:

- LINE: -11 or +4dBu nominal level (switchable).
- MIC: Accepts -65 to -24 dBu in 2 ranges (switchable). Mic impedance </= 1000 Ohms
- Clip point: 15 dB above chosen nominal level.
- Impedance: Line > 10K Ohms (x2)
- Connector: XLR female/quarter-inch TRS combo connector.

Limiter

Zephyr Xstream MXP:

- Internal DSP-based AGC/limiter with Omnia® audio processing. Includes presets for music & voice, selectable per channel.

Zephy Xstream:

- Analog soft-clipper prevents A/D converter overload without loss of dynamic range.

Line Bit Rates (ISDN)

- 56 or 64kbps per channel, front panel selectable.

Bit Rates (V.35/X.21)

- 56, 64, 112 (imuxed), 128 (imuxed), 96, 128, 256, 384 kbps front panel selectable.

Receive Output

- Active differential.
- Level: Front panel selectable for -10 or +4dBu, nominal.
- Impedance: < 33 Ohms (x2)
- XLR male

AES/EBU Digital I/O (rackmount version only)

- Sample rates supported: 32, 44.1 and 48kHz
- Rate conversion: Input and output independently selectable. Can be bypassed.
- Input clock: From external source or Telco clock.
- Output clock: From transmission sample rate, external source, or AES/EBU input.

Inverse Multiplex/Demultiplex

Internal channel splitting/combining of two network channels for stereo modes.

- AAC: Telos Zephyr protocol.
- AAC-LD: Telos Zephyr protocol.
- Layer-3: FHG/Telos Zephyr (Buchta) protocol.
- Layer-2: CCS CDQ™ protocol compatible.

Optional V.35/X.21 Direct Digital Interface

- Two ports, both V.35/X.21. Automatically selected when the appropriate cable is connected.

ISDN Interface

- Compatible with National ISDN, AT&T 5ESS custom, Northern Telecom DMS-100 custom, Siemens EWSD, INS 64(Japan) and EURO-ISDN (ETS-300). Compatibility and approval pending in some countries; contact Telos for current status.

LAN Interface

- 100BASE-T Ethernet port using RJ-45 connector. Full Duplex Supports TCP/IP (HTML, Telnet and FTP).

ISDN Voice Telephone Mode

- Two channels using G.711 standard, μ-Law or A-Law. 300–3,400Hz. DTMF signaling provided (CCITT standard).
- Remote Control and Ancillary Data
- RS-232 9-pin D-Sub female (DCE): Asynchronous; 8 data, no parity, 1/2 stop, 2400-57,600 bps.
- 100BASE-T Ethernet port using RJ-45-style connector using Telnet or web browser (HTML).

Regulatory

North America: FCC and CE tested and compliant, power supply is UL approved.

Europe: Complies with the European Union Directive 2002/95/EC on the restriction of the use of certain hazardous substances in electrical and electronic equipment (RoHS), as amended by Commission Decisions 2005/618/EC, 2005/717/ EC, 2005/747/EC (RoHS Directive), and WEEE.

Telos® Zephyr® iPort PLUS Multi-CODEC Gateway

16 Stereo Codecs in a Livewire® Gateway

OVERVIEW

Zephyr iPort PLUS is a networked multi-codec gateway that enables transport of multiple channels of stereo audio across any QoS-enabled IP network, including T1 and T3 connections and private WANs with MPLS – perfect for large-scale distribution of audio to single or multiple locations.

Zephyr iPort PLUS is the workhorse of codecs, configurable as eight stereo bi-directional MPEG codecs, or for encode / decode of up to 16 uni-directional stereo streams. Zephyr iPort PLUS connects to Axia® IP-Audio networks using a single CAT-6 cable for all I/O. Don't have a Livewire network yet? Pair Zephyr iPort PLUS with Telos Alliance® xNode audio interfaces for use as a standalone multiple-stream codec.

Coding algorithms include AAC, AAC-LD, HE-AAC (plus v2), MP2, MP3, linear, and optional aptX® Enhanced*. Bit rates range from 24 to 320 kbps for MPEG codecs, plus standard fixed rates for aptX and linear to over 2 Mbps. In addition, iPort offers dual, parallel-path end-to-end streaming for ultra-reliability and redundancy. Up to 20 unidirectional GPIO contact closures per codec are available in several modes to allow considerable flexibility of control. End to end GPIO is supported for each codec. For network operators, a unique Content Delay feature allows independent local storage and scheduled delayed playout of any or all coded audio channels for up to six hours.

FEATURES

- Distributes multiple channels of coded audio between broadcast facilities over QoS-enabled IP links.
- Configurable as a CODEC with 8 bi-directional channels, each with GPIO and PAD — or, as a 16-channel stereo encoder or 16-channel stereo decoder.
- 8 PCM Stereo channels are available for use simultaneously alongside CODEC channels, (dependent upon available bandwidth).
- Can also deliver streaming audio channels for Internet transmission via SHOUTcast, Steamcast or compatible stream replication server.
- Wide choice of genuine Fraunhofer codecs, including Standard AAC, high-efficiency AAC-HE (aacPlus), AAC-HEv2, low-delay AAC-LD, and MP3, with a choice of bit rates from 24 kbps to 320 kbps, definable per stream.
- Optional aptX Enhanced audio coding may be ordered at time of purchase or added later, as desired.
- When used as part of a Livewire network, allows audio from remote facilities to be used as if they were local sources, with associated logic and control.
- Eight 5-input Virtual Mixer (VMIX) channels each allow combining and mixing of up to 5 networked Livewire audio streams on a single channel.
- Eight Virtual Mode (VMODE) channels allow audio to be split into left/right channels, summed L+R, and more, prior to encoding and transmission.
- Content Delay option enables delayed playout of any or all selected receive audio channels, along with time-synchronized ancillary data, for up to six hours. Each playback delay time is independently configurable on a per-channel basis, making Zephyr iPort PLUS ideal for network operators, program distribution networks, or delayed playout of received audio at network-affiliated stations.
- Remote control/configuration via any computer with a standard Web browser.
- Separate LAN and WAN ports help ensure network security.
- Fanless, convection-cooled DSP-powered platform with dual-redundant, auto-switching powersupplies for maximum uptime. Power supply modules are field-replaceable in minutes.
- Optional Time Zone Delay upgrade allows the iPort to delay the playout of material for one, two, or several hours, facilitating its use over different time zones.
- AES67 Support for increased interoperability available in vMode.

IN DEPTH

Powerful, advanced program distribution and facility connection.

If your facility is like most, rack space is a precious commodity. That's why Telos® engineers invented Zephyr iPort PLUS, a sophisticated multiple-CODEC device that saves you money and rack space by housing 16 broadcast-quality stereo codecs in one 2RU device.

A pair of Zephyr iPort PLUS on each end of a QoS-controlled IP link can send and receive 8 channels of bi-directional stereo MPEG audio. Or, use iPort as a one-way "push" link to encode and deliver 16 channels of broadcast-quality one-way audio to a remote destination. With its ability to send multiple MPEG channels over IP connections, Zephyr iPort PLUS is perfect for audio transmission over VPNs, satellite links, Ethernet radio systems, and Telco or ISP-provided QoS-controlled IP services such as T1, T3 or OC-3 links.

You can use iPort for studio-to-transmitter links, network distribution systems, multi-channel links to remote studios. Install a QoS-enabled IP link between two studios with Axia Livewire networks, put an iPort at each end, and you can pass audio and GPIO between locations as if they were just next door. Paired with an appropriate server, you can even use Zephyr iPort PLUS to generate multiple channels of MP3 or AAC coded audio for Internet streaming, broadcasting to mobile phones, and audio distribution systems.

Finally, Zephyr iPort PLUS' exclusive Content Delay option (available at extra cost) adds hardware and software that enables delayed playout of select received audio channels. Associated GPO and ancillary data is likewise delayed and synchronized with audio. Delay any or all coded audio channels up to six hours; each channel's delay time is independently configurable.

The Zephyr iPort PLUS rear panel is remarkably simple, thanks to the use of Livewire AoIP I/O. A single Ethernet cable is all that's needed for all inputs, outputs, GPIO and remote control. Uncompressed 24-bit/48kHz audio goes in from your network via Ethernet; compressed MPEG streams go out on the same cable — eliminating expensive, space-consuming converters and connectors. Or, use the separate WAN connection to send your audio over an outside network.

If you don't have an Axia network yet, that's no problem — just pair Zephyr iPort PLUS with Telos VX analog or digital audio interfaces, or Telos Alliance xNodes, to make a standalone high-density audio codec package.

Zephyr iPort PLUS streams sound fantastic, thanks to our long-standing relationship with Fraunhofer IIS, the inventor of MP3 and co-inventor of AAC. The encoding algorithms inside iPort are genuine FhG, not no-name knockoffs. A full range of state-of-the-art codec types and bitrates are supported; the highest-quality implementations possible, run by a powerful Intel floating-point processor. Choose AAC-LD for delay-sensitive applications, AAC-HE and AAC-HEv2 for low bitrate requirements, standard MPEG AAC for best quality and resilience to packet loss at higher bitrates, MP3 and MP2 for legacy applications.

You'd expect all this to cost a lot, but it doesn't: we built Zephyr iPort PLUS on a single industrial motherboard, rather than the usual "multiple DSP cards in a frame" approach. Together with the Livewire-only audio interface, Zephyr iPort PLUS delivers more power than a legacy cardframe design, at only a fraction of the cost.

SPECIFICATIONS

Audio

Zephyr iPort PLUS has no native audio I/O, operating on streams provided by attached Livewire audio devices. All audio specifications below are representative of Axia Livewire audio interfaces.

Analog Line Inputs

- Input Impedance: >40 k ohms, balanced
- Nominal Input Range: Selectable, +4 dBor -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 dB(-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
- AES67 Support for increased interoperability available in vMode

Frequency Response

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

Network

- 1 LAN port, 1 WAN port; 100/1000BASE-T Ethernet interfaces.

Codecs

- Standard AAC, high-efficiency AAC-HE (aacPlus), AAC-HEv2, low-delay AAC-LD, MP3, MP2. Optional: apt-X® from CSR.

Power

- Dual-redundant internal auto-ranging power supplies, 90 – 132 / 187 – 264 VAC, 50Hz/60Hz.
- Power consumption: 100 Watts.

Regulatory

North America: FCC and CE tested and compliant, power supply is UL approved.

Europe: Complies with the European Union Directive 2002/95/EC on the restriction of the use of certain hazardous substances in electrical and electronic equipment (RoHS), as amended by Commission Decisions 2005/618/EC, 2005/717/ EC, 2005/747/EC (RoHS Directive), and WEEE.

Telos VX® Prime+

Big Performance for Small Facilities

OVERVIEW

Telos VX® talk-show systems are the world's first true VoIP-based broadcast phone systems and have been proven to deliver the power of VoIP to the broadcast studio like no other. The Telos VX Prime+, with built-in support for AES67, is the next evolution of Telos VX VoIP phone systems in a powerful new 1RU hardware unit. Additionally, support for the G.722 voice codec ensures the highest quality calls from supported mobile devices. With capacity of 8 fixed hybrids/faders, VX Prime+ is ideal for facilities with 2 to 4 studios. (For larger facilities, check out VX Enterprise with up to 120-hybrid capacity.)

AES67 support brings a new level of compatibility and flexibility to VX phone systems. Support for AES67 gives broadcasters the flexibility of integrating VX Prime+ into *any* AES67 environment, in addition to our own Axia® Livewire® network. With plug-and-play connectivity, you can network multiple channels of audio with any manufacturer's AES67-compliant hardware. Beyond AES67, Livewire users have the added convenience and power of networking control (GPIO), advertising/discovery, and program associated data throughout the network.

Using VoIP, VX Prime+ gives you remarkable-sounding on-air phone calls with no 'gotchas'. It uses standard SIP protocol that works with many VoIP PBX systems and SIP Telco to take advantage of low-cost and high-reliability service offerings. VX Prime+ can also connect to traditional telco lines via Asterisk PBX systems, which can be customized for specific facility requirements.

VX Prime+ gives you incredible operational power, flexible, adaptable workflows, and superior audio quality, while making it easier than ever for talent to have complete mastery of their callers. With VX Prime+, the world's leading broadcast phone system is now available to those with smaller budgets, offering Big Performance for Small Facilities.

Why VX Prime+?

Cost-Efficient Way to Upgrade to IP

- Lower-capacity alternative to VX Enterprise Broadcast System
- Potentially save thousands monthly on expensive ISDN/POTS lines
- Ideal for smaller studios (2-4 studios) & smaller budgets

Audio Quality

- Native support of G.722 "HD Voice" codec
- Fifth-generation Telos Adaptive Digital Hybrid on every line for cleanest, clearest caller audio
- Smart AGC ensures consistent caller audio levels
- Digital Dynamic EQ (DDEQ) by Omnia® adjust EQ automatically to ensure call-to-call consistency and the best intelligibility

Simple Setup

- Connects to your Axia Livewire—or other AES67 network—with a single Ethernet cable
- Non-Livewire studios can use Telos Alliance Mixed Signal xNode for audio and GPIO connectivity to studio consoles
- Provides phone hybrids for each of your studios without need for any additional wiring or physical audio connections

Flexible Use

- No restriction to the number of SIP lines or phone numbers that can come into the system
- 8 fixed hybrid/faders (not expandable)

Support

- Industry-leading 24/7 support
- Free Xscreen Lite call-screening software from Broadcast Bionics

FEATURES

Features At A Glance

- A true VoIP telephone system designed and built specifically for broadcasting; VX Prime+ is ideal for small to medium facilities with 2 to 4 studios.
- Includes support for AES67, giving broadcasters added flexibility of integrating VX Prime+ into any AES67 network, in addition to our own Axia Livewire network.
- SIP call-handling throughout—no internal conversion to analog call handling like some other so-called "VoIP" systems.
- Standards-based SIP interface integrates with Asterisk open-source SIP phone servers and most VoIP-based PBX systems to allow transfers and common telco services for business and studio phones.
- Standard Ethernet backbone provides a common transport path for both studio audio and telecom needs, resulting in cost savings and a simplified studio infrastructure.
- System capacity of 8 hybrids. Each call placed on the air receives a dedicated hybrid for unmatched clarity and superior conferencing.
- Native Livewire integration—one connection integrates caller audio, program-on-hold, mix-minus, and logic directly into Axia AoIP consoles and networks.
- Connect VX systems to any third-party radio console or other broadcast equipment using available Telos Alliance Mixed Signal, AES/EBU, and GPIO xNodes. xNodes feature 48 kHz sampling rate and studio-grade 24-bit A/D converters with 256x oversampling.
- Powerful dynamic line management enables instant reallocation of call-in lines to studios requiring increased capacity.
- VSet phone controllers with full-color LCD displays and Telos Status Symbols present producers and talent with a rich graphical information display. Each VSet features its own address book and call log.
- The "Drop-in" Vset Call Controller™ modules can integrate VX phone control directly into your mixing consoles.
- XScreen Lite screening software included.
- Clear, clean caller audio from 5th-generation Telos Adaptive Hybrid technology, including Digital Dynamic EQ, AGC, adjustable caller ducking, and send- and receive-audio dynamics processing by Omnia.
- Support for G.722 codec enables high-fidelity phone calls from iPhone and Android SIP softphones using an SIP server.
- Wideband acoustic echo cancellation from Fraunhofer completely eliminates open-speaker feedback.

- Works with POTS, T1/E1, ISDN and SIP Trunking telco services for maximum flexibility and cost savings, via Asterisk servers.*

  * Due to the wide variation in how traditional phone service can be delivered, and the complexities that can be involved in converting those services to SIP, we really want to talk with you about your system design before you order. Telos has VX System engineers standing by to help you draw up a configuration that will ensure your VX purchase will perform to your expectations when using traditional POTS and ISDN lines.

IN DEPTH

This Is Where It All Started

Steve Church founded Telos Systems® in 1985. As both a talk-show host and radio group Technical Director, Steve was only too familiar with the frustrations of "bad phones" and even less responsive equipment manufacturers, so he set about eliminating the technical problems that plagued radio call-in segments. In 1984, he invented the Telos 10, the first DSP-based telephone-to-broadcast interface system—allowing radio stations to significantly improve the technical quality of call-in segments. The overwhelming response to Steve's economical and technically elegant solution to a nagging problem provided the spark from which Telos was born.

A lot's happened since then. Telos pioneered the use of MPEG Layer 3 coding in the revolutionary Zephyr ISDN codec. We produced the first hardware MP3 streaming encoder for broadcast. We developed the world's first "whole-plant" broadcast phone system. We invented the IP-networked radio console, and then integrated broadcast phones into that network via Ethernet.

Telos has grown steadily since our initial production run of 25 Telos 10 units in 1985! With tens of thousands of systems in the field, it's now is hard to find a broadcast facility in the world without at least one piece of our gear. Our organization, now called the Telos Alliance, includes the Omnia®, Axia®, 25-Seven®, Minnetonka Audio®, and Linear Acoustic®, and our R&D department— the largest research team in broadcasting— continues to develop innovative audio products for radio and television broadcasting, telephony, and the Internet.

VX Prime+ Broadcast VoIP Phone System

Telos VX marries the flexibility and capabilities of IP networks to the remarkable power of today's digital signal processing, and brings the benefits to broadcast facilities. With a VX system, you can move and share lines between studios at the touch of a button. VX Prime+ is an 8-hybrid system that brings VoIP flexibility to medium and small facilities without breaking the bank.

VX systems are naturally flexible, naturally powerful. Your broadcasts benefit from superb, crystal-clear caller audio while callers hear clean, intelligible audio from your console. VX systems are surprisingly cost-effective, even when deployed in single-station facilities.

Why VoIP For Broadcast?

VoIP has taken the business world by storm, increasing the flexibility of office phone systems and PBXs while simultaneously lowering maintenance and equipment costs. In fact, most Fortune 500 companies have replaced their old PBX systems with VoIP for just these reasons.

VoIP is a natural for broadcasters, interconnecting the phone system with audio interfaces, phone sets, console controllers, and PCs running screening software by way of efficient, low-cost Ethernet. Using VoIP, you can finally share phone lines among multiple studios and route caller audio anywhere in your facility, easily, and instantly. Got a hot talk-show that suddenly needs more lines in a certain studio? Just a few keystrokes at a computer and you're ready...no delays, no cables to pull. VX systems can even interconnect with your business office's VoIP PBX to allow easy call transfers.

Reduced Cost. Increased Flexibility.

The use of sophisticated, modern IP networking for Telos VX Prime+ allows rich communication between devices. For example, caller information entered by a producer is displayed on the studio phone's color LCD. Caller audio is available on studio PCs for easy recording. Operators at mixing consoles can directly control line switching without diverting their attention from the board. The result? Talk shows that run like clockwork, sound better, and flow without errors.

This standards-based VoIP architecture helps you save money, too, by widening your choices in telco providers. Most carriers now offer VoIP services using the SIP protocol, which can deliver substantial savings to stations that need any number of lines. (You can also connect to traditional T-1/PRI, POTS or ISDN phone lines using open-source Asterisk-based phone servers.)

But VX systems don't stop at providing the benefits of VoIP—they also carry the broadcast-phone technology expertise of Telos.

Every incoming line has its own 5th-generation Telos Adaptive Digital Hybrid, our most advanced ever—packed full of technology engineered to extract the cleanest, clearest caller audio from just about any phone line, even cellular calls. Multiple lines can be conferenced with superior clarity and fidelity. Smart AGC ensures consistent caller audio levels. And calls from mobile callers using SIP clients on their smartphones benefit from native support for the G.722 "HD Voice" codec, improving caller speech quality and intelligibility.

Added Flexibility to Connect to Livewire or Any Other AES67 Network

AES67 support brings a new level of compatibility and flexibility to VX phone systems. Support for AES67 gives broadcasters the flexibility of integrating VX Prime+ into *any* AES67 environment, not just our own Axia® Livewire® network. With plug-and-play connectivity, you can network multiple channels of audio with any manufacturer's AES67-compliant hardware. Beyond AES67, Livewire users have the added convenience and power of networking control (GPIO), advertising/discovery, and program associated data throughout the network.

VX System Components

VX Prime+ Engine

The heart of any VX system is the Engine. The fixed-capacity VX Prime+ system is powered by a 1RU rack-mount Engine with enormous processing power. In fact, the VX Prime+ Engine provides all the call control and audio processing needed for your entire on-air phone system.

With VX Prime+, you are equipped with 8 high-performance VoIP hybrids, to support multiple lines of concurrent on-air phones for two to four studios (depending on configuration).

Each VX Prime+ Engine features two Gigabit Ethernet ports, a high-density, cost-effective interface to both telephone lines and studio audio via proven Livewire Audio over IP (AoIP). VX systems are web-based, so remote control and configuration are easy—engineers can work from any place they can get online.

Call workflow for VX users is sophisticated and flexible. Lines may be readily shared among studios; the Web interface allows easy assignment of lines to "shows," which can then be selected by users on VSet phone controllers and console drop-in modules. Each studio may be configured with its own Program-on-Hold as well.

The processing power of the VX Engine provides sophisticated DSP hybrids for every line, allowing multiple calls to be conferenced and aired simultaneously with excellent quality. The hybrids are equipped with a rich processing toolbox to make caller audio sound its best, no matter what kind of line or phone the caller uses.

Caller audio benefits from Smart AGC coupled with famous Telos three-band adaptive Digital Dynamic EQ and a three-band adaptive spectral processor. Call ducking and host override are part of the VX audio toolkit as well.

You'll notice that there are no audio I/O or telco ports on VX Engines themselves. That's because they're meant for fast connection to Livewire AoIP systems; using Livewire, all I/O is handled via Ethernet. The

Livewire network supports a wide variety of peripherals such as Axia audio consoles, VSet phones, PC-based screener applications, console-integrated controllers, and more. SIP servers and telecom providers connect through a dedicated WAN Ethernet jack for routing simplicity and easy maintenance. For traditional phone services, VX works seamlessly with open-source Asterisk SIP servers, and most SIP PBXs. Telos VX experts speak fluent Asterisk, and are ready to assist you in specifying and configuring an installation to suit your studio's requirements. VX also works with standard telco gateways to connect to T1/E1, ISDN, and POTS providers. And, if you already have a VoIP-based PBX or SIP endpoint service, VX systems can work with those as well.

VSet12

The Telos VSet12 phone is beautifully designed, with a friendly LCD color display that uses exclusive Status Symbols to let talent know what's going on instantly. VSet12 works with up to 12 phone lines; the info-rich display provides caller ID for each line, along with time ringing-in or on-hold, and even screener comments from the screening software applications.

VSet12 gives talent unprecedented flexibility. You can map groups of lines to a single fader, making it simple to take a queue of calls to air sequentially. One-touch controls let talent step through queued calls, "busy out" incoming lines, lock calls on-air to prevent unintentional disconnection of a VIP. Telos-exclusive "Next Call" key speeds workflow for producers, screeners, and talent. But because VX systems provide a hybrid per line, much more functionality is unlocked: You can now spread multiple calls over a number of faders, using one for each call so that operators can control each line's level individually. You can hard-assign individual lines to fixed faders, such as for VIP calls. A built-in address book and call history log round out VSet12's features.

VSet6

VSet6 is a 6-line phone controller for VX systems. Like the VSet12, it has a bright, attractive LCD color display with Status Symbols that feed talent instant information about line and caller status and controls that enable talent to step through queued calls, busy incoming lines, lock calls on-air, activate the dump button on a profanity delay, and more.

VSet Phone Controls

TRANSFER

ADDRESS BOOK

BUSY ALL

MISSED CALLS

ON AIR

AUTO ANSWER

DEVICE CONFIG.

The LCD displays deliver detailed line status, caller information, caller ID, time ringing-in or on-hold, and even comments entered in screening software applications. Shown above are a few of the attractive, instantly-understandable Status Symbols that help talent run tight, mistake-free shows.
Each VSet phone has its own web server for easy remote configuration and software upgrades, and flexible power options include PoE (Power over Ethernet) from a Telos-approved PoE switch or Axia xSwitch, or an in-line power injector.

On-Console Control

Whether calls are live or pre-recorded, interviews or audience participation, one thing's certain: phone segments are an integral part of today's fast-paced radio. But up to now, the phone system was separate from the on-air console; audio was shared, but little else. Wouldn't it be great if talent could take control of phones without ever having to divert their attention from the board? They can: the Axia Console Controller provides the ideal way to integrate broadcast phones into the on-air console—the control center of every studio.

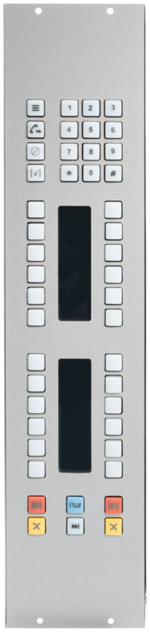
There are plenty of advantages to melding phones with consoles. Like ease of installation: IP-Audio consoles with built-in phone controllers don't need any additional wires or connections. Their control signaling, caller audio, and backfeeds ride on the network connection that's already there. Bringing caller audio into the IP-Audio domain makes it routable like any other audio source. You can even dynamically conference multiple lines using just a single fader.

VX systems connect directly to Axia Fusion, Element, iQ and Radius mixing consoles using Livewire+™ AES67 IP-Audio to eliminate the cost and complexity of old-style inputs, outputs, and mix-minuses. And now, VX Systems have the added flexibility of AES67 support. Multiple phone lines— each with a dedicated hybrid—can automatically map to individual console faders for complete control of caller audio. And users enjoy seamless console integration, with phone controls right on the board so that talent can dial, answer, screen, and drop calls without ever diverting their attention from the console. Information about line and caller status can be displayed right on the console as well.

Axia Call Controller

For VX clients with an Axia Fusion mixing console. The Axia Call Controller module puts control of VX telephone systems right into the console. The two-fader telephone control module features an integrated Telos Call Controller with renowned Status Symbols visual call management. These include Transfer, Drop, and Block All keys plus the Telos-exclusive "Next Call" key that allows fast airing of pre-screened calls. The rotary Options Control knobs can be programmed to trim source or fader gain when turned, and alphanumeric channel displays give complete information on source assignment, channel options, and more.

VSet Call Controller

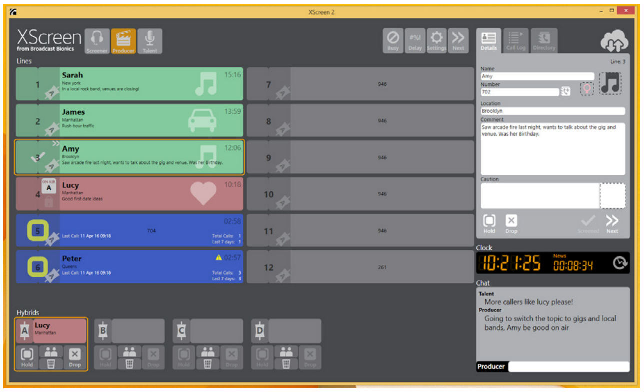


Want a VX system, but don't have an Axia mixing console? No problem — Telos provides VSet Console Controller electronics packages, which may be fitted to your console using panels supplied by your OEM console provider or preferred third-party fabricator. Like the VSet12 phone set, the VSet Console Controller provides visual line-status indicators and fast-take keys for selection and control of up to 12 callers, along with standard controls such as Take, Drop, Hold and Busy keys, and the Telos-exclusive "Next Call" key to speed workflow for producers, screeners, and talent. There's also a built-in keypad for on-console dialing of outgoing numbers.

VSet Desktop Controller

The Telos VSet Desktop Controller with visual line-status indicators provides selection and control of up to 12 callers. Includes standard controls to allow fast, error-free talent operation, including Take, Drop, Hold, and Busy keys. Telos exclusive "Next Call" key speeds workflow for producers, and talent; built-in keypad allows on-hybrid dialing of outgoing numbers. VSet Desktop Controller works in producer mode only. Make and answer calls using VSet6, VSet12, or on hybrid with VSet Desktop Controller.

Broadcast Bionics XScreen Lite Call-Screening Software Included



XScreen Lite software comes with every VX Prime+ purchase and provides Unlimited Lite users, dial, hold, hang up, screened hold and next, conference control, dump mode, lock call, VSet control, telephone number, location, name, point & disposition, chat, clock, and call log (6 hours only) functionality. Please download your XScreen software from www.xscreen2.com.

VX Prime+ Interfaces

Telos Alliance xNode Audio Interfaces

Telos Alliance xNodes let you connect VX Prime+ to any non-networked radio console or other broadcast equipment, using standard AES/EBU interfaces. A GPIO Logic xNode provides control logic where needed. To cover all your bases, the Telos Alliance Mixed Signal xNode provides one mic/line analog input (switchable); two analog line inputs (dedicated); three analog line outputs; one AES3 input, one AES3 output, and two GPIO ports, each with five opto-isolated ins and outs.

The Telos Alliance AES/EBU audio xNode 4 AES/EBU inputs and 4 AES/EBU outputs. Left and right input signals may be split and routed independently as mono signals. Stunning performance specs include 48 kHz sampling rate, 126dB of dynamic range, and <0.0003% THD.

Each Telos Alliance GPIO logic xNode interface provides six general-purpose logic ports each with five opto-isolated inputs and five outputs. A logic port can be associated with any audio input or output and routes control data transparently along with the audio.

Telos Alliance xSwitch Zero-Configuration Ethernet Switch

xSwitch is the world's only zero-configuration Ethernet switch optimized for Livewire IP-Audio applications. Fast setup requires only IP address assignment via front-panel OLED display or Axia iProbe software. Features 8 10/100MBit Ethernet ports — 4 with Power-over-Ethernet to power Axia xNodes, Telos VSet phones, and other networked devices compatible with the IEEE 802.1af PoE standard. Noiseless and fan-free, xSwitch can be conveniently placed adjacent to your audio devices, rack-mounted using included hardware, or wall-mounted (with an accessory kit available separately).

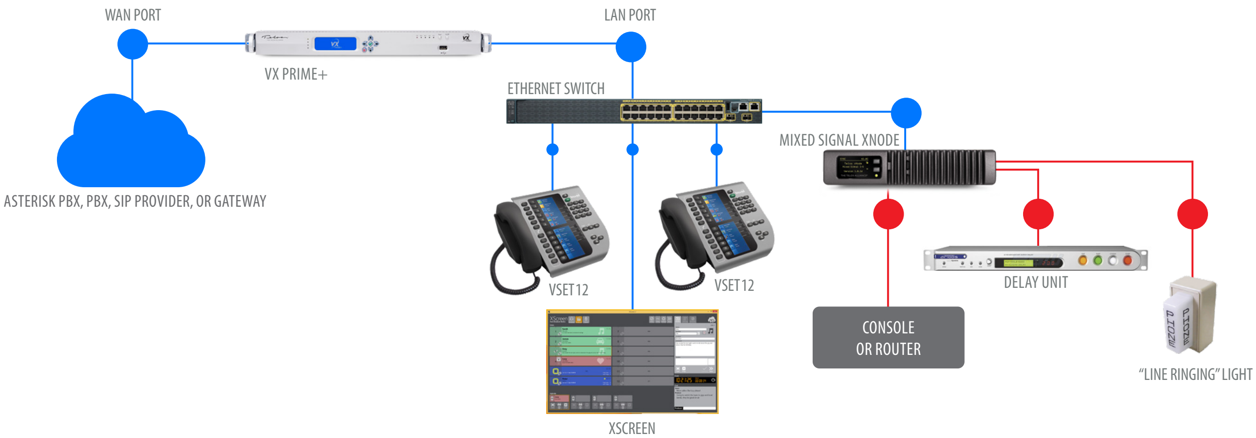
The Power of IP Realized

With VX Prime+, there's no need for the maze of discrete cables once required by multi-line talk show systems. All VX components are linked with standard Ethernet, so a single CAT-5 cable provides:

- Connection to the telco interface
- Line switching commands
- Data communication between the VX Prime+ and VSet12 phones
- Transport of caller audio to mixing consoles
- Return of mix-minus and program-on-hold audio to the caller
- Data messages (such as call notes and IM) between producer and talent
- Livewire audio for the recording of calls
- Transfer of recorded call files from the producer to the studio

Now... how many discrete cables does that save you from having to wire up?

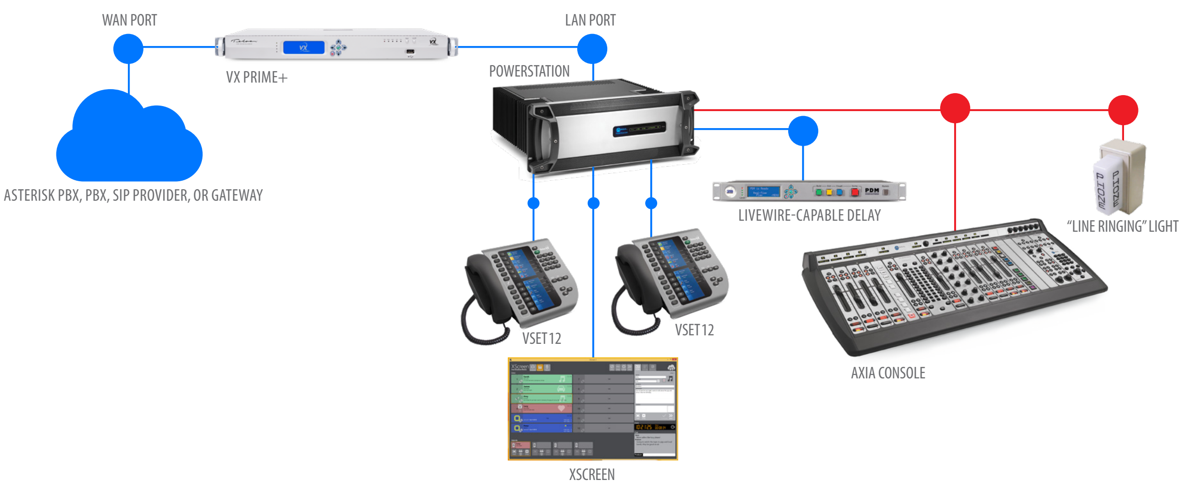
Hook It Up Your Way: Non-Axia Installation Diagram



Got an Axia Livewire AoIP studio network? Great! Your new VX Prime+ phone system will plug right into it. It's the seamless integration of studio phones, mixing consoles, and routing network you've dreamt about.

Don't have IP-Audio networking yet? Not to worry... VX will work with all console brands, networked or not, using Telos Alliance xNodes, and the VX Call Controller drop-in controller for your console. Telos VX systems are "facility wide" broadcast phone products. That means multiple studios, multiple stations, multiple shows—with minimal hardware requirements. Telco is delivered via IP from your SIP PBX, or through a dedicated IP circuit using SIP trunking. POTS, ISDN, or T1 phone service can be brought in using an open-source Asterisk server or a standalone gateway device. Once connected, all line and audio connectivity flows via Ethernet. The diagram above shows a typical studio with an analog mixer, using a Telos Alliance Mixed Signal xNode to connect to the console and other broadcast equipment.

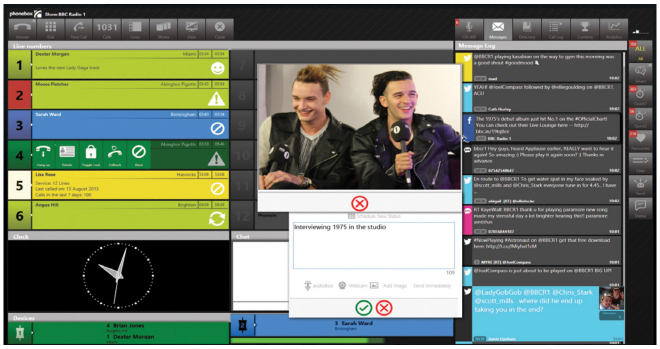
Hook It Up Your Way: Axia Installation Diagram



Installing a VX phone system in facilities already powered by Axia Livewire+™ AES67 networks requires even less time and hardware than described above. The audio inputs and outputs used and produced by VX are Livewire real-time audio channels and travel over the Axia AoIP network just like the rest of your audio. Axia console GPIO ports can be used for "phone ringing" tallies or remote control of profanity delay units.

VX Gives You Options

Broadcast Bionics



Broadcast Bionics offers PhoneBOX VX, a tailored-for-VX version of their original PhoneBOX software. PhoneBOX VX gives VX users an amazing amount of information and a high level of control over the VX system. There's prize management, call editing, and recording, sophisticated visual talkback, including a drag-and-drop database your show's calls, plus a rich phonebook and visual warnings, tied to Caller ID, for persistent or nuisance callers.

Find out more from www.phoneboxvx.com .

NeoSoft

NeoSoft offers NeoScreener, a call management solution that interfaces Telos NX12, NX6, IQ6, VX, HX6, 2x12 and 2101 systems, allowing for line control and database lookup using caller ID. The solution can interface to NeoWinners which is NeoGroupe's contest management software. It is designed for radio and television stations that need to manage their flow of incoming phone calls.

NeoScreener also handles external inputs, like SMS, Website, iPhone. Database driven, it enhances the phone-call workflow. With NeoScreener, call screeners can easily welcome calls and present them to the Talent on a specific display. Visit www.neogroupe.com to learn more.

Arctic Palm CS Call Management

The CS Call Management package provides producers and talent with the tools to capture and control callers while staying in touch with each other in a single Caller Control window. Designed for the VX VOIP systems, both local and remote users are in constant communication.

For more information, visit www.arcticpalm.com/CSScreener.htm.

SPECIFICATIONS

System

- Maximum number of simultaneous calls on-air, VX Prime+: 8 (more with conferencing)
- Maximum number of SIP numbers, VX Prime+: 96
- One rack unit - 1.75"H x 19"W x 15.5"D (44 x 483 x 394 mm)

Audio Performance (Node)

Analog Line Inputs

- Input Impedance: >40 k ohms, balanced
- Nominal Level 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) h Signal Format: AES-3 (AES/EBU)
- AES-3 Input Compliance: 24-bit with selectable sample rate conversion, 32 kHz to 96kHz input sample rate capable.
- AES-3 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
- Latency <3 ms, mic in to monitor out, including network and processor loop

Frequency Response

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

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

Total Harmonic Distortion + Noise

- 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, Stereo Separation And CMRR

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

VX Prime+ Engine

IP/Ethernet Connections

- One 1 Gigabit Ethernet via RJ-45 LAN connection (livewire)
- One 1 Gigabit Ethernet via RJ-45 WAN Connection (SIP provider)

Processing Functions

- All processing is performed at 32-bit floating-point resolution.
- Send AGC/limiter
- Send filter
- Gated Receive AGC
- Receive filter
- Receive dynamic EQ (3 band)
- Ducker
- Sample rate converter

Power Supply AC Input

- Hot-swap capable dual-redundant internal auto-ranging power supplies. 90 – 132 / 187 – 264 VAC, 50Hz/60Hz. IEC receptacle, internal fuse.
- Power consumption: 100 Watts

Operating Temperatures

- -10 degree C to +40 degree C, <90% humidity, no condensation

Dimensions and Weight

- Rackmount, 2RU
- 3.5 inches x 17 inches x 15 inches
- 10 pounds

Studio Audio Connections

- Via Livewire Ethernet. Each selectable group and fixed line has a send and receive input/output.
- Each studio may be configured with its own Program-on-Hold input.
- Livewire-equipped studios take audio directly from the network.
- Telos Alliance xNodes are available for professional-level analog and AES3 connection breakouts for clients without Livewire AoIP networking.
- VX Prime+ supports AES67 connectivity.

Telco Connections

- Audio: standard RTP. Codecs: G.711u-Law and A-Law, and G.722.
- Control: standard SIP endpoints, ISDN PRI/T-1, ISDN BRI and POTS may be supported with the appropriate interfaces using an Asterisk Open source PBX.

Telos VX® Enterprise

The Whole-Plant Broadcast Talkshow System

OVERVIEW

Telos VX® is the world's first VoIP (Voice over IP) talkshow system — a broadcast phone system that's so powerful, it can run all the on-air phones for your entire plant. Telos VX Enterprise™, with built-in support for AES67, is the next evolution of Telos VX VoIP phone system in a powerful new 1RU hardware unit. Additionally, support for the G.722 voice codec ensures the highest quality calls from supported mobile devices. With capacity expandable to up to 120 hybrids/faders, VX Enterprise is ideal for medium to large facilities and can grow with your station over time. (For smaller facilities, check out VX Prime+ with 8-hybrid capacity.)

AES67 support brings a new level of compatibility and flexibility to VX phone systems. Support for AES67 gives broadcasters the flexibility of integrating VX Enterprise into any AES67 environment, in addition to our own Axia® Livewire® network. With plug-and-play connectivity, you can network multiple channels of audio with any manufacturer's AES67-compliant hardware. Beyond AES67, Livewire users have the added convenience and power of networking control (GPIO), advertising/discovery, and program associated data throughout the network.

Using VoIP, VX Enterprise gives you remarkable-sounding on-air phone calls with no 'gotchas'. It weds modern networking to the remarkable power of digital signal processing. VX Enterprise uses Ethernet as its connection backbone, significantly cutting the cost of phone system installation, maintenance, and cabling. It uses standard SIP protocol that works with many VoIP PBX systems and SIP Telco to take advantage of low-cost and high-reliability service offerings. VX Enterprise can also connect to traditional telco lines via Asterisk PBX systems, which can be customized for specific facility requirements.

Don't have an IP-Audio network yet? Optional Telos Alliance xNodes, like the Telos Alliance Mixed Signal Node, break out audio into analog and digital formats, along with GPIO logic commands. And with informative VSet phones, talent finds it easier than ever to take control of their callers, moving and sharing lines between studios at the touch of a button.

FEATURES

- VX is the world's first VoIP telephone system designed and built specifically for broadcasting.
- Includes support for AES67, giving broadcasters added flexibility of integrating VX Enterprise into any AES67 network, in addition to our own Axia Livewire network.
- Works directly with SIP endpoint telco or PBX services, and in conjunction with a PBX may support POTS, T1/E1, and ISDN BRI for maximum flexibility and cost savings.*
- Standards-based SIP/IP interface integrates with most VoIP-based PBX systems to allow transfers, line-sharing, Caller ID and common telco services for business and studio phones.
- Standard Ethernet backbone provides a common transport path for both studio audio and telecom needs, resulting in cost savings and a simplified studio infrastructure. Connection of up to 100 control devices (software or hardware) is possible.
- Modular, scalable system can be easily expanded and configured to manage a network of 60 or even more studios, each with a dedicated Program-On-Hold input—truly a "whole-plant" solution for on-air phones.
- Base system is licensed for up to 24 hybrids, may be expanded in license increments of 8 up to a total of 120 hybrids. A Telos system engineering consultation is required for any system configuration over 72 hybrids. Please contact us at vx-presales@telosalliance.com for assistance.
- Each call receives a dedicated hybrid for unmatched clarity and superior conferencing.
- Native Livewire integration: One connection integrates caller audio, program-on-hold, mix-minus, and logic directly into Axia AoIP consoles and networks.
- Connect VX Enterprise to any radio console or other broadcast equipment using available Telos Alliance AES/EBU, Mixed Signal, and GPIO xNodes. Audio interfaces feature 48 kHz sampling rate and studio-grade 24-bit A/D converters with 256x oversampling.

- Powerful dynamic line management enables instant reallocation of call-in lines to studios requiring increased capacity.
- VSet Call Controllers™ with full-color LCD displays and Telos Status Symbols™ present producers and talent with a rich graphical information display. Each VSet features its own address book and call log.
- Drop-in modules can integrate VX Enterprise phone control directly into your Axia mixing consoles.
- XScreen Lite Screening software included.
- Clear, clean caller audio from 5th-generation Telos Adaptive Hybrid technology, including Digital Dynamic EQ, AGC, adjustable caller ducking, and send- and receive-audio dynamics processing by Omnia®.
- Wideband acoustic echo cancellation from Fraunhofer completely eliminates open-speaker feedback.
- Support for G.722 codec enables high-fidelity phone calls from iPhone and Android SIP softphones using an SIP server.

* Due to the wide variation in how traditional phone service can be delivered, and the complexities that can be involved in converting those services to SIP, we really want to talk with you about your system design before you order. Telos has VX System engineers standing by to help you draw up a configuration that will ensure your VX purchase will perform to your expectations when using traditional POTS and ISDN lines.

IN DEPTH

VoIP for Broadcast. From Telos, Naturally.

VX is the world's first VoIP (Voice over IP) talkshow system, and VX Enterprise is the next evolution of this legendary system, now with built-in support for AES67 in a 1RU chassis. This whole-plant broadcast phone system is incredibly powerful, very flexible, and highly scalable.

VoIP has already taken the business world by storm, increasing the flexibility of office phone systems and PBXs while simultaneously lowering maintenance and equipment costs. In fact, most Fortune 500 companies have replaced their older PBX systems with VoIP for just these reasons. There's no reason broadcasters shouldn't take advantage of this cost-saving technology as well. In addition to cost savings from digital phone service provisioning, VX Enterprise significantly eases the cost of installation, maintenance, and cabling by using standard Ethernet as its data backbone.

As a result VX Enterprise is naturally scalable, capable of serving even the largest of facilities. There are major operational benefits as well. VX Enterprise combines the flexibility and economy of modern SIP networking with powerful digital signal and audio processing—making it easier than ever for talent to take control of their phone system. You can move and share lines between studios at the touch of a button. VX Enterprise is truly the future of broadcast phones.

Why VoIP for Broadcast?

VoIP is a natural for broadcasters. Using VoIP, you can interconnect the phone system CPU with audio interfaces, phone sets, console controllers, and PCs running screening software using efficient, low-cost Ethernet. You can finally share phone lines among multiple studios and route caller audio anywhere in your facility, easily and instantly. Got a hot talk show that suddenly needs more lines in a certain studio? Just a few keystrokes at a computer and you're ready—no delays, and no cables to pull. VX Enterprise can even connect with your business office's VoIP PBX to facilitate easy call transfers.

Of course, it's got to sound good. And it does, thanks to more than two decades of DSP hybrid technology developed by Telos. Every incoming line has its own 5th-generation digital hybrid, our most advanced ever, packed full of technology engineered to extract the cleanest, clearest caller audio from any phone line—even noisy cellular calls. Multiple lines can be conferenced with superior clarity and fidelity. Smart AGC ensures consistent caller audio levels. New Acoustic Echo Cancellation from FhG removes feedback and echo in open-speaker studio situations. And if you choose to use SIP Trunking telco services, calls from mobile handsets with SIP clients, HD capable telephone sets and PC apps will benefit from VX Enterprise's native support of the G.722 codec, instantly improving caller speech quality.

Since VX Enterprise uses Ethernet as its network backbone, it naturally plugs right into Axia IP-Audio networks, connecting multiple channels of audio and control using a single Ethernet cable. If you don't have an IP-Audio network yet, that's OK; Telos Alliance xNodes provide AES audio and GPIO connections that work with your existing studio equipment.

VX Enterprise Components

VX Enterprise

VX Enterprise 1RU rack-mount device is the heart of the system. It provides all the call control and audio processing needed for the system, and supports up to 120 active calls on-air simultaneously. Its two Gigabit Ethernet ports provide a cost-effective interface to both telephone lines and studio audio via proven Livewire AoIP. VX Enterprise is Web-based, so remote control and configuration are a snap — engineers can work with it from any place they can get online.

Call-processing is sophisticated and flexible. Lines may be readily shared among studios; the Web interface allows easy assignment of lines to "shows," which can then be selected by users on the studio controllers. Each studio can provide its own Program-on-Hold audio to callers.

Audio processing features also have taken a leap forward. The processing power of VX Enterprise allows multiple calls to be conferenced and aired simultaneously, with excellent quality. The hybrids are equipped with a rich processing toolbox to make caller audio sound its best, no matter what kind of line or phone the caller uses. Caller audio benefits from Smart AGC coupled with famous Telos three-band adaptive Digital Dynamic EQ and a three-band adaptive spectral processor. Send audio gets its own sweetening with an AGC/limiter and FhG's Acoustic Echo Cancellation technology that literally eliminates open-mic feedback. Call ducking and host override are part of the VX Enterprise toolkit as well, and talent can manage and customize their telephone settings and workflow using VX Enterprise Show Profiles to store and recall commonly used show configurations.

You'll notice that there are no audio I/O or phone jacks on VX Enterprise. All connections to the Engine are via the two Ethernet jacks that connect to your system's Ethernet switch to support a wide variety of peripherals: telephone lines, Livewire studio audio, VSet phones, console-integrated controllers, etc. If you have a VoIP-based PBX or SIP endpoint telco service, VX Enterprise uses standard SIP (Session Initiation Protocol).

The Coolest Broadcast Phone Controllers Ever

With decades of experience designing broadcast phone systems, it's no wonder broadcasters agree that Telos makes the industry's most powerful, most flexible system controllers. All VSet phones can be powered by PoE from a Telos-approved switch, a PoE port on an Axia console engine, or by using a power injector.

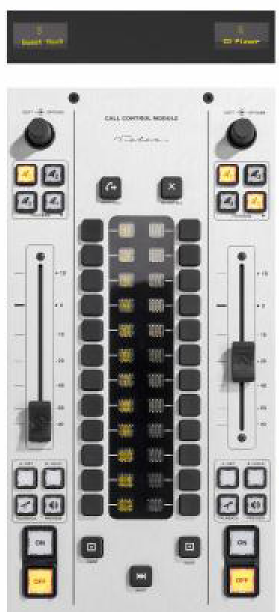
VSet12

The VSet12 phone controller is an IP-based phone set with two large, high-contrast color LCD panels that provide line status and caller information. VSet phones can work like a traditional Telos controller, with calls being selected, held, and dropped in the way to which operators have grown accustomed. But because the VX Enterprise system is so powerful, much more functionality is unlocked: You can now spread multiple calls over a number of faders, using one for each call so that operators can control each line's level individually. You can hard-assign individual lines to fixed faders, such as for VIP calls.

VSet6

VSet6 is a 6-line phone controller for VX Enterprise. Like the VSet12, it has a bright, attractive LCD color display with Status Symbols that feed talent instant information about line and caller status, and controls that enable talent to step through queued calls, busy incoming lines, lock calls on-air, start an external recording device, et cetera. Next Call functionality speeds workflow for producers, screeners, and talent. With all the control functions of the VSet12, it's great for smaller or secondary studios.

On-Console Control



Live calls or pre-recorded, interviews or audience participation, one thing's certain: Phone segments are an integral part of today's fast-paced radio. But up to now, the phone system was separate from the on-air console; audio was shared, but little else. Wouldn't it be great if talent could take control of phones without ever having to divert their attention from the board?

They can: the Axia Console Controller provides the ideal way to integrate broadcast phones into the on-air console—the control center of every studio. VX systems connect directly to Axia consoles using Livewire+™ AES67 IP-Audio to eliminate the cost and complexity of old-style inputs, outputs, and mix-minuses. And now, VX Systems have the added flexibility of AES67 support. Multiple phone lines—each with a dedicated hybrid—can automatically map to individual console faders for complete control of caller audio.Multiple phone lines—each with a dedicated hybrid—can automatically map to individual console faders for complete control of caller audio. And users enjoy seamless console integration, with phone controls right on the board so that talent can dial, answer, screen, and drop calls without ever diverting their attention from the console. Information about line and caller status can be displayed right on the console as well.

There are plenty of other advantages to melding phones with consoles. Like ease of installation: IP-Audio consoles with built-in phone controllers don't need any additional wires or connections. Their control signaling, caller audio, and backfeeds ride on the network connection that's already there. Bringing caller audio into the IP-Audio domain makes it routable like any other audio source. With the Virtual Mixers built into Axia consoles, you could even choose to dynamically conference multiple lines and control their gain with a single fader.

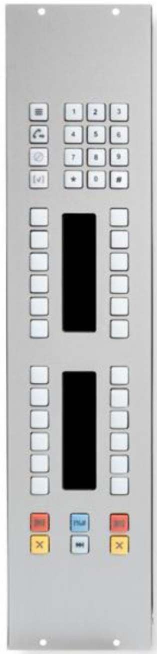
Telos Alliance xNode Audio Interfaces

Telos Alliance xNodes let you connect VX Entrprise to any non-networked radio console or other broadcast equipment, using standard AES/EBU interfaces. A GPIO Logic xNode provides control logic where needed. To cover all your bases, the Telos Alliance Mixed Signal xNode provides one mic/line analog input (switchable); two analog line inputs (dedicated); three analog line outputs; one AES3 input, one AES3 output, and two GPIO ports, each with five opto-isolated ins and outs.

The Telos Alliance AES/EBU audio xNode 4 AES/EBU inputs and 4 AES/EBU outputs. Left and right input signals may be split and routed independently as mono signals. Stunning performance specs include 48 kHz sampling rate, 126dB of dynamic range, and <0.0003% THD.

Each Telos Alliance GPIO logic xNode interface provides six general-purpose logic ports each with five opto-isolated inputs and five outputs. A logic port can be associated with any audio input or output and routes control data transparently along with the audio.

VSet Call Controller

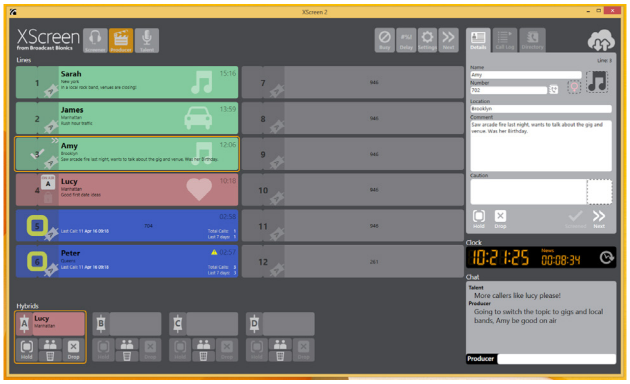


Want a VX Enterprise system, but don't have an Axia mixing console? No problem—Telos provides VSet Console Controller electronics packages, which may be fitted to your console using panels supplied by your OEM console provider or preferred third-party fabricator. Like the VSet12 phoneset, the VSet Console Controller provides visual line-status indicators and fast-take keys for selection and control of up to 12 callers, along with standard controls such as Take, Drop, Hold and Busy keys, and the Telos-exclusive "Next Call" key to speed workflow for producers, screeners, and talent. There's also a built-in keypad for on-console dialing of outgoing numbers.

VSet Desktop Controller

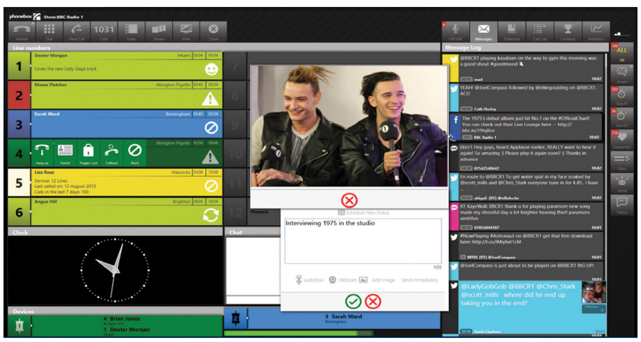
The Telos VSet Desktop Controller with visual line-status indicators provides selection and control of up to 12 callers. Includes standard controls to allow fast, error-free talent operation, including Take, Drop, Hold, and Busy keys. Telos exclusive "Next Call" key speeds workflow for producers, and talent; built-in keypad allows on-hybrid dialing of outgoing numbers. VSet Desktop Controller works in producer mode only. Make and answer calls using VSet6, VSet12, or on hybrid with VSet Desktop Controller.

Broadcast Bionics XScreen Lite Call-Screening Software Included



XScreen Lite software comes with every VX Enterprise purchase and provides Unlimited Lite users, dial, hold, hang up, screened hold and next, conference control, dump mode, lock call, VSet control, telephone number, location, name, point & disposition, chat, clock, and call log (6 hours only) functionality. Please download your XScreen software from www.xscreen2.com.

Broadcast Bionics PhoneBOX VX



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For more information, visit www.arcticpalm.com/CSScreener.htm.

SPECIFICATIONS

General

- Telos 5th-generation Adaptive Digital Hybrids
- Maximum number of hybrids: 120, when used with a-Law or u-Law codecs for VoIP lines. (Higher-quality codecs, such as G.722, consume more system resources and result in a decreased number of total lines available.) A Telos system engineering consultation is required for any system configuration over 72 hybrids. Please contact us at vx-presales@telosalliance.com for assistance.
- Maximum number of SIP numbers: unlimited
- Maximum active on-air calls: 120
- Maximum on-air calls on one fader: 12
- One rack unit - 1.75"H x 19"W x 15.5"D (44 x 483 x 394 mm)

Analog Inputs (with Telos Alliance xNode)

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

Analog Outputs (with Telos Alliance xNode)

- 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: AES-3 (AES/EBU)
- AES-3 Input Compliance: 24-bit with selectable sample rate conversion, 32 kHz to 96kHz input sample rate capable
- AES-3 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
- Latency <3 ms, mic in to monitor out, including network and processor loop

Frequency Response

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

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

Total Harmonic Distortion + Noise

- 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, Stereo Separation, and CMRR

- Analog Line channel to channel isolation: 90 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

VX Enterprise

IP/Ethernet Connections

- One 100/1000BASE-T Ethernet via RJ-45 LAN connection
- One 100/1000BASE-T Ethernet via RJ-45 WAN connection

Processing Functions

- All processing is performed at 32-bit floating-point resolution
- Send AGC/limiter
- Send filter
- Gated Receive AGC
- Receive filter
- Receive dynamic EQ
- Ducker
- Sample rate converter
- Line Echo Canceller (hybrid)
- Acoustic Echo Canceller

Power Supply AC Input

- Hot-swap capable dual-redundant internal auto-ranging power supplies. 90 – 132 / 187 – 264 VAC, 50Hz/60Hz. IEC receptacle, internal fuse.
- Power consumption: 150 Watts

Operating Temperatures

- -10 degree C to +40 degree C, <90% humidity, no condensation

Studio Audio Connections

- Via Livewire IP/Ethernet. Each selectable group and fixed line has a send and receive input/output
- Each studio has a Program-on-Hold input
- Each Acoustic Echo Canceller has two inputs (signal and reference) and one output
- Livewire+™ AES67 equipped studios may take and supply audio directly to/from the network. Telos Alliance xNodes are available for pro analog and AES3 breakout.
- VX Enterprise supports AES67 connectivity.

Telco Connections

- Audio: standard RTP. Codecs: g.711μ-Law and A-Law, and G.722.
- Control: standard SIP Endpoint, ISDN PRI/T-1, ISDN BRI and POTS may be supported with the appropriate interfaces using an Asterisk Open source PBX.

Regulatory

North America: FCC and CE tested and compliant, redundant power supplies are UL approved.

Europe: Complies with the European Union Directive 2002/95/EC on the restriction of the use of certain hazardous substances in electrical and electronic equipment (RoHS), as amended by Commission Decisions 2005/618/EC, 2005/717/ EC, 2005/747/EC (RoHS Directive), and WEEE.

Telos® Hx6 Six-Line POTS Talkshow System

Give Your Phones an Instant Upgrade

OVERVIEW

Hx6 is our most advanced six-line digital Talkshow system. It features two high-performance digital hybrids and includes our famous Digital Dynamic EQ, noise gate, caller ducking, and acoustic echo cancellation. Works with POTS analog phone lines. Single-cable Ethernet hookup via Axia® Livewire® I/O, or choice of analog or AES/EBU I/O with one input and one output per hybrid, and one Program On-Hold input. Includes complimentary XScreen Lite call-screening software from Broadcast Bionics.

FEATURES

- Six-line capacity; works with POTS (analog) phone lines.
- Our most advanced digital hybrids, with DSP algorithms optimized for superior performance with today's wide variety of far-end call types (VoIP, cell, POTS, app-based).
- Telos® DDEQ (Digital Dynamic EQ) and adjustable smart-level AGC ensure spectrally consistent audio from call to call — even on notoriously tough cellular calls.
- Excellent trans-hybrid loss of >55dB.
- Smooth, proven, symmetrical wide-range AGC by the audio processing experts at Omnia®.
- Studio adaptation and a subtle, inaudible pitch shifter to prevent feedback in open-speaker studio environments.
- A sophisticated caller override that improves performance and allows precision adjustment of the degree to which talent audio "ducks" the caller audio.
- Striking Telos VSet6 six-line phone controllers with large, colorful VGA LCD displays that provide intuitive operation and setup. Telos-exclusive Status Symbols provide producers and talent with animated, high-contrast icons that communicate line and caller status at a glance.
- Caller ID displayed on the VSet6 phoneset and the included XScreen Lite call-screening application.
- Livewire IP-Audio allows fast, one-cable integration with Axia networks, and provides Axia board operators with seamless, on-console control of multiple lines and hybrids. Standard Ethernet backbone provides a common transport path for both studio audio and telecom needs, resulting in cost savings and a simplified studio infrastructure.
- Choice of standard Analog I/O or optional, extra-cost AES/EBU I/O.
- Easy setup and configuration via Ethernet using any PC and your favorite Web browser.
- XScreen Lite call-screening software from Broadcast Bionics, provided at no cost.

IN DEPTH

Advanced caller management and superior sound

Say hello to Hx6, the most advanced six-line broadcast phone system Telos has ever made. Thanks to its Telos DSP hybrids and a full suite of audio processing capabilities, an Hx6 in your studio is like an instant audio upgrade for on-air phone calls — song requests, morning show phoners, or call-intensive talk shows.

Hx6 works with POTS phone lines, and comes equipped with two advanced telephone hybrids (each with its own independent AGC, noise gate, and caller override dynamics) for high-quality conferencing — the same advanced DSP technology used in the best-selling Telos Hx1 and Hx2 telephone hybrids.

The DSP toolkit in Hx6 is full-featured, to say the least. Telos Digital Dynamic EQ, our renowned adaptive 3-band processor, analyzes and adjusts received audio spectral characteristics so that calls sound smooth and consistent despite today's wide variety of phone sets and connection types. Adjustable Omnia smart-level AGC with noise gating provides spectrally consistent audio from call to call — even on notoriously tough cellular calls. A sophisticated caller override allows precision adjustment of the degree to which talent audio "ducks" the caller audio, and exclusive feedback reduction functions help eliminate open-speaker howl.

Like all Telos talkshow systems, the Hx6 front panel is simple and informative, with separate send and receive meters for each hybrid, a Program-On-Hold audio presence indicator, a high-resolution OLED display for setup, and navigation keys for quick adjustments.

Around back, you'll find audio I/O, GPIO, and Telco connections. Hx6 connects directly to 6 POTS lines. Separate analog or optional AES digital I/O is provided for each hybrid, as well as a Program-On-Hold input, GPIO connections for speaker muting, ring tallies, et cetera.

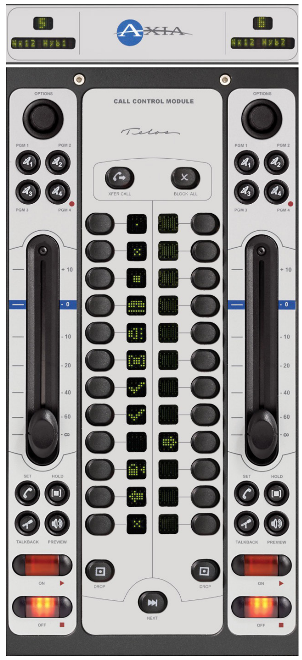
There's also an Ethernet port. This provides connection of as many as six Telos VSet phones, but that's not all: It's also an Axia Livewire port. Through that jack, Hx6 puts audio, hybrid control, and mix-minus for all six phone lines onto one single skinny CAT-5 cable. Livewire setup is simple: Plug it into your Axia network, do some fast web-based configuration, and your talent can control Hx6 directly from an Axia mixing console equipped with Call Controller modules. The Ethernet connection also allows for convenient remote setup and administration.

With all of these capabilities, you'd expect Hx6 to cost twice as much — but it doesn't. In fact, you can have an Hx6 for about what you'd pay for some other companies' "premium" systems.

Intuitive, easy-to-use controllers

This is the Telos VSet6 six-line phone controller, an IP-based phoneset with a large, high-contrast color LCD panel that provides line status and caller information. There's almost no learning curve; VSet phones work like traditional Telos controllers, with calls selected, held, and dropped in the way to which operators have grown accustomed. Exclusive animated Telos Status Symbol icons show line and caller status at a glance; easy VSet controls let talent manage incoming lines, lock calls on-air, start an external recording device, and take a queue of calls to air sequentially, for precise management of multi-call interviews or conferences. Next Call functionality speeds workflow for producers, screeners, and talent. The LCD display delivers detailed line status, caller information, caller ID, time ringing-in or on-hold, and even comments entered in the included Xscreen screening software. A built-in address book and call history log round out VSet6's features. And, just like the Hx6 itself, each VSet6 has its own web server for easy remote configuration and software upgrades.

Axia On-Console Control

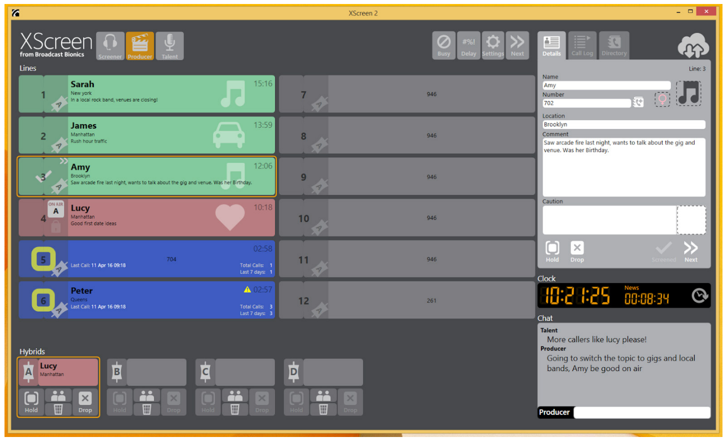


Hx6 works with any brand of broadcast console. But wouldn't it be great if talent could take control of phones without ever having to divert their attention from the board? Whether your shows consist of live calls or pre-recorded interviews, phone segments are usually fast-paced with little room for error. But traditionally, the phone system was separate from the on-air console, making it hard to use both together efficiently, leading engineers and talent to ask: "Why can't the console and the phone system work together?"

Now, they can. Hx6 can connect directly to Axia mixing consoles using Livewire IP-Audio to eliminate the cost and complexity of old-style inputs, outputs, and mix-minuses. IP-Audio networking technology provides the ideal way to integrate broadcast phones into the on-air console — the control center of every studio. Users enjoy seamless console integration, with phone controls right on the board so that talent can dial, answer, screen, and drop calls without ever diverting their attention from the console. Information about line and caller status can be displayed right on the console as well.

There are plenty of other advantages to melding phones with consoles. Like ease of installation: IP-Audio consoles with built-in phone controllers don't need any additional wires or connections. Their control signaling, caller audio, and backfeeds ride on the network connection that's already there. Bringing caller audio into the IP-Audio domain makes it routable like any other audio source. And since the console now communicates directly with the phone hybrid, mundane tasks such as mix-minus generation, starting recording devices, and playback of recorded off-air conversations can all be automated. All of which means faster, more precise phone segments — since operators' eyes never need to leave the console.

Broadcast Bionics XScreen Lite Call-Screening Software Included



XScreen Lite software comes with every Hx6 purchase and provides Unlimited Lite users, dial, hold, hang up, screened hold and next, conference control, dump mode, lock call, VSet control, telephone number, location, name, point & disposition, chat, clock, and call log (6 hours only) functionality. Please download your XScreen software from www.xscreen2.com.

SPECIFICATIONS

General

- Telos 3rd-generation Adaptive Digital Hybrids
- Telos Exclusive Feedback Reduction Functions
- Send-to-Caller Processing: High-pass Filter, Frequency Shifter, AGC/Limiter, Program-on-Hold AGC/ Limiter, Sample Rate Conversion (with AES option)
- Receive-From-Caller Processing: High-pass "Hum" Filter, Smart AGC / Platform Leveler, Noise Gate, Telos DDEQ (Digital Dynamic Equalization) 3-band Adaptive Spectral Processor, Sample Rate Conversion (with AES option)

Analog Inputs

- Send Analog Inputs: 2x
- Program-on-Hold Analog Inputs: 1x
- Connector: XLR Female, Pin 2 High (Active Balanced with Protection)
- Input Level: Adjustable from -7 to +8 dBu (nominal)
- Analog Clip Point: +21 dBu
- Impedance: Bridging, > 10K Ohms
- Analog-to-Digital Converter Resolution: 20 bits

Analog Outputs

- Receive Analog Outputs: 2x
- Connector: XLR Male, Pin 3 High
- Output Level: Adjustable from -7 to +8 dBu (nominal)
- Impedance: <50 ohms
- Digital-to-Analog Converter Resolution: 24 bits
- Headroom Before Clipping: 20 dB headroom above 4dBU nominal levels

Switching Matrix and Conferencing

- Audio Routing and Switch: All Digital
- Telephone Lines: 6
- Hybrids: 2
- Studio Inputs: 2
- Studio Outputs: 2
- Program-on-Hold: 1

Control Ports

Ethernet 100BASE-T

- Web server for configuration and software update
- Telnet for command line control and diagnostics
- Call Screening Interface server allows up to 8 instances of call-screening software to connect simultaneously

General purpose Input/Output

- 2x 15-pin D-sub with status outputs and control inputs

Control Interface

- Up to 12 attached controllers (any mix of VSet6 phones, Console Controllers or screening software) via Ethernet connection

Power Supply

- Type: Internal auto-ranging, 85–250 VAC auto-switching, 50–60 Hz
- Power consumption: 14.2 Watts

Analog Telephone Connectivity

- Universal interface for worldwide application
- Programmable loop current
- Programmable ring and disconnect signaling (loop drop or tone)
- Programmable Flash time
- Caller ID decoding using Bellcore 212 modem standard

Regulatory

North America: FCC and CE tested and compliant, power supply is UL approved.

Europe: Complies with the European Union Directive 2002/95/EC on the restriction of the use of certain hazardous substances in electrical and electronic equipment (RoHS), as amended by Commission Decisions 2005/618/EC, 2005/717/ EC, 2005/747/EC (RoHS Directive), and WEEE.

Telos® iQ6 Six-Line POTS Telco Gateway

Talkshow System for Axia® IP-Audio Networks

OVERVIEW

Telos® iQ6 is a six-line digital phone system designed specifically for use with Axia networked mixing consoles — that's why we call it a "Telco gateway." iQ6 acts as a portal for Axia systems, supplying caller audio, mix-minus, Program-On-Hold audio, and switching control for six POTS phone lines, using a single RJ-45 network connection.

iQ6 is built around the Telos Hx6, our most advanced six-line POTS Talkshow system. It features two high-performance digital hybrids and includes Telos' famous Digital Dynamic EQ, noise gate, caller ducking, and acoustic echo cancellation. Includes complimentary XScreen Lite call-screening software from Broadcast Bionics.

FEATURES

- Single-cable Ethernet connection to Axia IP-Audio networks transports caller audio, mix-minus, Program-On-Hold audio and hybrid switching control – no separate audio connections or contact closures to solder.
- Direct, on-console control of iQ6 operations, with add-on modules for popular Axia iQ and Axia Fusion® mixing consoles. Talent never needs to take their eyes off the control board; shows run smoother with less errors.
- Works with Telos VSet6 six-line phone controllers with large, colorful VGA LCD displays that provide intuitive operation and setup.
- Telos-exclusive Status Symbols on-console and phone controllers provide producers and talent with animated, high-contrast icons that communicate line and caller status at a glance.
- Six-line capacity; works with POTS (analog) phone lines.
- Our most advanced digital hybrids, with DSP algorithms optimized for superior performance with today's wide variety of far-end call types (VoIP, cell, POTS, app-based).
- Telos DDEQ (Digital Dynamic EQ) and adjustable smart-level AGC ensure spectrally consistent audio from call to call — even on notoriously tough cellular calls.
- Excellent trans-hybrid loss of >55dB.
- Smooth, proven, symmetrical wide-range AGC by the audio processing experts at Omnia®.
- Studio adaptation and a subtle, inaudible pitch shifter to prevent feedback in open-speaker studio environments.
- A sophisticated caller override that improves performance and allows precision adjustment of the degree to which talent audio "ducks" the caller audio.
- iQ6 versions matched to your choice of analog POTS phone lines.
- Caller ID.
- Easy setup and configuration via Ethernet using any PC and your favorite Web browser.
- XScreen Lite call-screening software from Broadcast Bionics, provided at no cost.

IN DEPTH

Six lines of crystal-clear caller audio. One easy RJ-45 connection.

A multi-line phone system that connects to your console with just one cable? Smooth, detailed caller audio — even from cellular callers? That's iQ6, the no-hassle Telco gateway for Axia mixing consoles.

iQ6 plugs right into Livewire® AoIP network networks, saving money and time by eliminating the cost and labor of old-fashioned discrete I/O, cabling, and soldered connectors. All connections to and from the iQ6 system — receive and send audio, hybrid control, mix-minus for six phone lines, even connections to VSet6 phone controllers and included PC-based call-screening software — travel over a single skinny CAT-5 cable. Setup is simple: Plug it into your Axia network, do some fast web-based configuration, and voila! you're taking calls.

The photo above shows a complete Axia iQ console system, with QOR.32 console engine, iQ6 Talkshow system, and iQ control surface with onboard phone controller. Control of both iQ6 hybrids and Status Symbols information icons are right on the mixer's surface.

You can also pair iQ6 with Telos Vset phones and their full-color, high-contrast display screens. iQ6 is extremely flexible: You can connect up to 12 control devices at once — phones, PCs, or console controllers — to take charge from nearly anywhere. Separate Send and Receive level meters for each hybrid are conveniently located right on the front panel for extra monitoring confidence.

How does iQ6 sound? Like a Telos, of course! Inside, two of our most advanced hybrids handle up to six POTS phone lines. Those hybrids are equipped with Digital Dynamic EQ and adjustable smart-level, symmetrical wide-range AGC by Omnia to keep callers sounding clean, clear, and spectrally consistent call after call. An adjustable caller override lets you dial-in just the right amount of call ducking. Our subtle, inaudible pitch-shifter helps prevent open-speaker feedback. And conference linking lets you set up high-quality conferencing between callers at the touch of a button — no external equipment needed.

The iQ6 front panel is simple and informative, with separate send and receive meters for each hybrid, a Program-On-Hold audio presence indicator, a high-resolution OLED display for setup, and navigation keys for quick front-panel adjustments.

The back panel likely looks much different from any other phone system you've seen. There are no discrete audio I/O, GPIO, PoH or output connections — the RJ45 connection to your Axia Livewire network handles all of that. Like its brother, the Telos Hx6, iQ6 connects directly to 6 POTS lines.

On-Console Control

iQ6 connects directly to Axia mixing consoles using Livewire IP-Audio to eliminate the cost and complexity of old-style inputs, outputs, and mix-minuses. IP-Audio networking technology provides the ideal way to integrate broadcast phones into the on-air console. Users enjoy seamless console integration, with phone controls right on the board so that talent can dial, answer, screen, and drop calls without ever diverting their attention from the console. Information about line and caller status can be displayed right on the console as well, with Telos Status Symbols icons that communicate line and caller status at a glance — ensuring that phone segments are always smooth and error-free.

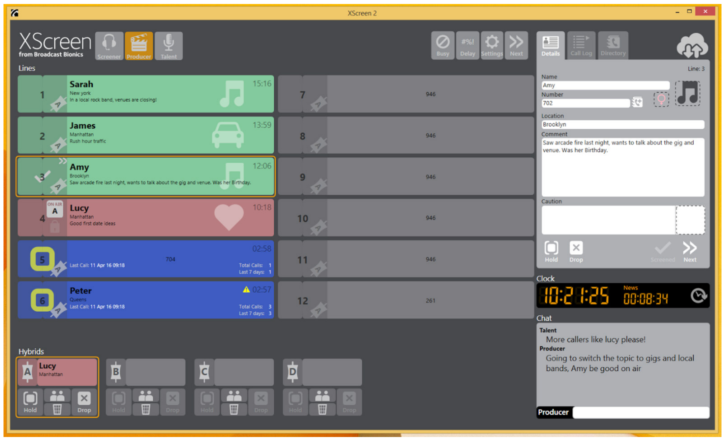
There are plenty of other advantages to melding phones with consoles. Like ease of installation: IP-Audio consoles with built-in phone controllers don't need any additional wires or connections. Their control signaling, caller audio and backfeeds ride on the network connection that's already there. Bringing caller audio into the IP-Audio domain makes it routable like any other audio source. And since the console now communicates directly with the phone hybrid, mundane tasks such as mix-minus generation, starting recording devices, and playback of recorded off-air conversations can all be automated. All of which means faster, more precise phone segments — since operators' eyes never need to leave the console.

VSet6 Phone Controllers

For off-console control, iQ6 works with functional, easy-to-use Telos VSet6 six-line phone controllers. Their big, colorful VGA LCD displays with animated high-contrast Status Symbols make fast work of call screening, queue placement and other tasks; built-in controls for profanity delay and record devices round out its useful toolset.

VSet6 is an IP-based phoneset that also connects to Axia networks with a single Ethernet connection. There's almost no learning curve; VSet phones work like traditional Telos controllers, with calls selected, held, and dropped in the way to which operators have grown accustomed. Easy VSet controls let talent manage incoming lines, lock calls on-air, start an external recording device, and take a queue of calls to air sequentially, for precise management of multi-call interviews or conferences. Next Call functionality speeds workflow for producers, screeners, and talent. The LCD display delivers detailed line status, caller information, caller ID, time ringing-in or on-hold, and even comments entered in the included Xscreen screening software. A built-in address book and call history log round out VSet6's features. Each VSet6 has its own web server for easy remote configuration and software upgrades.

Broadcast Bionics XScreen Lite Call-Screening Software Included



XScreen Lite software comes with every iQ6 purchase and provides Unlimited Lite users, dial, hold, hang up, screened hold and next, conference control, dump mode, lock call, VSet control, telephone number, location, name, point & disposition, chat, clock, and call log (6 hours only) functionality. Please download your XScreen software from www.xscreen2.com.

SPECIFICATIONS

General

- Telos 3rd-generation Adaptive Digital Hybrids
- Telos Exclusive Feedback Reduction Functions
- Send-to-Caller Processing: High-Pass Filter, Frequency Shifter, AGC/Limiter, Program-on-Hold AGC/Limiter, Sample Rate Conversion (with AES option)
- Receive-From-Caller Processing: High-pass “Hum” Filter, Smart AGC / Platform Leveler, Noise Gate, Telos DDEQ (Digital Dynamic Equalization) 3-band Adaptive Spectral Processor, Sample Rate Conversion (with AES option)

Input / Output Channels

- Send channels: 2x
- Receive channels: 2x
- Program-on-Hold channels: 1x
- Connection: 100-BaseT Ethernet (Livewire)

Switching Matrix and Conferencing

- Audio Routing and Switch: All Digital
- Telephone Lines: 6
- Hybrids: 2

Control Ports

- Ethernet 100BASE-T
  - Web server for configuration and software update
  - Call-Screening Interface server allows up to 8 instances of call screening software to connect simultaneously
- GPIO channels: 2x
- Control Interface: Up to 12 attached controllers (any mix of VSet6 phones, Console Controllers or screening software) via Ethernet connection

Power Supply

- Type: Internal auto-ranging, 85–250 VAC auto-switching, 50–60 Hz
- Power consumption: 14.2 Watts

Analog Telephone Connectivity

- Universal interface for worldwide application
- Programmable loop current
- Programmable ring and disconnect signaling (loop drop or tone)
- Programmable Flash time
- Caller ID decoding using Bellcore 212 modem standard

Regulatory

North America: FCC and CE tested and compliant, power supply is UL approved.

Europe: Complies with the European Union Directive 2002/95/EC on the restriction of the use of certain hazardous substances in electrical and electronic equipment (RoHS), as amended by Commission Decisions 2005/618/EC, 2005/717/ EC, 2005/747/EC (RoHS Directive), and WEEE.

Telos® Hx1 & Hx2 Digital Hybrids

POTS Phones Never Sounded So Good

OVERVIEW

Telos® Hx1 one-line and Hx2 two-line POTS telephone hybrids are the most advanced hybrids ever developed for use with analog phone lines. Hx hybrids contain advanced 3rd-generation Telos hybrids for superior audio quality; universal POTS interface features disconnect-signal detection, which works with Telco providers worldwide. Hx hybrids include unique features to make operators' lives easier, such as Auto-Answer with selectable ring count, a switchable mic/line input, call screening and line-hold features, and front-panel send and receive audio metering. Audio sweetening tools include Telos Digital Dynamic EQ (DDEQ) and adjustable smart leveler, symmetrical wide-range AGC and noise gating by Omnia®, studio adaption and pitch shifter for use in open-speaker applications, and adjustable caller override.

FEATURES

- Single (Hx1) or two-line (Hx2) capacity with standard analog I/O (1 each send and receive in/out for Hx1, 2 each for Hx2).
- Convenient switchable mic/line input.
- AES/EBU digital audio I/O option available at time of order, or as a field upgrade kit.
- Our most advanced digital POTS hybrids ever, with DSP algorithms optimized for superior performance with today's wide variety of incoming call types.
- Front-panel send and receive audio metering.
- Telos DDEQ (Digital Dynamic EQ) and adjustable smart-level AGC ensure spectrally consistent audio from call to call — even on notoriously tough cellular callers.
- Excellent trans-hybrid loss of >55dB.
- Smooth, proven, symmetrical wide-range AGC by the audio processing experts at Omnia Audio.
- Studio adaptation and a subtle, inaudible pitch shifter to prevent feedback in open-speaker studio environments.
- Precision adjustable caller override.
- Analog I/O 24-bit A/D/A sample rate conversion, 20 dB headroom from +4 dBu nominal levels.
- AES/EBU I/O sample rate converters accept 32, 44.1, and 48 KHz rates. Clock for outputs may be sourced from the AES inputs or internally-generated at 48 KHz.
- Incredible dynamic range of > 92 db (analog in to analog out, studio loop mode,
  10 hz – 20 khz A-weighted)
- Auto-Answer with selectable ring count and disconnect-signal detection.
- Call screening and line-hold features.

IN DEPTH

Take total control of your talk shows and call-in segments.

In the mid-1980s, Telos pioneered the very first digital adaptive telephone hybrid. Since then, our POTS phone hybrids have earned a worldwide reputation for extracting clean, clear caller audio from even the most difficult calls.

We've contributed plenty of improvements to POTS hybrid technology in the past 20 years, and the Telos Hx1 and Hx2 represent the highest state-of-the-art in hybrid performance. Advances in DSP have been pretty great as well. We've used every bit of knowledge gained to make Hx1 and Hx2 the best, most advanced POTS hybrids we've ever made, without much doubt.

Inside the single-hybrid Hx1 and dual-hybrid Hx2, you'll find Telos processing technologies that take the POTS hybrid to a new level of consistently superior performance, regardless of telephone line characteristics. This advanced hybrid technology brings new standard features that sweeten and control caller audio better than ever before; features you won't find in other POTS hybrids.

On the front panel, you'll find EQ Meters for each hybrid, to tell you exactly how much DDEQ is being applied. Next to those, separate Send and Receive level meters monitor each hybrid. There's also an animated line status display that visually indicates when a line is ringing in, on air, on hold or available. A complement of Take, Hold and Drop buttons complete the front-panel control set.

Around back (Hx2 rear panel shown above), you'll find a switchable mic/line input, balanced analog receive-out output, RJ ports for telco input and phoneset, input level adjustment, and a DB9 remote control connector with GPIO closures for hybrid control and status indicator lamps. Need digital I/O? No problem — Hx comes in an AES/EBU version with built-in sample-rate converter.

Hx1 and Hx2 are probably the most fully-featured POTS hybrids ever created, with Auto-Answer, caller disconnect detection, audio-leveling and anti-feedback routines for open-speaker applications, call screening and line-hold features, and much, much more. Audio processing tools include a new symmetrical wide-range AGC and noise gate by Omnia, with adjustable gain settings to help keep caller audio smooth and consistent from call to call. Adjustable caller override improves performance even further, and allows you to individualize the degree to which the announcer ducks the caller audio. Finally, our famous Digital Dynamic EQ, coupled with an adjustable smart leveler, keeps audio spectrally consistent from call to call.

SPECIFICATIONS

General

- Telos 3rd-generation Adaptive Digital Hybrid.
- Telos Exclusive Feedback Reduction Functions.
- Send-to-Caller Processing: High-pass Filter, Frequency Shifter, AGC/Limiter, Sample Rate Conversion (with AES option).
- Receive-From-Caller Processing: High-pass "Hum" Filter, Smart AGC / Platform Leveler, Noise Gate, Telos DDEQ (Digital Dynamic Equalization) 3-band Adaptive Spectral Processor, Sample Rate Conversion (with AES option)

Analog Inputs:

- Send Analog Inputs: 1 for Hx1, 2 for Hx2 (one per hybrid)
- Connector : XLR Female, Pin 2 High (Active Balanced with RF Protection)
- Input Range: Selectable between MIC and LINE levels
- Line Input Level: Adjustable from -10dBV to +8 dBu (nominal)
- Analog Clip Point : +21 dBu
- Impedance: Bridging, > 50 Ohms
- Analog-to-Digital Converter Resolution: 24 bits

Analog Outputs:

- Receive Analog Outputs: 1 for Hx1, 2 for Hx2 (one per hybrid)
- Connector: XLR Male, Pin 3 High (Active Balanced, RF suppressed
- Output Level: Nominal +4 dBu, fixed
- Impedance: < 50 Ohms
- Digital-to-Analog Converter Resolution: 24 bits
- Headroom Before Clipping: 20 dB headroom above 4 dBU nominal levels

AES3 Digital Inputs/Outputs (optional)

Plug-in module converts standard analog XLR inputs and outputs to AES3 (one input or output on left channel of AES stream)

- Conforms to AES3 standard
- Sample rates: 32kHz to 48kHz.
- Rate conversion: Input and output, independently selectable
- Output Clock: AES input or 48kHz internal.
- Input Level: Nominal at -20 dBFs.
- Output Level: Nominal at -20 dBFs

Audio Performance

- Frequency Response: 200 to 3400 Hz, +/- 1 dB
- THD+N: < 0.5% THD+N using 1 KHz sine wave
- Dynamic Range: Analog in to Analog out, studio loop mode, 10Hz-20Khz. A-weighted: > 92 dB
- SNR: Analog output, referred to -12dBm phone line signal (+4dBu studio out), 10Hz-20Khz a-weighted: > 72 dB
- Trans-Hybrid Loss: Analog phone line with ducking, gate, AGC, EQ
- all OFF relative to +4dBu input level: >55 dB

Analog Telephone Line Connectivity

- Universal interface for worldwide application. Programmable loop current, ring signaling, and flash time. Includes caller ID decoding using Bellcore 212 modem standard.

Power Supply

- Type: Internal auto-ranging, 90–265 VAC auto-switching, 50–60 Hz.
- Power consumption: 100 Watts.

Regulatory

North America: FCC and CE tested and compliant, power supply is UL approved.

Europe: Complies with the European Union Directive 2002/95/EC on the restriction of the use of certain hazardous substances in electrical and electronic equipment (RoHS), as amended by Commission Decisions 2005/618/EC, 2005/717/ EC, 2005/747/EC (RoHS Directive), and WEEE.

Omnia.11 FM and FM/HD

Flagship Class FM Processing for the Most Competitive Broadcasters on the Planet

OVERVIEW

Omnia.11 is available in FM+HD with separate processing paths for FM and HD/DRM or FM without HD/DRM. The FM-only model is upgradeable to FM/HD at a later date. Switchable Single Sideband Suppressed Carrier (SSBSC) technology for potential reduction of multipath is a standard feature. A front panel touch screen GUI, on a 10.5" diagonal screen, provides ease of use and enhanced metering and diagnostics. Remote access is available via any web browser. Livewire, AES/EBU digital and analog I/O are standard. Fanless cooling. Rugged 4 RU chassis.

Omnia.11s now ship with the G-Force™ Dynamics Engine standard. G-Force is also available as an optional Plug-In for Omnia.11 units already in the field. Designed by Frank Foti and Cornelius Gould, Omnia.11 with G-Force represents a significant update to Omnia.11's dynamics—so significant, in fact, that Omnia has updated the GUI to a vivid cobalt blue. It sounds even better than it looks thanks to a dynamics processing framework that enables the Omnia.11 to set the overall EQ for signature consistency, making it sound cleaner, clearer, louder, more consistent, more open, and more pleasing. You hear the music. You hear the voice. You don't hear the processor. Both FM+HD and FM-only models can be upgraded with the optional Perfect Declipper Plug-In, a revolutionary new algorithm that restores clipped areas in audio recordings. This algorithm not only restores dynamics, but removes distortion.

FEATURES

G-Force™ Dynamics Engine

The G-Force Plug-In (which ships standard on all new Omnia.11 units and can be added as an optional upgrade for existing units in the field) lets Omnia.11 handle rapidly changing, hyper-compressed source material better than ever with new, sophisticated improvements. The G-Force dynamics processing framework enables the Omnia.11 set the overall EQ for signature consistency, making it sound cleaner, clearer, louder, more consistent, more open, and more pleasing. You hear the music. You hear the voice. You don't hear the processor.

"Pepino" Clipper

The latest FM final clipper from Frank Foti, custom-engineered to take G-Force Dynamics to the next level. Includes "Pepino Clipper Mode 2," which preserves brightness at aggressive clipping levels.

Presets

Always updating, always evolving, Omnia engineers are constantly striving to provide new and powerful presets. From the best ears in the industry, find your custom sound easier than ever.

Transient Detail Enhancer

In the dynamics section - smarter more powerful RMS control in the AGCs, producing stable solid increase in loudness to match the performance of Pepino.

Solar Plexus

For deep, tight bass that you can feel!

Unified FM/HD Bass Clipper

Improves audio consistency between the FM and HD channels and overall bass quality.

1 Louder

To gain that extra db of loudness.

Intelligent Ultra-Multiband Limiter System

Self-adjusting attack/release functions guarantee crystal clear music and voice. The limiters are self-adapting and can tune themselves to the activity of the AGC section on the fly, providing more powerful and transparent limiter action than possible before. Makes adjustment of limiters a breeze!

Bass Management

Manages harmonics for a natural and undistorted bottom end.

Ultra LoIMD Distortion Controlled Clipper System

Dramatically reduces intermodulation distortion (IMD) for more loudness headroom.

MPX Composite Baseband Over AES (Omnia Direct)

Output of the Omnia.11's stereo generator can be coupled directly to the modulator of the transmitter's exciter. This enables the exciter to modulate with more precision and clarity.

Perfect Declipper Plug-In (Optional)

A revolutionary new algorithm to restore clipped segments and remove distortion in aggressively mastered audio recordings results in a clearer, more open texture that also gives more flexibly with processing choices. (Must be running v3.0 and G-Force.)

SSBSC Technology

Omnia.11 Single Sideband Suppressed Carrier (SSBSC) technology may reduce multipath distortion.

Extra Wide Touchscreen

10.5" diagonal screen clearly shows all controls.

Looks Cool and Stays Cool

Military-grade, fanless industrial design stays cool with heatsinks in rugged chassis.

IN DEPTH

New G-Force™ Dynamics Engine

G-Force (which ships standard on new units and is available as an optional Plug-In to upgrade existing field units) has a highly refined density detection scheme, which means rock-solid performance across a wide range of recordings. Program adaptive attack, release, and ratio values let you set the characteristic elements of your signature sound and make audio acceleration and deceleration smoother than ever. A Makeup Threshold allows for gain management and control without sudden, audible swings. Additionally, AGC sections synchronize with program material. Multiband Limiters now self-adapt to the Multiband AGC activity and also feature program-controlled attack and release, actively reducing limiter-induced inter-mod distortion. Limiters are more responsive and active, yet remarkably transparent, even under extreme activity. G-Force requires v3.0, outlined below.

Software Updates

The G-Force plug-in runs on Omnia.11 v3.0 or later. Included in this general system release are many improvements, including: Static RDS; the flexibility of analog, AES/EBU or Livewire patch points; patch point location for PPM® chains so you can integrate your Voltair/Encoder combo into your audio processing chain; patch point input and output meters for easy level references; upper subharmonic control of Solar Plexus over tonal balance; and compatibility with third party FM-HD time alignment systems.

Version 3.6 is also now available, which features many improvements over v3.0, including the New "Pepino" clipper now with two modes. G-Force processing improvements to take advantage of newly designed clipper. Improved high-frequency handling and more consistent bass response. "Transient Detail Enhancer" in the dynamics section with smarter more powerful RMS control in the AGCs, producing stable solid increase in loudness to match the performance of Pepino. Smoother switching between presets. Unified FM/HD Bass Clipper improves audio consistency between the FM & HD channels and overall bass quality. New Phat Bass update for richer stronger bass presence. Improved sound in the low-delay DJ section. Greatly improved HD look-ahead limiter in HD-enabled units. Warmer and cleaner live DJ voice.

The latest Omnia.11 software is available as a complimentary, downloadable field update at TelosAlliance.com/omnia/omnia11.

The Perfect Declipper Plug-In Option

Just when you thought nothing could sound any better, G-Force-enabled Omnia.11s can be upgraded with the Perfect Declipper Plug-In. Engineered by audio-processing legend Hans van Zutphen, the Perfect Declipper uses a revolutionary new algorithm to replace clipped areas in audio recordings, restoring dynamics and removing distortion. You must be running v3.0 or later and G-Force to install the Perfect Declipper Plug-In.

Intelligent Ultra-Multiband Limiter System

Traditional limiting technology has often resulted in various forms of audio corruption. Omnia.11's new LoIMD technology coupled with smart gain reduction algorithms now have limiters that sound amazingly transparent. All AGC and limiting algorithms employ an auto acceleration / deceleration mechanism, which tunes out perceptible intermodulation distortion. The attack/release functions adjust themselves based upon content density. This breakthrough method literally analyzes the audio content in both the amplitude and frequency domain, then adapts the timing networks—on the fly—to transparently control the signal, without the control being heard. The result is revealed in added detail, clarity, and quality, yet maintaining the desired competitive loudness level. Special attention was paid to the behavior of live voice quality. The improved performance of the AGC and limiter functions generate live voice clarity and impact far beyond that which was previously possible.

Bass Management

The bass enhancement algorithm is a key feature of the Omnia.11. Low end is now broadcast with recording studio-like punch and impact, with no traditional side-effects whatsoever.

Omnia.11's exclusive bass-management method is a mixture of innovation, as well as a rearrangement of the system topology. Achieving great-sounding bass requires the most effort, partly due to the fact that the bass spectrum has the most number of harmonics, and all of these must be kept properly accounted for in the time domain. Also, any additional spectra created (enhancement) must have its harmonic content managed, or the bass region begins to sound distorted and unnatural. This process requires much more than just traditional EQ, bass clipping/filtering, or any ordinary attempt at bass enhancement. Even the location where the function is inserted matters, as well as how it maintains its frequency range along with the rest of the system. An entire dissertation could be done on the bass enhancement/management system alone. The classic Omnia dynamically flat & time aligned crossover system has been further refined to produce smooth, rich, and full tonality. The AGC and limiter sections cannot be fooled into false gain control due to spectral density (or lack thereof) from the crossover network.

G-Force takes this all to another level of greatness that allows broadcasters to adjust the Omnia.11's bass via a single knob. Advanced adjustment mode allows more precise bass sculpting, including Omnia's Solar Plexus bass enhancement feature. An intelligent active bass clipper system allows the full power of the new bass enhancement scheme to come through on the dial.

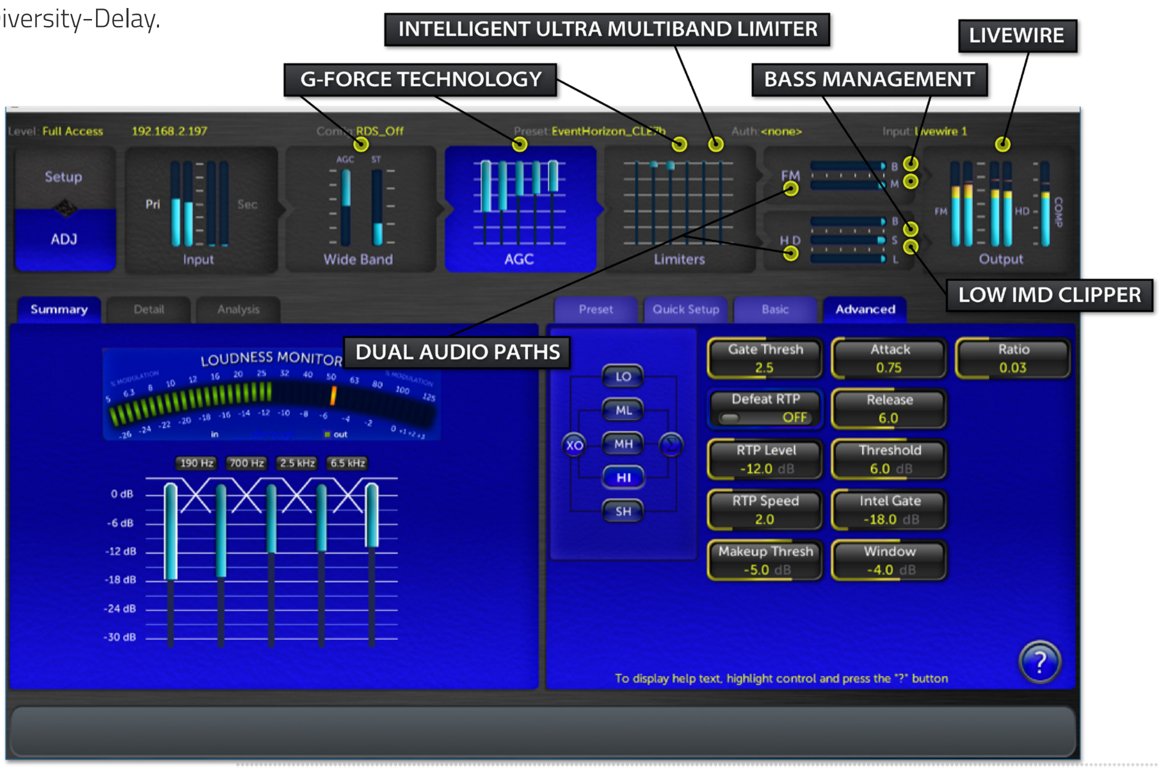
Ultra LoIMD Distortion Controlled Clipper System

Audio processing for conventional broadcast (FM and AM) has, in some applications, reached extreme levels. Various methods are available today capable of creating LOUD competitive signals, but at the expense of perceptible quality. Through critical listening, extensive research, and evaluation of processing methods, it has been determined the single most annoying quotient is due to intermodulation distortion (IMD) induced by aggressive functions within the processing system. The algorithms are pushed to the limits, and beyond. One of the most crucial, aggressively used algorithms in the FM processor is the pre-emphasized final limiter/clipper. Omnia Engineering has developed the new Ultra LoIMD Distortion controlled clipper system specifically to reduce IMD in this critical stage of the processing.

For those who feel the need to use it, there's also a composite clipper embedded in the stereo generator. However, to date, all of our testing has been done without any composite clipping. Pilot protection is on the order of magnitude close to 90 dB, which is considerably more protection than necessary for even the best FM receiver. Integrated laboratory-grade stereo generator with dual MPX outputs, 19 kHz reference output for external RDS/RBDS systems and pilot protection that provides >80dB pilot protection—with or without composite clipping. MPX spectral low-pass filter protects RDS/RBDS and SCA signals if composite clipping is employed. There are multiple ways to adjust the system to achieve the exact sound you're looking for.

Unprecedented Access

- A front panel touch screen GUI, on a 10.5" diagonal screen, provides ease of use and enhanced metering and diagnostics. Remote access is via any web browser.
- Livewire, AES/EBU digital and analog I/O is standard. Headphone soft "patch points" are available for listening through the processing chain.
- Diversity-Delay.



SPECIFICATIONS

General

- Non-linear Crosstalk: > -80 dB, main to sub or sub to main channel (referenced to 100% modulation).
- 38 kHz Suppression: > 70 dB (referenced to 100% modulation).
- 76 kHz Suppression: > 80 dB (referenced to 100% modulation).
- Pilot Protection: > -65 dB relative to 9% pilot injection, ± 1 kHz.
- 57 kHz (RDS/RBDS) Protection: better than -50 dB.
- Connectors: Two EMI suppressed female BNC, floating over chassis ground
- Maximum Load Capacitance: 5nF (at 10 ohms source impedance).
- Maximum cable length: 100 feet/30 meters RG-58A/U.

Analog Audio Input

- Left/Right Stereo. Electronically balanced.
- Input impedance 10k ohms resistive.
- Maximum Input Level: +22 dBu.
- Nominal Input Level: +4dBu, which nets a -18dBFS input meterreading on a steady-state signal when the Input Gain controlis set to 0.0dB. Program material with a nominal average level(VU reading) of +4dBu will typically produce peak readings on the input meter in the range of -12 dBFS to -6dBFS. This is the correct operating level.

A/D Conversion

- Crystal Semiconductor CS5361, 24 bit 128x over-sampled delta-sigma converter with linear-phase anti-aliasing filter. Pre-ADCanti-alias filter, with high-pass filter at <10 Hz.
- Connectors: Two, EMI-suppressed XLR-female. Pin 1 chassis ground, Pin 2 "Hot."

Analog Audio Output

- Left/Right Stereo. Electronically balanced.
- Output Impedance 20 ohms.
- Minimum load Impedance: 600 ohms.
- Output Level adjustable from -2 dBu to +22dBu peak in 0.1dB steps.

D/A Conversion

- Crystal Semiconductor CS4391, 24 bit, 128x oversampled.
- Connectors: Two, EMI-suppressed XLR-male. Pin 1 chassis ground, Pin 2 "Hot."

Frequency Response

- Complies with the standard 50 or 75 microsecond pre-emphasis curve within ± 0.5 dB, 30 Hz to 15 kHz. The analog left/right output and AES/EBU Digital outputs can be configured for flat or pre-emphasized output.

System Distortion

- Less than 0.01% THD, 20 Hz – 7.5 kHz. Second harmonic distortion above 7.5 kHz is not audible in the FM system.
- Signal-Noise Ratio: > -80 dB de-emphasized, 20 Hz – -15 kHz bandwidth, referenced to 100% modulation.
- The measured noise floor will depend upon the settings of the Input and Output Gain controls and is primarily governed by dynamic range of the Crystal Semiconductor CS5361 A/D Converter which is specified as >110 dB. The dynamic range of the internal digital signal processing chain is >144 dB.

Stereo Separation

- Greater than 65 dB, 20 Hz – -15 kHz; 70 dB typical.

Crosstalk

- > -70 dB, 20 Hz -- 15 kHz.

System Latency

- 36-50ms dependant on processing and clipper selection through "FM" channel, as measured from the analog inputs through the composite MPX output.

Composite Outputs

- Source Impedance: 5 ohms or 75 ohms, jumper-selectable. Single ended and floating over chassis ground. Output Level: 0V to 10V in 0.05V steps, software adjustable.

D/A Conversion

- Texas Instruments/Burr Brown PCM1798, 24-bit sigma-delta converter.

Configuration

- Two electrically independent outputs. Software based level adjustment.

Load Impedance

- 50 ohms or greater load is suggested.

Pilot Level

- Adjustable from 4.0% to 12.0% in 0.1% steps and OFF.

Pilot Stability

- 19 kHz, ± 0.5 Hz.

Signal-to-Noise Ratio

- -85 dB typical, 75 μs de-emphasized, 15 kHz bandwidth, referenced to 100% modulation).

Distortion

- < 0.02% THD 20 Hz – 15 kHz bandwidth, 75 μs de-emphasized, referenced to 100% modulation.
- Stereo Separation: > 65 dB, 30 Hz – 15 kHz.
- Linear Crosstalk: > -80 dB, main to sub or sub to main channel, referenced to 100% modulation.

Connector

- XLR-female, EMI-suppressed. Pin 1 chassis ground, Pin 2-3 transformer isolated, balanced, and floating. Standard AES3 specified balanced 110 ohm input impedance.

External Sync Range

- Automatically accepts sample rates between 32kHz and 96kHz. Connector: XLR-female, EMI-suppressed. Pin 1 chassis ground, pins 2 and 3 transformer isolated, balanced, and floating – AES3 standard 110 ohm impedance.

Remote Control

- Via Ethernet using built-in Java (TM) based remote control program integrated into web page interface. All software is served from the built-in web server to any standard web browser; there is nothing to install on the user's computer.

Connectors

- Ethernet - Industry standard EMI-suppressed RJ-45 connector.

GPI Interface

- Connector: EMI suppressed DB-15 female connector.

Power Requirements

- Voltage: 100-250 VAC, 47-63 Hz, Typical: 65W RMS, Max: 90W RMS.

Power Connector

- EMI suppressed IEC male. Detachable 3-wire power cords supplied for US and European use.

Power Supply

- Internal. Overvoltage and short circuit protected.

Digital Audio Input

- Configuration: Stereo per AES/EBU standard, CS8420 Digital Audio Transceiver with 24 bit resolution, software selection of stereo, mono from left, mono from right or mono from sum.
- Automatically accepts and locks to input sample rates between 30 and 108 kHz.
- Connector: XLR-female, EMI-suppressed. Pin 1 chassis ground, pins 2 and 3 transformer isolated, balanced, and floating – AES3 standard 110 ohm impedance.

Digital Audio Output #1

- Stereo per AES3 standard. Output can be configured in software for flat or pre-emphasized response at 50 or 75 microseconds.
- Digital Sample Rates: Output sample rates software selectable for 48kHz, Sync to Input or Sync to External.
- Connector: XLR-male, EMI-suppressed. Pin 1 chassis ground, pins 2 and 3 transformer isolated, balanced, and floating. Standard AES3 specified 110 ohm source impedance.
- Digital Output Level: -22.0 to 0.0 dBFS software adjustable.

Digital Audio Output #2:

- Stereo per AES3 standard. Output can be configured in software for flat pre-emphasized response at 50 or 75 microseconds.
- Digital Sample Rates: Output sample rates software selectable for 48kHz, 44.1kHz or Sync to External.
- Connector: XLR-male, EMI-suppressed. Pin 1 chassis ground, pins 2 and 3 transformer isolated, balanced, and floating. Standard AES3 specified 110 ohm source impedance.
- Digital Output Level: -22.0 to 0.0 dBFS software adjustable.

External Sync Input:

- External Sync: Output sample rate can be synchronized to the signal present on the AES/EBU input, or to an AES3 signal applied to the Ext. Sync input connector. (Does not accept Word Clock Inputs)

Regulatory

North America: FCC and CE tested and compliant, power supply is UL approved.

Europe: Complies with the European Union Directive 2002/95/EC on the restriction of the use of certain hazardous substances in electrical and electronic equipment (RoHS), as amended by Commission Decisions 2005/618/EC, 2005/717/ EC, 2005/747/EC (RoHS Directive), and WEEE.

Omnia.9

Top-of-the-Line FM or AM Processing. Optional HD-1, HD-2, HD-3 Processing and Streaming/ Encoding, RDS. Also Available in Dual Path Model.

OVERVIEW

Omnia.9 is the most powerful and flexible multi-transmission platform audio processor in the world. One Omnia.9 can be configured for FM/AM, dual FM, up to 3 DAB signals, and a separate streaming engine all in one 3RU chassis. With the clear, clean sound that the Omnia.9 is known for, this processor is the choice of stations from classical to hot CHR and everything in between. The newest MKII upgrade includes Livewire+ AES67 I/O, an even more advanced final clipper, and additional logging capabilities.

FEATURES

Exclusive “Undo” Technology

“Undo” is a two-step process that restores peaks and dynamic range to - and removes distortion from - source material that has been damaged by over-compression and clipping during the mastering process.

Psychoacoustically Controlled Distortion-Masking Clipper

Omnia.9’s final FM clipper takes into account how the human ear hears and perceives distortion and uses that information to effectively mask it, leaving only clean, distortion-free audio on the air.

Livewire + AES67

For even more flexible I/O.

Omnia µMPX Encoding

Omnia 9 is the the FIRST hardware processor to support this revolutionary codec. Full composite MPX over a 320kbps pipe.

Kantar Watermarking

For our customers in France, support for Kantar watermarking.

Omnia® Toolbox

Every Omnia.9 comes with a “toolbox” that includes loudness metering, a digital oscilloscope, an FFT spectrum analyzer, and real time analyzer (RTA) to help you adjust your processing and “see what you’re hearing”.

Speaker Calibration

With the addition of a calibrated microphone, Omnia.9’s built-in real time analyzer (RTA), pink noise generator, and parametric equalizer can be used to calibrate any speaker system.

Dry Voice Detector

The dry voice detector is a selectable feature that helps eliminate the audible distortion sometimes evident on bare voice (voice with no music mixed under it) when very aggressive processing settings are used.

Remote Client

Omnia.9’s client software allows full remote control of the processor from any Windows-based PC or tablet, including touch screen devices, on the local network. The remote interface looks and functions just like the front panel screen.

Auto Pilot

A selectable feature that automatically turns off the 19kHz stereo pilot when Omnia.9 detects that the source material is mono, and turns it back on when the input is determined to be stereo, automatically gaining a 20 dB increase in signal to noise during mono programming.

Additional Features

- Multiband downward expansion (source noise reduction)
- 3-stage wideband AGC with adjustable sidechain equalization
- Phase Correction with Mono Bass reduces multipath distortion
- Supports Shoutcast 2 and lossless streaming. Integrated internal stream server
- Shared processing path for AM+HD units (allowing them to share a watermark encoder)
- Program-dependent multiband compression
- Multiband look-ahead limiting
- Selectable phase linear high pass filter, 15, 30 or 45 Hz
- Two-band final look-ahead limiting on digital paths
- 7 inch front panel touch screen
- Full remote control via included NFRemote software with client audio streaming for remote adjustment and monitoring. Also supports full remote control and monitoring via HTTP request
- On-screen keyboard with several layouts (QWERTY, QWERTZ, AZERTY, Dvorak and ABC sequential) for easy setup and preset name typing
- Selectable SSB (Single Sideband) stereo encoder
- HTTP push support for automation, such as RDS and streaming song titles, preset recall
- Studio Output with very low latency for talent monitoring
- Dual independent power supplies
- Composite pass-through (relay bypass) for your backup processor, with full analysis tools for external composite signals
- Internal patch point capability so you can optimize placement of your Voltair / PPM® encoder within your audio processing chain.

IN DEPTH

"Undo"

The first step of Undo is the de-clipper, which examines and recreates audio peaks that were clipped during mastering. The second step is a multi-band expander that creates dynamic range. Clean, well-recorded audio has always been able to withstand greater degrees of processing. This was true decades ago and it's still true (and more relevant than ever) today. An FM processor, by its very nature, compresses dynamic range and employs some form of clipping to deliver a "signature sound" and a competitively loud signal on the air. It is an unfortunate but well-accepted fact that recordings made in the past two decades have been on the decline in terms of quality, as mastering engineers seem to be waging their very own "loudness wars". The result is source material that is hyper-compressed right out of the jewel case with only a dB or two of dynamic range at most. As if that weren't bad enough, the music is run through unsophisticated, brute-force clippers to make them louder still. The result is that the audio going INTO a processor today sounds more distorted than the audio coming OUT of an FM air chain 10 years ago! Before it even gets touched by the compressors, limiters, and clippers in the processor itself, it has been damaged. Rip a track from the modern CD of your choice and look at the waveform in your favorite editor if you need proof. Processors add more distortion still, and the resulting "music" heard on the air is nearly unlistenable. By repairing the damaged audio first, "Undo" gives Omnia.9 cleaner and more dynamic audio to work with, which can better stand up to the rigors of on-air processing. The result is a clean, dynamic, and listenable sound on the air. In fact, audio processed by Omnia.9 for FM often sounds far better than the original CD.

Psychoacoustically Controlled Distortion-Masking Clipper

Clipping is typically the final stage of an FM processing chain. The majority of clipping is usually done in the final L/R audio, with additional, optional clipping available in the composite signal. The final clipper is also where the classic (and oft dreaded) "loud v. clean" tradeoff is made. When more clipping is used to gain loudness on the dial, clipper distortion becomes more and more pronounced. The clipped peaks fall back into the audio and manifest themselves as audible distortion.

There are ways to get around that problem, but they come at a price. You could back down on the clipper drive to clean up the sound, but then you lose loudness. Or, you could put more of the "heavy lifting" on the compressors and limiters preceding the clipper, but that results in an overly busy, dense sound that robs the music of life and causes listener fatigue. Some processors HAVE to resort to building excess density in the dynamics section because their simple or old-technology clippers simply aren't up to the job. The Omnia.9 identifies clipper distortion and uses a proprietary psychoacoustic-controlled algorithm in the composite signal to mask it, effectively eliminating it from the final audio. It is so robust that it boasts an additional 3dB of high-frequency headroom and is capable of 140% L/R modulation within 100% total modulation. That means Omnia.9 can be significantly cleaner for a given loudness level or substantially louder for a given level of quality. It comes closer to eliminating the "either/or" compromise than any other processor on the market today.

Omnia Toolbox

When Leif Claesson was creating Omnia.9, he knew that having diagnostic and measurement tools would be necessary. The original plan was to keep them in place only for development, but he quickly realized that engineers would find great value in them as well, and decided to leave them in place. Audio processing is largely a “hearing” process, but there is much to be learned by seeing what your adjustments are doing to your sound as well. Some stations still have an oscilloscope on the test bench or a spectrum analyzer at the transmitter, but it’s not always convenient (or possible) to hook up a processor to them while it’s on the air.

Even if you did so, you’re pretty much limited to monitoring only the composite output of your own station’s processing. Omnia.9’s built-in solution means there’s no extra test equipment to buy (‘scopes and analyzers aren’t cheap) and no cables to hook up. It also means you can visually monitor the signal at the input, the output, and dozens of other points throughout the processing path so you can tell what’s happening to the audio every step of the way. As an added bonus, Omnia.9’s composite inputs can be fed from a calibrated tuner or frequency-agile mod monitor so that you can monitor the other signals in your market too!

In addition to these tools, Omnia.9 also includes RTA and speaker calibration tools to further assist with monitoring and fine-tuning your processing. While it is certainly good practice to listen to your station on a variety of radios and speakers as you adjust your processing, it is also good practice to have at least one set of calibrated speakers available. Otherwise, the changes you make to your processing will be influenced by listening to speakers that don’t accurately reflect the frequency response of your processing adjustments. By adding an inexpensive calibrated microphone and using the included pink noise generator and RTA, you can quickly and easily calibrate a set of speakers to use as a reference as you adjust your sound.

Speaker Calibration

If you make decisions about your processing on uncalibrated monitors, you are making choices that are influenced by the differences in frequency response present in every speaker, not to mention the coloration imposed by the room in which you are listening. Simply put, you're dealing with subjective, not objective, information. By using the pink noise generator and RTA built into Omnia.9 and adding an inexpensive calibrated microphone, it is possible to calibrate any speaker system to deliver as flat a response as the speakers themselves will allow (small speakers still won't reproduce low frequencies as well as larger ones – the laws of physics still apply after all). With speaker and room influences removed from the equation, you are in a position to adjust your audio based only upon "the facts." When explaining this process to someone in person, this is the point in the conversation where they inevitably say, "But listeners aren't hearing my station on calibrated speakers! They're listening in their cars, at their computer, and through cheap ear buds, so I should too!" It's true - that's exactly how your listeners are hearing your station in the real world, and why it is always important to listen on a wide variety of radios in many different environments. But adjusting your processing this way is a shortcut to a lot of tail-chasing frustration and lousy audio. Let's say you listen first in an inexpensive compact car with a typical factory stereo. You notice there isn't much bass, so you adjust your processing to deliver more low end. It sounds good. Then you move into a high-end luxury car with 10 speakers and a subwoofer, and the bass is muddy, boomy, and overwhelming. Why? Because you adjusted the bass in the processor to make up for deficiencies you thought were in your processing, but in fact were in your speakers! Having at least one pair of high quality, calibrated speakers to go back to as your reference will dramatically improve your on-air sound, save you valuable time, and help preserve your sanity at the same time. (Don't worry – there are still plenty of people at your station to chip away at your mental well-being – we just don't want to be among them!).

Dry Voice Detector

We know that the human voice can present a tough challenge to an FM processor. If it's bare voice – that is, voice alone with no music mixed underneath – any distortion created in the processing really stands out. We also know that all-out loudness comes at a price: At some point, you have to give up "clean" to get "loud." Even Omnia.9's psychoacoustically-controlled distortion-masking clipper, which really minimizes the dreaded "clean v. loud" tradeoff, can reveal some distortion on dry voice when the overall processing is set up to really push for loudness. So ensure clean voice quality in these situations, the dry voice detector first determines that the incoming audio is actually bare voice. It then automatically and inaudibly transfers more of the "heavy lifting" to the compressor and limiter sections, thereby reducing the amount of overall clipping needed to maintain the same level of loudness.

Remote Client

Every modern processor provides some means by which to control it or adjust its settings remotely, which is handy if the processor is at a transmitter site miles (and often mountains) away from the studio. Most employ web-based interfaces, which on the surface sounds convenient because it allows you to remote in from a browser on any computer at any location, but even the best of them fall short when it comes to a great user experience. They require browser plug-ins, typically feel "laggy" when viewing meters or adjusting controls, and don't always have the same look and feel as the front panel interface. Omnia.9's client software delivers exactly the same experience whether you're standing in front of the processor or controlling it from your PC or tablet. If you have Omnia.9's on more than one station in your group (who can buy just one?) you can connect to any of them through a single connection window, and can run multiple remotes simultaneously.

Providing your connection has sufficient bandwidth, you can even stream audio from various patch points within the processing chain back to the client computer. This allows you to hear what effect your adjustments have on your audio in the environment of your choice instead of a rack room or transmitter building, locations which almost never have decent monitors but offer noise in abundance!

Auto Pilot

The ability to transmit 2-channel source audio in FM stereo certainly has a sonic advantage, but it's far from a "free ride". The stereo pilot typically claims around 9% of total modulation; stereo signals are more susceptible to noise and multipath distortion than mono FM signals; and for a given RF level at the receiver, the signal-to-noise ratio of a stereo signal will be worse than that of a mono signal. Those are acceptable tradeoffs if you're actually playing stereo music, but if your source material is mono (be it mono music or talk programming), it hardly seems fair to force those compromises upon your signal when there's no reason to do so.

Thus, the Auto Pilot feature will automatically turn off the pilot (resulting in mono transmission) when the audio is mono, lowering the noise floor by 20dB and stopping multipath in its tracks when it is most audible.

SPECIFICATIONS

Frequency Response

- +/-0.5dB 20Hz to 15kHz, 17.5kHz in extended mode

Signal to Noise Ratio

- Greater than -80dBu de-emphasized, 20Hz to 15kHz

System Distortion

- Less than 0.01% THD below pre-emphasis, inaudible above

Stereo Separation

- 65dB minimum, 20Hz to 15kHz, 70dB typical

Digital Output Level

- Adjustable from -24.0dBFS to 0.0dBFS in 0.1dB increments

Stereo Baseband Output

- Adjustable from -2dBU to +22dBU (0.1dB increments) into 600-Ohms, 20-Ohm output impedance

A/D Conversion

- Crystal Semiconductor CS5361, 24 bit 128x over-sampled delta sigma converter with linear-phase anti-aliasing filter.
- Pre-ADC anti-alias filter, with high-pass filter at <10 Hz

D/A Conversion

- Crystal Semiconductor CS4391, 24-bit, 128x oversampled

External Sync Input

- Per AES11 Digital Audio Reference Signal (DARS), reference for digital output sample rate. Range is 32kHz to 96kHz.

Analog I/O

- Two balanced, EMI filtered XLR connectors

Stereo Generator Connections

- Four 75-Ohm BNC female, two inputs, two outputs
- (FM style only) AES/EBU In & External Sync

Digital I/O

- AES/EBU via four XLR connectors for Main and Aux Digital programs (two stereo in, two stereo out)

Ethernet

- Shared RJ45 supporting 100 and 1000BASE-T Ethernet connections

Power Requirements

- 100-264 VAC, 47-63Hz autosensing, dual PSU

Power Connector

- Dual IEC male, detachable 3-wire power cords supplied

Power Supply

- Internal dual redundant, hot-swappable

Dimensions and Weight:

- 3RU - 5"H x 19"W x 18"D (12.7 x 48.26 x 45.72 cm)
- Net weight: 28 lbs. (12.7006 kg)

Environmental

- Operating: 0 to 50 degrees C
- Non-operating: –20 to 70 degrees C.

Regulatory

North America: FCC and CE tested and compliant, power supply is UL approved.

Europe: Complies with the European Union Directive 2002/95/EC on the restriction of the use of certain hazardous substances in electrical and electronic equipment (RoHS), as amended by Commission Decisions 2005/618/EC, 2005/717/ EC, 2005/747/EC (RoHS Directive), and WEEE.

Omnia.9sg

Your Processor's Secret Weapon

AES67 Livewire+

OVERVIEW

For split audio processing applications, no one does it better than the processor-agnostic 9sg!

We know that the main processor must be installed at either the studio or at the transmitter. Each location has its advantages and disadvantages. Placing a processor at the studio is often more convenient as some transmitter sites are difficult to access, but doing so can compromise quality and loudness as STL audio quality varies and the clippers found in other stereo generators are often mediocre at best. Some transmitters have built-in stereo generators but quality and features vary.

Placing the processor at the transmitter site allows the composite signal from the processor to be fed directly into the transmitter, which provides the best audio quality and the most loudness, but not all transmitter sites have adequate network connectivity for remote control and are often located in remote or difficult-to-access areas. This means making adjustments to the processing is often a challenge.

Split processing—placing the main processor at the studio and performing final stage limiting and stereo generation at the transmitter—can be an ideal, no-compromise solution with the Omnia 9sg.

The processor-agnostic Omnia.9sg was always more than just a stereo generator. With the latest software, this final-stage processor takes its next leap forward with a new clipper design, Livewire+ AES67, audio playout with internal processing for localization or backup, and more.

FEATURES

- Full IP remote with remote audio streaming. Because transmitters are often located in hard-to-reach locations, full-featured remote access is critical.
- Optional RDS encoder supports UECP protocol, allowing each Omnia.9sg to be individually addressed to customize and localize RDS information.
- Built-in http server with push support for dynamic RDS data (with optional internal RDS encoder option).
- Selectable SSB (single sideband) stereo encoding makes SSB compatible with nearly all receivers.
- RF bandwidth controller reduces multipath distortion.
- ITU-R BS.412 power limiter for European countries.
- Auto Pilot turns off pilot for mono content, reducing noise, great for spoken word/news/talk/sports formats.
- Relay bypass including composite pass-through. Should Omnia.9sg fail or lose power, a backup standalone processor can be fed through it and be put on-air automatically and immediately.
- Built-in internal playback capability with processing. Should normal audio be lost, the built-in player in Omnia.9sg can immediately and automatically provide back-up content and audio processing until the problem is resolved.
- Optional local audio insertion allows each Omnia.9sg to interrupt normal program audio and insert local content such as traffic, weather, and geo-targeting advertising, with audio processing.
- “Omnia Toolbox” features including oscilloscope, FFT, and RTA, valuable signal-analysis tools built right into the product, eliminating the need for standalone engineering tools.
- Dual redundant power supplies. Back up if one supply fails. Each supply can be fed from a different electrical circuit; if one circuit fails, the unit stays on air.
- Two composite inputs, two composite outputs. A “hot” backup processor can be looped through the Omnia.9sg and be automatically and immediately placed on air should the 9sg fail or lose power; also makes it possible to use an external RDS encoder if desired.
- Built in stream receiver allows for a web stream to serve as an additional audio source
- Optional Kantar-certified watermarking support

IN DEPTH

New Clipper Design

Audio processing architect Hans van Zutphen designed the new clipper now featured in the Omnia.9sg. This psychoacoustically controlled distortion masking clipper is louder, cleaner, and more efficient. It takes into account how the human ear perceives distortion and uses that information to effectively mask it, leaving only clean, distortion-free audio on the air. The new clipper also uses less internal processing power from the CPU to get the job done faster, resulting in lower latency.

Omnia.9sg Is Processor Agnostic

Omina.9sg is processor agnostic, making the processor you love even more loveable. That is, it can be used to improve the audio quality and loudness of any station with any processor from any manufacturer. If a station can't afford a brand new top-of-the-line processor, or if they like the sound of the front end of their current processor but want better back-end performance, they can add a 9sg for less than half the cost of a new all-in-one box.

No-Compromise Split Processing

We know that the main processor must be installed at either the studio or at the transmitter. Each location has its advantages and disadvantages. Placing a processor at the studio is often more convenient as some transmitter sites are difficult to access, but doing so can compromise quality and loudness as STL audio quality varies and the clippers found in other stereo generators are often mediocre at best. Some transmitters have built-in stereo generators but quality and features vary.

Placing the processor at the transmitter site allows the composite signal from the processor to be fed directly into the transmitter, which provides the best audio quality and the most loudness, but not all transmitter sites have adequate network connectivity for remote control and are often located in remote or difficult-to-access areas. This means making adjustments to the processing is often a challenge.

Split processing—placing the main processor at the studio and performing final stage limiting and stereo generation at the transmitter—can be an ideal, no-compromise solution with the Omnia 9sg.

Multiple Transmitter Sites

Many FM broadcasters have their main transmitter at one tower site and their backup transmitter at another. A stereo generator is required at both transmitters, which often means two complete standalone processors. Installing an Omnia.9sg at each transmitter site allows a station to use the same main processor at the studio to feed both sites, which represents a potential cost savings and consistent processing between the main and auxiliary sites. This applies to applications where a common STL is shared between the sites or when individual STLs are used.

In Europe, it is common for a national broadcaster to work from a single studio location and have dozens or even hundreds of transmitter sites located throughout the country. The appeal of having one main processor at the studio and an Omnia.9sg at each transmitter site works brilliantly on this larger scale as well.

When configured with the local audio insertion option, these national broadcasters can also interrupt network content and insert localized content at each transmitter site such as local traffic, weather, or geo-targeted advertising.

Internal Playback with Processing

Omnia.9sg can now store audio on its SSD for internal file playback. The playback system can be used as a backup source in case something upstream in the chain fails (STL, studio, playback automation system), or for the insertion of local ads, IDs, weather, etc. To support local playout at the transmitter, Omnia.9sg includes a highly capable built-in 4-band processor of its own. Based upon the dynamics processing found in Omnia.7 and Omnia.9, it includes over 20 newly created presets designed specifically to work with Omnia.9sg's clipper.

Optionally, this internal processing can be licensed to process auxilliary input streams, turning your 9SG into a fully functional backup processor.

Ratings Encoder & Enhancement Applications

Research shows that ratings encoders and/or enhancement devices such as Voltair benefit from being fed processed audio. Some processors have special "insert points" that make this possible internally, but many do not. Placing the main processor and the encoder at the studio and using Omnia.9sg at the transmitter can help facilitate this.

SPECIFICATIONS

Frequency Response:
+/- .5 dB 20 Hz to 15 kHz

Signal-Noise Ratio:
Greater than -80 dBu de-emphasized, 20 Hz to 15 kHz

Stereo Separation:
65 dB minimum, 20 Hz to 15 kHz, 70 dB typical

Stereo Baseband Output:
Adjustable from -2 dBu to +22 dBu (0.1 dB increments) into 600-Ohms, 20-Ohm output impedance

AoIP Networking
Livewire+ AES67 Compatible

Analog Inputs:
Two balanced, EMI filtered XLR connectors

Digital Inputs:
AES/EBU In & External Sync

Composite I/O:
Four 75-Ohm BNC female, two inputs, two outputs

Remote Control:
RJ45 supporting 100BASE-T Ethernet connections

Power Requirements:

100-264 VAC, 47-63Hz autosensing, dual PSU

Power Connector:

Dual IEC male, detachable 3-wire power cords supplied

Environmental:

Operating: 0 to 50 degrees C

Non-Operating: -20 to 70 degrees C

Regulatory

North America: FCC and CE tested and compliant, power supply is UL approved.

Europe: Complies with the European Union Directive 2002/95/EC on the restriction of the use of certain hazardous substances in electrical and electronic equipment (RoHS), as amended by Commission Decisions 2005/618/EC, 2005/717/ EC, 2005/747/EC (RoHS Directive), and WEEE.

Omnia.7AM

AM Never Sounded This Good

OVERVIEW

Do you want your AM broadcasts to sound CLEANER, CLEARER, and LOUDER? Then you need a processor designed and built for today's AM radio.

Meet Omnia.7AM, a feature-rich, competitively priced AM audio processor engineered to do just that, and the first processor dedicated to AM to appear on the scene in many years.

While competitors are using old and outdated technology in their processors, the Omnia.7AM uses the best of current technology to meet current challenges faced by AM broadcasters. Omnia.7 AM delivers the powerful, clear, and precise Omnia signature sound that's the first choice of top stations worldwide.

The Omnia.7AM employs most of the features of the 7FM. All aspects of the processing infrastructure, bandwidth, and their output signals, however, have been specially engineered for maximum efficiency and performance within the AM spectrum.

AM Never Sounded This Good

FEATURES

- "Undo," exclusive Omnia technology that removes distortion and mathematically re-creates the peaks sliced from today's poorly mastered contemporary music. Undo restores life, brilliance, and dynamic range to any type of music.
- An exclusive Psychoacoustic Controlled Distortion Masking Clipper analyzes and masks distortion perceptible to the human ear, leaving only clean, clear audio.
- A complete toolbox of sophisticated Omnia sound-shaping technology gives you the power to analyze and refine your signature sound using a variety of sonic tools ranging from Real-Time Analyzers to Oscilloscopes, FFTs, and more.
- Remote client software allows full remote control of processor and all metering tools from any Windows-based PC or tablet on the local network—including touch-screen devices.
- Dry Voice Detector detects speech and applies appropriate processing for clearest possible voice quality.
- Built-in Speaker Calibration tool.
- Multiband downward expansion (source noise reduction).
- Three-stage wideband AGC with adjustable sidechain equalization.
- Program-dependent two- to five-band multiband AGCs and limiters.
- 4.3" / 10.9 cm. front panel screen.
- Full remote control with audio monitoring.
- Dual, independent power supplies.
- Composite pass-through (relay bypass) for backup processor.
- Asymmetrical output with 150% maximum positive peak modulation, 100% maximum negative peak modulation, and peak inversion controls for the input and output audio.
- Available with optional HD and/or streaming.

IN DEPTH

“Undo”

A processor, by its very nature, compresses dynamic range and employs some form of clipping to deliver a “signature sound” and a competitively loud signal. Clean, well-recorded audio has always been able to withstand greater degrees of processing; this was true decades ago and it’s more relevant than ever today.

Unfortunately, recordings made in the past two decades have declined in terms of quality, as mastering engineers wage their very own “loudness wars.” (Rip a track from the modern CD of your choice and look at the waveform in your favorite editor if you need proof!)

The result is source material that is hyper-compressed from the studio, with only a dB or two of dynamic range at most. As if that weren’t bad enough, the music is run through unsophisticated, brute-force clippers to make them louder still. Before it even gets touched by the compressors, limiters, and clippers in the processor itself, it has been damaged.

By repairing the damaged audio before processing, “Undo” gives Omnia.7AM cleaner and more dynamic audio to work with. The first step of Undo is the declipper, which examines and mathematically re-creates audio peaks that were flattened during mastering. The second step, a multi-band expander, increases dynamic range. The result: clean, dynamic, enjoyable sound.

Psychoacoustically Controlled Distortion-Masking Clipper

Clipping is typically the final stage of a processing chain. The final clipper is also where the classic (and oft dreaded) “loud versus clean” tradeoff is made. When more clipping is used to gain loudness on the dial, clipper distortion becomes more and more pronounced. The clipped peaks fall back into the audio and manifest themselves as audible distortion.

How to solve this problem? You could back down the clipper drive to clean up the sound — but you lose loudness. You could dial up the compressors and limiters that precede the clipper — but that results in busy, dense sound that leads to listener fatigue.

To put it plainly: Omnia.7AM sounds significantly cleaner than other processors at a given loudness level — and substantially louder at any given level of quality. It comes closer to eliminating the “loud / clean” compromise than any other processor on the market today. Voices sound clean, while music and production sounds surprisingly vibrant on AM transmission.

This is especially important for AM because unlike FM where loudness is desired for dial-dominating sound, loudness on AM directly translates to increased signal coverage, especially on the fringes of the signal. So it’s even more important to have the ability to be loud AND clean on AM since more and better coverage means more potential listeners, ultimately resulting in potentially more revenue.

Omnia Toolbox

While audio processing is largely a "hearing" process, there is still much to be learned by seeing what your adjustments are doing to your sound. Some stations still have an oscilloscope on the test bench or a spectrum analyzer at the transmitter, but it's not always convenient (or possible) to hook up a processor to them while on the air.

With Omnia.7AM, there's no extra test equipment to buy ('scopes and analyzers aren't cheap) and no cables to hook up. You also have the built-in capability to visually monitor the signal at the input, the output, and dozens of in-between points throughout the processing path so you can tell what's happening to your audio every step of the way.

Speaker Calibration

If you make decisions about your processing on uncalibrated monitors, your choices are colored by the audio characteristics of the speaker itself — not to mention those of your listening room.

The pink noise generator and RTA built into Omnia.7AM, used with an inexpensive calibrated microphone, makes it possible to calibrate any speaker system to deliver as flat a response as the speakers will allow. (Small speakers still won't reproduce low frequencies well; the laws of physics still apply!) With speaker and room influences removed from the equation, you are free to adjust your audio based only upon "the facts."

"But," you say, "listeners aren't hearing my station on calibrated speakers! They're listening in their cars, at their computer, and through cheap ear buds, so I should too." It's true — that's exactly how your listeners hear your station, and why listening on a variety of radios, in many different environments, is a good idea. But adjusting your processing this way invariably results in frustration and lousy audio.

Here's why: You listen first in a compact car with a typical factory stereo. You don't hear much bass, so you adjust your processing to deliver more low end. Then, you move to a luxury car with 10 speakers and a subwoofer, and the bass is muddy, boomy, and overwhelming. Why? Because you adjusted the processing to make up for its perceived deficiencies, when the real deficiency was in your speakers!

Having at least one pair of high-quality, calibrated speakers as your reference will dramatically improve your on-air sound, save you valuable time—and help preserve your sanity, too!

Dry Voice Detector

The human voice can often present a real challenge. Even Omnia.7AM's psychoacoustically controlled distortion-masking clipper, which dramatically minimizes the dreaded "clean / loud" tradeoff, can reveal some distortion on voices when overall loudness is the goal.

To ensure clean voice quality in these situations, the Dry Voice Detector listens for speech, then automatically and inaudibly adjusts the compressor and limiter sections, reducing the amount of overall clipping needed to maintain the same level of loudness.

Remote Client

Remote control is a must — especially when your processor is miles (and often mountains) away from the studio. Omnia.7AM takes remote control to a new high, with a high-performance Web interface that eliminates interface lag. And, if you have multiple Omnia.7AM processors, its single connection window enables you to manage multiple remotes simultaneously.

Provided your network has sufficient bandwidth, you can even stream audio from various patch points within the processing chain back to your computer, so you can hear the effect of your adjustments in the quiet of your office — not inside a noisy transmitter building.

SPECIFICATIONS

Frequency Response

- 20Hz to 10.0kHz, +/-0.5dB.
- Adjustable NRSC pre-emphasis curve implementation. You may instead choose pre-emphasis at 50us or 75us, or even a flat output if desired. Low pass filter can be set between 3.0kHz and 10.0kHz in 0.5kHz increments. This allows maximum high-frequency transmission while allowing for station-specific bandwidth restriction scenarios such as HD Radio, 9 kHz channel spacing, or even more narrow-band shortwave transmission.
- +/- 0.5dB 20Hz to 15kHz; 16.5kHz in extended mode.

Signal-to-Noise Ratio

- Greater than -80 dB, de-emphasized, 20Hz to 15kHz

System Distortion

- Less than 0.01% THD+N below pre-emphasis, inaudible above

Stereo Separation

- 65dB minimum, 20Hz to 15kHz; 70dB typical

Stereo Baseband Output

- Adjustable from -24.0dBFS to 0.0dBFS in 0.1 dB increments

Digital Output Level

- Adjustable from -24.0dBFS to 0.0dBFS in 0.1dB increments

A/D Conversion

- Crystal Semiconductor CS5361, 24 bit 128x over-sampled delta sigma converter with linear-phase anti-aliasing filter
- Pre-ADC anti-alias filter, with high-pass filter at <10 Hz
- Delta sigma converter with linear-phase and anti-aliasing filter
- MPX Inputs have high pass filter <0.1Hz

D/A Conversion

- Crystal Semiconductor CS4391, 24-bit, 128x oversampled
- External sync input
- Per AES11 Digital Audio Reference Signal (DARS), reference for digital output sample rate
- MPX Outputs are DC coupled

Analog I/O

- Two balanced, EMI filtered XLR connectors

AES Digital I/O

- AES/EBU In; Out via XLR connectors
- Input accepts 32000 – 96000 Hz. Output is 44100 or 48000 depending on the rate selected in software. AES Reference Input via BNC connector. Accepts 44100 or 48000 Hz only, and the correct rate must first be selected in the software.

External Sync Range

- 44.1kHz or 48kHz

Inputs/Outputs

- Balanced, EMI-filtered, L/R analog input and output on XLR connectors
- AES input and output on XLR connectors, including recognition of external sync signal
- Ethernet RJ-45 port supporting 100 and 1000 BASE-T Ethernet. at 44.1 or 48kHz

Power Requirements

- 100-264 VAC, 47-63Hz autosensing, 100W maximum

Power Connector

- IEC male, detachable 3-wire power cords supplied

Power Supply

- Internal dual redundant

Environmental

- Operating: 0 to 50 degrees C
- Non-operating: –20 to 70 degrees C

Physical Specifications

- Unit weight: 11 pounds
- Total shipping weight: 15 pounds
- Dimensions: 2RU at 3.5" H x 19" W x 12.5&" D

Regulatory

North America: FCC and CE tested and compliant, power supply is UL approved.

Europe: Complies with the European Union Directive 2002/95/EC on the restriction of the use of certain hazardous substances in electrical and electronic equipment (RoHS), as amended by Commission Decisions 2005/618/EC, 2005/717/ EC, 2005/747/EC (RoHS Directive), and WEEE.

Omnia.7FM

Premium Performance, Priced Right.

OVERVIEW

Up to now, there have been two choices when it comes to audio processing: all the features and advanced audio-shaping tools (with a five-figure price tag), or gear that fits your budget — but made you compromise on performance and capabilities.

No more compromises. Meet Omnia.7FM, the premium, feature-rich FM audio processor that's surprisingly affordable. But low price doesn't mean low performance: Omnia.7FM delivers the powerful, clear and precise Omnia® signature sound that's the first choice of top stations worldwide.

Omnia.7FM comes with a host of standard features:

- Selectable FM or HD+Streaming/Encoding
- "Undo," exclusive Omnia technology that removes distortion and mathematically re-creates the peaks sliced from today's poorly mastered contemporary music. Undo restores life, brilliance, and dynamic range to any type of music.
- An exclusive Psychoacoustically Controlled Distortion Masking Clipper analyzes and masks distortion perceptible to the human ear, leaving only clean, clear audio.
- A complete toolbox of sophisticated Omnia sound-shaping technology gives you the power to analyze and refine your signature sound using a variety of sonic tools ranging from loudness metering to Real-Time Analyzers to Oscilloscopes, FFTs, and more.

Simultaneous HD, Internet streaming / encoding and RDS options are also available, putting Omnia.7FM head-and-shoulders above any other comparably priced audio processor in features, performance, and value.

FEATURES

- Selectable FM or HD+streaming standard; optional upgrade to combinations of simultaneous FM+HD+Streaming+RDS.
- Omnia-exclusive "Undo" Audio Restoration Technology
- Psychoacoustically Controlled Distortion Masking FM Clipper
- Two-band final look-ahead limiter for HD Radio and streaming
- Full Omnia Toolbox, with loudness metering, a digital oscilloscope, an FFT spectrum analyzer, and Real Time Analyzer (RTA)
- Remote client software allows full remote control of processor and all metering tools from any Windows-based PC or tablet on the local network — including touch-screen devices
- Dry Voice Detector detects speech and applies appropriate processing for clearest possible voice quality
- Built-in Speaker Calibration tool
- Multiband downward expansion (source noise reduction)
- Three-stage wideband AGC with adjustable sidechain equalization
- Program-dependent two-to-five-band multiband AGCs and limiters
- 4.3" / 10.9 cm. front panel screen
- Full remote control with audio monitoring
- HTTP push support for automation such as dynamic RDS and streaming song titles, with preset recall
- Dual, independent power supplies
- Composite pass-through (relay bypass) for backup processor

Optional Features

- Simultaneous streaming processing / encoding
- Simultaneous processing for HD Radio or DAB
- RDS encoding

IN DEPTH

"Undo"

The first step of Undo is the de-clipper, which examines and recreates audio peaks that were clipped during mastering. The second step is a multi-band expander that creates dynamic range. Clean, well-recorded audio has always been able to withstand greater degrees of processing. This was true decades ago and it's still true (and more relevant than ever) today. An FM processor, by its very nature, compresses dynamic range and employs some form of clipping to deliver a "signature sound" and a competitively loud signal on the air. It is an unfortunate but well-accepted fact that recordings made in the past two decades have been on the decline in terms of quality, as mastering engineers seem to be waging their very own "loudness wars."

The result is source material that is hyper-compressed right out of the jewel case with only a dB or two of dynamic range at most. As if that weren't bad enough, the music is run through unsophisticated, brute-force clippers to make them louder still. The result is that the audio going INTO a processor today sounds more distorted than the audio coming OUT of an FM air chain 10 years ago! Before it even gets touched by the compressors, limiters, and clippers in the processor itself, it has been damaged. Rip a track from the modern CD of your choice and look at the waveform in your favorite editor if you need proof. Processors add more distortion still, and the resulting "music" heard on the air is nearly unlistenable. By repairing the damaged audio first, "Undo" gives Omnia.7FM cleaner and more dynamic audio to work with, which can better stand up to the rigors of on-air processing. The result is a clean, dynamic, and listenable sound on the air. In fact, audio processed by Omnia.7FM often sounds far better than the original CD.

Psychoacoustically Controlled Distortion-Masking Clipper

Clipping is typically the final stage of an FM processing chain. The majority of clipping is usually done in the final L/R audio, with additional, optional clipping available in the composite signal. The final clipper is also where the classic (and oft dreaded) "loud v. clean" tradeoff is made. When more clipping is used to gain loudness on the dial, clipper distortion becomes more and more pronounced. The clipped peaks fall back into the audio and manifest themselves as audible distortion.

There are ways to get around that problem, but they come at a price. You could back down on the clipper drive to clean up the sound, but then you lose loudness. Or, you could put more of the "heavy lifting" on the compressors and limiters preceding the clipper, but that results in an overly busy, dense sound that robs the music of life and causes listener fatigue. Some processors HAVE to resort to building excess density in the dynamics section because their simple or old-technology clippers simply aren't up to the job. The Omnia.7FM identifies clipper distortion and uses a proprietary psychoacoustic-controlled algorithm in the composite signal to mask it, effectively eliminating it from the final audio. It is so robust that it boasts an additional 3dB of high-frequency headroom and is capable of 140% L/R modulation within 100% total modulation. That means Omnia.7FM can be significantly cleaner for a given loudness level or substantially louder for a given level of quality. It comes closer to eliminating the "either/or" compromise than any other processor on the market today.

Omnia Toolbox

When Leif Claesson was first creating Omnia.9, he knew that having diagnostic and measurement tools would be necessary. The original plan was to keep them in place only for development, but he quickly realized that engineers would find great value in them as well, and decided to leave them in place.

The Omnia.7FM carries on the tradition!

Audio processing is largely a “hearing” process, but there is much to be learned by seeing what your adjustments are doing to your sound as well. Some stations still have an oscilloscope on the test bench or a spectrum analyzer at the transmitter, but it’s not always convenient (or possible) to hook up a processor to them while it’s on the air.

Even if you did so, you’re pretty much limited to monitoring only the composite output of your own station’s processing. Also provided as part of the Omnia.7FM, it means there’s no extra test equipment to buy (‘scopes and analyzers aren’t cheap) and no cables to hook up. It also means you can visually monitor the signal at the input, the output, and dozens of other points throughout the processing path so you can tell what’s happening to the audio every step of the way. As an added bonus, Omnia.7’s composite inputs can be fed from a calibrated tuner or frequency-agile mod monitor so that you can monitor the other signals in your market, too!

In addition to these tools, Omnia.7FM also includes RTA and speaker calibration tools to further assist with monitoring and fine-tuning your processing. While it is certainly good practice to listen to your station on a variety of radios and speakers as you adjust your processing, it is also good practice to have at least one set of calibrated speakers available. Otherwise, the changes you make to your processing will be influenced by listening to speakers that don’t accurately reflect the frequency response of your processing adjustments. By adding an inexpensive calibrated microphone and using the included pink noise generator and RTA, you can quickly and easily calibrate a set of speakers to use as a reference as you adjust your sound.

Speaker Calibration

If you make decisions about your processing on uncalibrated monitors, you are making choices that are influenced by the differences in frequency response present in every speaker, not to mention the coloration imposed by the room in which you are listening. Simply put, you're dealing with subjective, not objective, information. By using the pink noise generator and RTA built into Omnia.7FM and adding an inexpensive calibrated microphone, it is possible to calibrate any speaker system to deliver as flat a response as the speakers themselves will allow (small speakers still won't reproduce low frequencies as well as larger ones – the laws of physics still apply after all). With speaker and room influences removed from the equation, you are in a position to adjust your audio based only upon "the facts." When explaining this process to someone in person, this is the point in the conversation where they inevitably say, "But listeners aren't hearing my station on calibrated speakers! They're listening in their cars, at their computer, and through cheap ear buds, so I should too!" It's true — that's exactly how your listeners are hearing your station in the real world, and why it is always important to listen on a wide variety of radios in many different environments. But adjusting your processing this way is a shortcut to a lot of tail-chasing frustration and lousy audio. Let's say you listen first in an inexpensive compact car with a typical factory stereo. You notice there isn't much bass, so you adjust your processing to deliver more low end. It sounds good. Then you move into a high-end luxury car with 10 speakers and a subwoofer, and the bass is muddy, boomy, and overwhelming. Why? Because you adjusted the bass in the processor to make up for deficiencies you thought were in your processing, but in fact were in your speakers! Having at least one pair of high quality, calibrated speakers to go back to as your reference will dramatically improve your on-air sound, save you valuable time, and help preserve your sanity at the same time. (Don't worry – there are still plenty of people at your station to chip away at your mental well-being — we just don't want to be among them!).

Dry Voice Detector

We know that the human voice can present a tough challenge to an FM processor. If it's bare voice — that is, voice alone with no music mixed underneath — any distortion created in the processing really stands out. We also know that all-out loudness comes at a price: At some point, you have to give up "clean" to get "loud." Even Omnia.7FM's psychoacoustically controlled distortion-masking clipper, which really minimizes the dreaded "clean v. loud" tradeoff, can reveal some distortion on dry voice when the overall processing is set up to really push for loudness. So ensure clean voice quality in these situations, the dry voice detector first determines that the incoming audio is actually bare voice. It then automatically and inaudibly transfers more of the "heavy lifting" to the compressor and limiter sections, thereby reducing the amount of overall clipping needed to maintain the same level of loudness.

Remote Client

Every modern processor provides some means by which to control it or adjust its settings remotely, which is handy if the processor is at a transmitter site miles (and often mountains) away from the studio. Most employ web-based interfaces, which on the surface sounds convenient because it allows you to remote in from a browser on any computer at any location, but even the best of them fall short when it comes to a great user experience. They require browser plug-ins, typically feel "laggy" when viewing meters or adjusting controls, and don't always have the same look and feel as the front panel interface. Omnia.7FM's client software delivers exactly the same experience whether you're standing in front of the processor or controlling it from your PC or tablet. If you have Omnia.7FMs on more than one station in your group (who can buy just one?) you can connect to any of them through a single connection window, and can run multiple remotes simultaneously.

Provided your connection has sufficient bandwidth, you can even stream audio from various patch points within the processing chain back to the client computer. This allows you to hear what effect your adjustments have on your audio in the environment of your choice instead of a rack room or transmitter building, locations which almost never have decent monitors but offer noise in abundance!

SPECIFICATIONS

Frequency Response

- +/-0.5dB 20Hz to 15kHz, 17.5kHz in extended mode

Signal to Noise Ratio

- Greater than -80dBu de-emphasized, 20Hz to 15kHz

System Distortion

- Less than 0.01% THD below pre-emphasis, inaudible above

Stereo Separation

- 65dB minimum, 20Hz to 15kHz, 70dB typical

Digital Output Level

- Adjustable from -24.0dBFS to 0.0dBFS in 0.1dB increments

Stereo Baseband Output

- Adjustable from -2dBU to +22dBU (0.1dB increments) into 600-Ohms, 20-Ohm output impedance

A/D Conversion

- Crystal Semiconductor CS5361, 24 bit 128x over-sampled delta sigma converter with linear-phase anti-aliasing filter.
- Pre-ADC anti-alias filter, with high-pass filter at <10 Hz

D/A Conversion

- Crystal Semiconductor CS4391, 24-bit, 128x oversampled

Analog I/O

- Two balanced, EMI filtered XLR connectors

Stereo Generator Connections

- Four 75-Ohm BNC female, two inputs, two outputs

Digital I/O

- AES/EBU In & Out via XLR connectors
- Supports stereo digital audio and Omnia Direct™

Ethernet

- Shared RJ45 supporting 100 and 1000BASE-T Ethernet connections

Power Requirements

- 100-264 VAC, 47-63Hz autosensing, dual PSU

Power Connector

- Dual IEC male, detachable 3-wire power cords supplied

Power Supply

- Internal dual redundant

Environmental

- Operating: 0 to 50 degrees C
- Non-operating: –20 to 70 degrees C.

Regulatory

North America: FCC and CE tested and compliant, power supply is UL approved.

Europe: Complies with the European Union Directive 2002/95/EC on the restriction of the use of certain hazardous substances in electrical and electronic equipment (RoHS), as amended by Commission Decisions 2005/618/EC, 2005/717/ EC, 2005/747/EC (RoHS Directive), and WEEE.

Omnia VOLT®

The 4-in-1 Processing Powerhouse with Audio Processing for FM, AM, SG, HD/DAB/DRM, or Studio Applications.

OVERVIEW

Omnia VOLT offers "best in class" processing performance from the people who brought you the multiple-award-winning Omnia.11, the acclaimed Omnia.9, the power-packed Omnia 7, and the 13,000+ Omnia ONEs currently in service. With thousands of units on the air, VOLT gives you more sonic performance and processing power in one rack unit than others give you in three. Now with new Version 2.0, the best just got better!

Here's how:
With VOLT, we rewrote the rules for broadcast DSP, fine-tuning our algorithms and creating the world's best-sounding, most powerful and versatile 1RU audio processor. The QuickTweak system helps users get just the sound they want quickly. You can get VOLT on the air and sounding great in minutes, select presets that we've designed for today's stations and programming, and create a unique sonic personality with just a few nudges.

VOLT may be the most value-packed audio processor on the market, thanks to its DSP|Core firmware module system. This system means you can repurpose your VOLT to serve different needs by simply updating firmware. The capability to turn your FM VOLT into an AM unit or an HD/DAB or SG box comes built into every VOLT, and the fact that each DSP|Core firmware package is FREE is like getting four processors in one package. Modules include VOLT FM, VOLT AM, VOLT HD PRO, and VOLT SG. Each Core has a feature set that has been fine-tuned for its application, optimizing system DSP to the unique demands of different use cases.

VOLT FM - Killer Omnia FM sound featuring 6 AGC sections (5 multiband + wide-band), 5-band, time-aligned limiters, Dynamics Engine architecture, the latest low distortion clipper design by Frank Foti, and a high-quality composite stereo generator. Dozens of modern presets give you an awesome sound straight out of the box. Run studio side to feed your STL, or generate FM composite at the transmitter.

VOLT AM - Is specially designed for the challenges of AM radio. Versatile pre- and post-limiting equalization plus superior narrow and wide-band presets created for today's AM stations give you the cleanest and most powerful signal on the band. Asymmetrical Modulation and Tilt, applied after the clipper, help you get better performance even from older transmitters!

VOLT HD PRO - Is purpose-built for HD Radio, DRM, DAB, web streams, and other compressed media, as well as syndicated program and studio productions. Selectable bandwidths and crossover points, plus our exclusive Sensus algorithms, help reduce lossy compression artifacts even at low bitrates. Put it in your syndicated program production chain before the uplink: every episode will be clear, clean, and have a consistency listeners can recognize. Super-low latency for processed headphone monitoring applications. Use VOLT HD PRO as a general-purpose processor for, gain control, limiting, and program distribution.

VOLT SG (Stereo Generator) - For use in cases where you want to split your system between the main processor—usually at the station—and a dedicated MPX stereo generator with advanced composite features at the transmitter. VOLT SG can be paired with a VOLT FM, one of our other Omnia processors, or with any other processor from another manufacturer. VOLT SG uniquely features the same "One Louder" embedded pilot as the Omnia 11.

FEATURES

VOLT FM

- A new generation Frank Foti-designed Clipper for stronger on-air sound.
- Flexible Pre-Emphasis Switching. This makes it easy to fit VOLT into any existing FM airchain.
- Dual Variable Composite Outputs to feed a main and backup transmitter.
- Variable Pilot Level and Phase let you fine-tune the signal for transmission.
- Adjustable SCA input for additional services, including RDS and specialty networks.
- 19 kHz sync output to synchronize external generators.
- Bass Pre-Clipper - Fully adjustable with Tightness and Girth controls. You'll have strong, listener-pleasing bass without worrying about intermodulation distortion.
- Adjustable BS-412 Threshold and Processing for full compliance with ITU standards.
- BS-412 and Low Latency settings that can be turned on as needed.
- Automatic Mono "Dry Voice" Sensing - Ideal for FM Analog Stereo stations using extreme processing: it keeps an extra hand on the clipper, to stop distortion when the L+R channel gets boosted by mono signals.
- Low Latency FM mode - lets you fine-tune the tradeoffs between processor latency and ultra-high quality.
- Stereo Enhancement for FM Analog, without Adding Multipath - You'll get a wider, more exciting signal that jumps out of the radio.
- Optional expansion module to support Kantar watermarking
- MIB2 compliant SNMP support

VOLT AM

- Extra equalization to control or enhance highs in two critical places, both before and after the multiband limiter.
- Wide and Narrow bandwidth options and presets to let you get the best combination of coverage and quality for your station's format.
- Crossover system is re-tuned for various narrow-band options so that all five bands of processing are used within the bandwidth constraints for maximum performance at all bandwidth settings.
- Adjustable Tilt to compensate for low-frequency 'droop' in older plate-modulated transmitters.
- Asymmetrical modulation for maximum AM power.
- Flexible Pre-Emphasis Switching on audio inputs and outputs. This makes it easy to fit VOLT into any existing AM airchain.

VOLT HD PRO

- Sensus® Processing for Digital Program Streams - Omnia's exclusive Sensus algorithms predict how HD, DRM, DAB, or streaming data reduction will affect your sound. They precondition your signal, making digital compression sound better—even at low bitrates, actually reducing the distortion added by psychoacoustic compression schemes.
- Switchable bandwidths with eight choices between 4 kHz – 24 kHz. They reconfigure VOLT's multiband processing frequencies as well as limit high frequencies, to give you the cleanest signal for your distribution medium.
- Crossover system is re-tuned for various narrow-band options so that all five bands of processing are used within the bandwidth constraints for maximum performance at all bandwidth settings.
- Look-ahead wideband limiter in addition to five fully adjustable band limiters guard against quick transients that can add distortion.

VOLT SG

- SSBSC support lets you select Single Sideband suppressed carrier for more multipath immunity, less noise, and more robust HD radio.
- "One Louder" technology embeds the stereo pilot directly into the composite signal. This frees up power, giving you a full decibel more loudness—effectively, 10% additional modulation—without exceeding legal limits.
- Flexible Composite Limiter can be calibrated to control overshoots from nonlinearities in your STL.
- Pilot Phase adjustment of ±32°, compared to the 38 kHz suppressed carrier, to compensate for timing errors elsewhere in your system.
- Advanced single sideband suppressed carrier processing [SSBSC] - for reduced multipath and more robust HD transmission.

FM, AM, and HD-Pro VOLT processors commonly feature:

- **Optimized Dynamics Engine and time-aligned crossover system** designed by our best Omnia engineers.
- **Six Separate AGC Sections** - Tunable midband crossover, one wideband, plus five separate time-aligned narrow band sections, each with separate controls for every important parameter give your station the loudness and consistent sound you want!
- **Five Separate Time-Aligned Limiter Sections** - each with separate Drive, Hold, Threshold, and Attack/ Decay controls. They give you protection against over-modulation while maintaining a loud signature sound.
- **QuickTweak™ Adjustment System** - lets you fine-tune your sound like a processing genius. Get exactly the processing you want in minutes, while you're on the air, right from the front panel or a connected computer or tablet.
- **Variable Deep Bass, Phat Bass, and Warmth enhancers** - Get that meaty Omnia sound, fine-tuned the way you want.

- **Clipper Silk Adjustment** - (FM & AM) If your format is prone to treble distortion, you can add just enough Silk to clean up those high frequencies.
- **Variable High-Pass and Switchable Phase Rotator** - Special processing in VOLT's input makes sure that ultra-low frequencies, too low to be perceived as bass by listeners, don't rob you of on-air power.
- **Totally Flexible Input / Output** - Use analog, AES/EBU digital, or Livewire® AoIP inputs; analog, AES/EBU digital, Livewire, or composite outputs (FM & SG only). Adjust channel balance and correct polarity separately on each input. Save and recall input/output setups for different applications. All outputs are always active, regardless of input type.
- **Switchable Insert Points for Voltair®, Watermark Encoders, or Other Downstream Encoding** - Optimize your airchain and eliminate the need for external pre-processing! You can feed encoders with a pre-processed signal from VOLT's multiband AGC and limiters, so your encoder sees a stronger, more reliable signal. Then feed the encoder's output back into VOLT for post-encoding clipping that protects you against overmodulation.
- **Automatic "Failover" signal switching** - Designate a backup input to use if your main signal drops out or STL fails. Switch to this source automatically, with adjustable sensitivity, or trigger it as needed.
- **Graphic User Interface** - is easy to navigate, but gives you the deep level of control you need.
- **Built-in HTML-5 Server** - for full control from any modern browser, tablet, or smartphone… without special plug-ins.
- **Rugged 1RU Construction** - fits any control room, technical center, or transmitter shack, with easy-to-see LED meters.
- **Cool Running, Fanless Operation** - VOLT can even be used near live mics.
- **Flexible Pre-Emphasis Switching** - makes it easy to fit VOLT into any airchain.
- **Built-In Tone Generator** - provides for quick setup and calibration.
- **Multiple Selectable Bandwidths for Digital Data Streams** - More than simple low-pass filters, HD Pro also modifies the VOLT's crossover to get the best results for any bitrate.

IN DEPTH

Nail Your Signature Sound Faster with QuickTweak™

Whether you are a processing novice or expert, Omnia VOLT gives you the tools to create a superior signature sound. Choose from some of the best factory presets available, designed by Omnia's processing experts, and by our favorite "insider" guest programmers. For those who want to push beyond stock presets, Omnia's new QuickTweak system lets you fine-tune your sound quickly. For experts, drill down into deeper parameter adjustments. Get exactly the processing you want, while you're on the air, whether you're at the front panel, or sitting in your car controlling VOLT over the web.

Nobody knows processing like Omnia. We've designed QuickTweak based on our decades of experience and market leadership, algorithmically linking complex and interactive parameters to create a core set of "meta" controls.

- QuickTweak is easy to understand: You can tune it by ear, and hear the results instantly.
- QuickTweak's six master controls allow millions of recallable variations, right from the front panel.
- You can use QuickTweak on the factory presets, or on your own custom presets.
- You can save your own settings after using QuickTweak to easily A/B compare preset modifications.
- Any preset can be adjusted with QuickTweak, or you can fine-tune using even deeper control layers. Presets can always be refined, then saved under a new title.
- Share presets by importing or exporting them with others in your company. Back-up preset files on common media.

Total Versatility with DSP|Core Firmware

VOLT's DSP|Core firmware modules rearrange and modify VOLT as your needs change. DSP|Cores aren't extra cost add-ons! Download the functionality you need for free, install the DSP|Core firmware package from a connected computer, and reboot. It's that simple.

- Use VOLT for FM Analog Stereo at the station, with high-quality baseband clipping to feed uncompressed STLs, or at the transmitter, with dual composite outputs.
- Use VOLT for AM Broadcast, with purpose-built presets for the challenges of AM radio. VOLT's Tunable Asymmetrical Modulation and Tilt controls help you get modern results, even from older transmitters!
- Use VOLT for Studio and Program Production or Syndication. It comes with the tools and presets you need for modern production styles.
- Use VOLT for HD/DAB/DRM/Web Streaming. Our exclusive Sensus algorithms reduce compression artifacts even at low bitrates.
- Use VOLT as a standalone FM Stereo Generator at the transmitter for direct connection to transmitters.
- Use VOLT for low-latency FM Stereo to comply with local regulations, using a high-efficiency clipper that's optimized for this kind of broadcast.

Front Panel

Rear Panel

SPECIFICATIONS

Frequency Response

- User selection of flat, 50 μs, or 75 μs pre-emphasis curve within ± 0.50 dB, 30 Hz to 15 kHz.

Signal-to-Noise Ratio

- Audio >95 dB analog, >120 dB digital I/O.

System Distortion

- Less than 0.01% THD, 20 Hz – 7.5 kHz (second harmonic distortion above 7.5 kHz is not audible in the FM system).

Latency

- 16ms nominal, +-0.5ms depending on IO selection. Low Latency FM version 10ms

Input / Output

- Composite: Output impedance 75Ω, single-ended and floating over chassis ground. BNC connectors with EMI suppression. Maximum cable 100' / 30M RG-58U.
- Output level: Separately adjustable for each of two outputs, 0V - 10V in 0.05V steps.
- Pilot Level: Adjustable from 4.0% to 12.0% in 0.1% steps and OFF. Pilot Stability: 19 kHz, ± 0.5 Hz. S/N: -85 dB typical, 75 μS de-emphasized across 15 kHz, at 100% modulation Distortion: < 0.02% THD 20 Hz – 15 kHz, 75 μS de-emphasized @ 100%.
- Stereo Separation: > 65 dB, 30 Hz – 15 kHz. Linear Crosstalk: > -80 dB, main to sub or sub to main channel @ 100%. Non-linear Crosstalk: > -80 dB, main to sub or sub to main @ 100%. 38 kHz Suppression: > 70 dB @ 100%. 76 kHz Suppression: > 80 dB @ 100%. Pilot Protection: > -65 dB relative to 9% pilot injection, ± 1 kHz. 57 kHz (RDS/RBDS) Protection: > -50 dB.

Analog

- Left and Right Stereo on EMI-suppressed XLR-3, balanced with "pin 2 hot."
- Input: Electronic balanced, impedance 10kΩ, nominal +4 dBu, max +22 dBu.
- Output: Impedance 20Ω for >600Ω load, +4 dBu nominal, +22 dBu peak. Converters: 24 bit, 128x oversampled with linear-phase anti-aliasing filter.
- Crystal Semiconductor CS5361, 24-bit 128x over-sampled delta sigma converter with linear-phase anti-aliasing filter.
- Pre-ADC anti-alias filter, with high-pass filter at <10 Hz.
- Delta sigma converter with linear-phase and anti-aliasing filter.

Digital

- Stereo per AES/EBU standard, 24-bit resolution. Input locks to any rate 32 kHz – 108 kHz. Output locks to input, internal 48 kHz, or separate external AES/EBU "digital black" reference 32 kHz – 96 kHz.

Audio over IP

- LiveWire Audio and control over IP, on the same RJ-45 used for Ethernet control.

Remote Control

- GPI: EMI suppressed DB-9 at logic levels, +5 V and ground supplied. Ethernet: 10/100BaseTX.
- Ethernet on EMI-suppressed RJ-45. TCP/IP control via HTML-5 internal web server, password protected. Manual addressing and port selection.

SNMP

- MIB 2 compliant SNMP support for remote monitoring and control

Electrical/Physical

- Power: 100 - 250 VAC, 47-63 Hz. < 40 VA. Typical draw 12W RMS, maximum 15W RMS. Internal supply with overVOLTage and short circuit protection. Meets EN55022, EN55011 Level B Conducted Emissions. EN61000-4-2, -3, -4, -5, -6 level 3 immunity compliant. Full international safety approval. CE marked. EMI suppressed IEC male connector. Detachable 3-wire power cords supplied for US and European use. Temperature: 32° to 122° F / 0° to 50° C for all operating VOLTage ranges.
- Humidity: 0-95% RH, non-condensing.
- Dimensions: 19" wide x 1.75" high x 16" deep (48.26cm x 13.335 cm x 40.64 cm) including connectors. Unit requires one EIA rack space for mounting.
- Shipping Weight: 12 lbs. / 5.5 kg

Regulatory

- North America: FCC and CE tested and compliant, power supply is UL approved.
- Europe: Complies with the European Union Directive 2002/95/EC on the restriction of the use of certain hazardous substances in electrical and electronic equipment (RoHS), as amended by Commission Decisions 2005/618/EC, 2005/717/ EC, 2005/747/EC (RoHS Directive), and WEEE.

OmniaSST
Audio Processing Software

OVERVIEW

This full-featured, professional audio processing software transforms Windows PCs into high-end audio processors. Based on a powerful processing engine designed by Hans van Zutphen, OmniaSST is the first processor to support the Telos Alliance's revolutionary Omnia μMPX® codec. μMPX, a specialized codec purpose-built for FM radio, is able to transport high-quality Multiplexed FM signals over a small 320kbps data pipe. Reduced data requirements mean high-quality multiplexed audio and RDS signals can be directly routed over IP from your processor into a decoder application, opening up tremendous possibilities for efficient MPX audio transport.

Omnia SST audio processing software is the only processor with the 3 Ds for your audio needs: Declipper, Delossifer, and Dehummer. With Declipper, OmniaSST repairs incoming audio, optimizing it before it hits compression, limiting, and final processing stages, Delossifer repairs the sound of lossy compressed audio such as MPEG2/MP3 material. While Dehummer removes unwanted sounds such as 50/60 Hz hum from bad cables. These remarkable pre-processing tools result in clean, loud, and open sound.

Finally, OmniaSST audio processing software includes purpose-built Omnia presets, FM pre-emphasis, stereo and RDS encoding, and shares the same FM reception-improving composite final clipper as found on the Omnia 9SG. Using an i7 class processor, latency can be reduced to as low as 5ms. A streamlined, HTML-5 based GUI means easy, intuitive operation.

FEATURES

Audio Processing Features

- **Low-Latency Performance –** Using an i7 class processor, latency can be reduced to as low as 5ms.
- **The Perfect Declipper –** A unique filter, also found in our Omnia.9 and 11 flagship processors. It detects and reconstructs clipping-damaged audio (distortion caused by too-loud input levels and overly aggressive mastering), prevalent in music.
- **Delossifier** – Improves the sound of lossy encoded audio, such as MP3- and MPEG2-encoded audio.
- **Dehummer –** Removes constant tones such as a 50/60 Hz hum on the fly.
- **Hiss Reduction Filter –** A powerful hiss-reduction filter that tackles hiss while leaving adjacent content alone. Especially useful when playing older songs or other noisy content.
- **Natural Dynamics –** Boosts transients. This mainly boosts percussion instruments, adding more dynamics, which gives the rest of the processing chain more to work with.
- **Automatic Gain Control (AGC) –** The ITU.1770-based AGC keeps the sound compliant to a defined loudness level.
- **Bass Harmonics –** This filter increases the bass level without increasing the maximum amplitude of the bass, and adds harmonics that make the bass more audible on speakers with poor bass reproduction.
- **Phase Delay –** Purposely adds phase-nonlinearity. Can be used among others to generate a more boomy bass.
- **Bass in Your Face –** Intelligently adds missing subharmonics.
- **Absolute Highs –** Reconstructs missing highs if damaged or removed by lowpass filters.
- **2-9–Band Multiband Compressor / Limiter, x2 –** OmniaSST has two multiband compressors that can be used in series. They have a very natural, pleasant, and non-fatiguing sound. Depending on the content, they move very quickly when they need to and slowly when they can.
- **Phasing Error (AZIMUTH) Correction Filter –** Detects and repairs phase-shift errors that are often present on tape recordings and some badly mastered CDs. These phasing errors can make listening very unpleasant, and can also cause severe artifacts when converting sound to mono, encoding it using a lossy codec, or when playing it on a surround system. This filter automatically detects and repairs such errors. For FM transmissions this can reduce multipath distortion and improve the audio quality when the radio blends to mono.
- **Stereo Booster –** Boosts the stereo width. It can be configured to not alter the total sound content.
- **Stereo to Mono Conversion –** Converts stereo to mono without removing sounds that normally cancel out during a mono sum. This leaves mono sounding just as full and powerful as the original stereo sound.
- **Distortion Masking Clipper –** The same clipper that's available in the Omnia.9sg. Now with a louder and cleaner "Rules are made to be broken" mode.

- **Lowpass Filter –** The extremely steep phase linear lowpass filter can be used to cut off frequencies that cannot be broadcast, removing unneeded signals from streaming audio, FM and AM transmission.
- **Auto EQ —**Automatically control the spectrum before the AGC, making it possible to use the multibands to precisely control how deep each band will go down and thus how much density you add, regardless of how the input material sounds.
- **μMPX—**Allows you to send a full multiplexed FM signal with RDS through a narrow 360kbps pipe.
- **Automatic Phase Repair—**Utilizes advanced DSP to allow the restoration and control of pure anti-phase signals, rivaling processors costing thousands more.

FM Transmitter Features

OmniaSST audio processing software is the first product to generate and decode Omnia's proprietary μMPX™ multiplex audio codec, allowing you to transport high-quality Multiplexed FM signals over a small 320kbps data pipe, slashing bandwidth needed for MPX by 83%! Reduced data requirements mean high-quality multiplexed audio can be economically routed from an audio processor, over IP, and directly to a decoder application, opening completely new possibilities for studio to transmitter links.

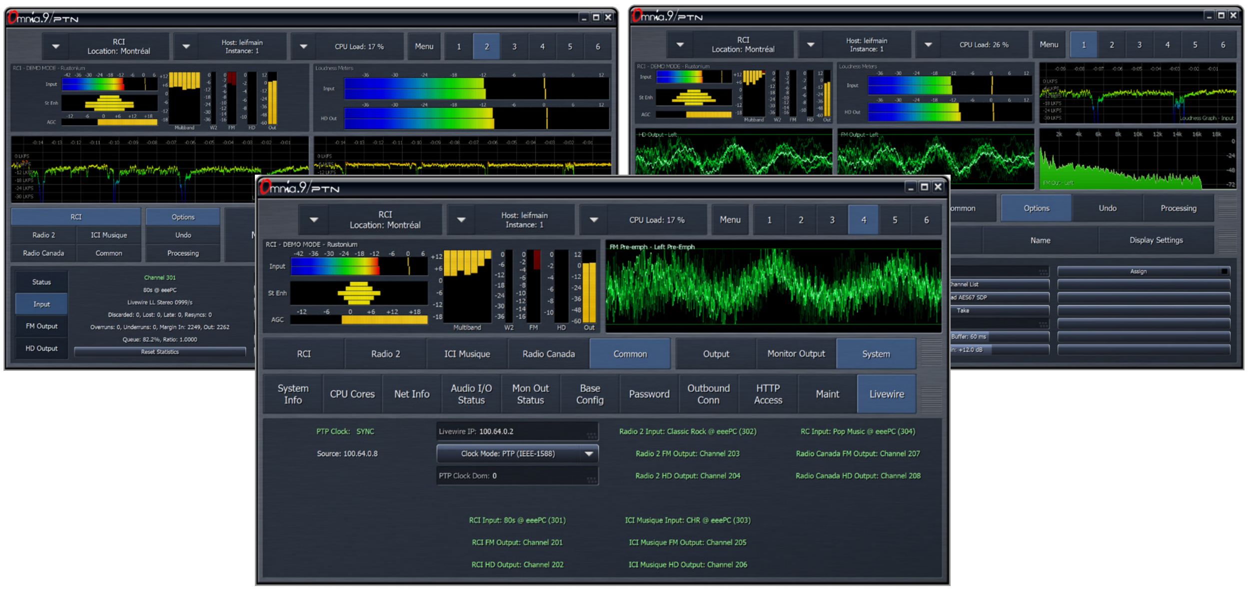
μMPX includes stereo and RDS data, which can be sent to the FM transmitter using a good-quality 192kHz capable sound card. No separate hardware stereo coder or hardware RDS encoder is needed.

The following FM transmitter specific options are provided:

- **FM Pre-Emphasis Filter –** Pre-emphasizes the sound, which is needed for FM transmissions.
- **Flexible Outputs** – Flexible outputs mean OmniaSST can output both the pre-emphasized and the de-emphasized sound, or the pre-emphasized sound and a partially separately processed streaming sound.
- **FM Stereo Encoder –** The software FM stereo encoder generates a high-quality MPX signal including stereo information, which can be fed directly into an FM transmitter.
- **FM RDS Encoder –** The software RDS encoder enables broadcasting RDS text without using a separate hardware RDS encoder.
- **Composite Clipper –** The composite clipper in OmniaSST can squeeze out multiple dBs of extra loudness with highest sonic quality. The stereo pilot tone and RDS signal are not affected by this filter, and the upper and lower sideband are perfectly symmetrical, so the MPX signal that comes out complies perfectly to loudness standards.
- **ITU-R.BS-412 Limiter –** Mandatory for several European countries. The BS412 limiter in OmniaSST has virtually no effect on the sound (no extra clipping, no pumping, no gain riding), but still holds the output level up to the allowed maximum.
- **ITU-R SM.1268 Clipper, Multipath Clipper –** OmniaSST contains a built-in software based exciter and spectrum analyzer, which are used to analyze what the output signal will look like after FM modulation. The spectrum analyzer feeds back into the clipper in multiple stages, to very precisely control the resulting spectrum, strictly adhering to and even exceeding what ITU-R SM.1268 recommends.

Omnia.9 PTN

High-Density Audio Processing Software



OVERVIEW

Omnia audio processors are the leading brand by a wide margin for both terrestrial and streaming broadcasters around the world. Adaptable to any environment, Omnia offers scalability and flexibility to match your specific broadcasting needs.

Meet Omnia.9 PTN, the high-density audio processing software solution designed with the flexibility to meet the rapidly changing infrastructure needs of broadcasters as they transition to virtualized environments. Omnia.9 PTN is a custom solution for high-density server-based (virtual) systems for customers with a large volume of signals that need to be processed. Talk with our sales team to design your Omnia.9 PTN solution based on your specific needs.

Scalable, Customizable, Centralized

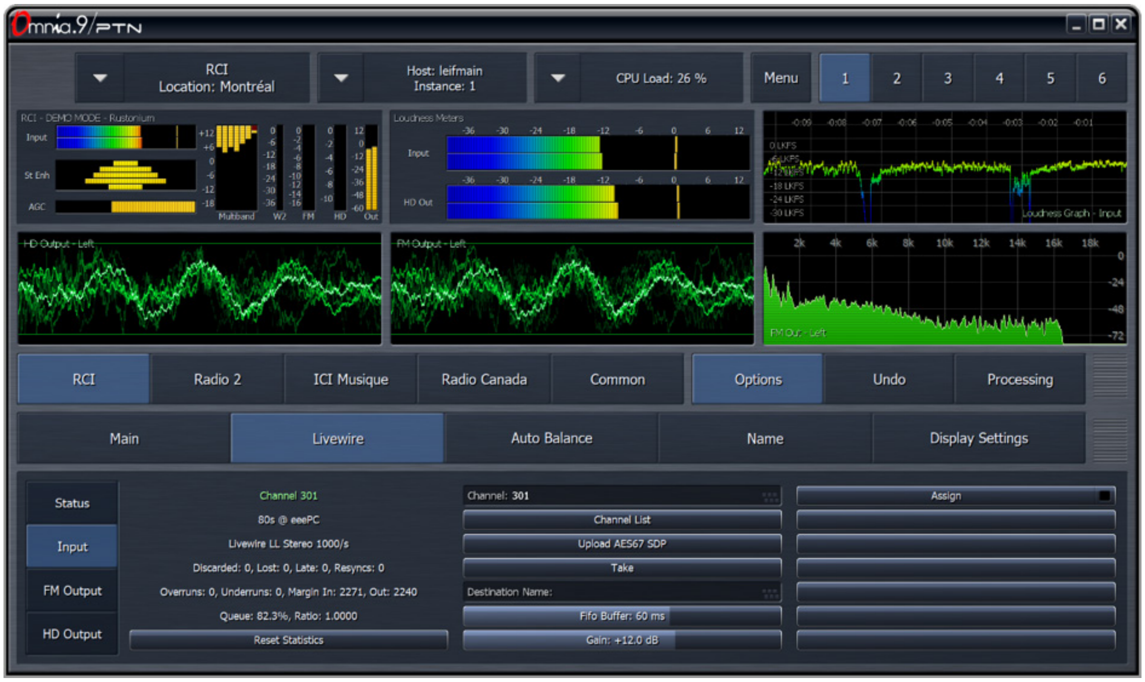
Omnia.9 PTN combines product modules in a new and unique way to deliver the software-only, high-density solution needed by today's broadcasters. Each system can be customized to your needs, which allows you to make changes as your requirements change. Omnia.9 PTN can be programmed for FM and streaming, for example, giving you the flexibility to quickly add specialty channels. As you add more streaming channels to your network, Omnia.9 PTN adapts to fit your needs.

Centralize your processing by using Omnia.9 PTN at the head end and transmit either L/R audio to each transmitter location or the full composite signal to the transmitter using the new Omnia MPX node.

Virtualized Audio Processing

Each stereo program is processed by its own unique processing engine, allowing each program to be uniquely tailored to suit the program material, audience, and delivery method. A variety of factory presets are included and can be used as-is or serve as starting points for custom user presets.

"Basic" adjustment mode greatly simplifies dialing in the desired sound by combining multiple processing parameters with a handful of controls. For instances where it is necessary to access every available individual control, "Intermediate" and "Expert" modes are available.



FEATURES

Audio Processing Features

- **Undo** – Our unique method of processing audio that has been clipped during the mastering process and/or hyper-compressed dynamically as is the case with most music recorded in the past 20 years. Undo is a two-stage process that consists of a declipper that reconstructs audio peaks (thereby removing the resulting distortion present in the original content) and a multiband expander that adds dynamic range to highly compressed material.
- **Phase Processing** – Including an adjustable phase rotator to help eliminate distortion in asymmetrical voices and a phase scrambler to minimize distortion on certain instruments with strong odd-order harmonics.
- **Downward Expanders** – Fully adjustable multiband expanders help minimize noise present in recorded audio sources or via microphone in noisy studio environments.

- **Input AGC** – The first processing gain stage, typically used to compensate for varying input levels from the board or automation playout system. Features a sidechain control EQ circuit to make the AGC more/less sensitive to certain frequencies, useful for ensuring strong bass or dynamic female vocals don't cause "hole punching" artifacts in the processed audio.
- **Wideband AGC 1/2/3** – Additional wideband compressors that can either be "stacked" atop the Input AGC for additional gain control, located after the multiband processing stage, or used as dedicated bass compressors to customize bass texture.
- **Parametric EQ** – Six bands of fully adjustable equalization to customize the EQ curve ahead of the multiband AGC stage.
- **Stereo Enhancer** – A multiband enhancer that can fully manage the stereo image by widening material that has less L-R content and narrowing material that has more, providing a very customizable and consistent stereo image regardless of the original content.
- **Multiband AGC/Limiters** – User-selectable between two and seven bands to provide an output that is more faithful spectrally to the input audio or extremely consistent spectrally regardless of the input. A dedicated peak limiter rides atop each AGC band.
- **Band Mix** – The final means of adjusting spectral balance before the final peak limiters, providing further customization of the sound.
- **Final Limiter** – A two-band "brick wall" look-ahead limiter to deliver precise peak control.

IN DEPTH

Input Routing

Each AES67 source available on the network is selectable from a Channel List and can be routed to any processing engine.

Customized Displays

Our unique “NfRemote” software client runs on any Windows computer including tablets and provides access to all system, I/O and processing parameters anywhere a network connection is available. Multiple users can connect to the same processor at the same time and a single user can connect to multiple processors simultaneously, making remote management of multiple program paths easy and efficient.

NfRemote also features a highly customizable display to facilitate setup and processing adjustment. Omnia.9 PTN also provides information about input and output levels, processing activity, LKFS loudness readings and graphs, and signal-specific analysis with tools such as a digital oscilloscope, FFT spectrum analyzer, and audio frequency RTA (real time analyzer). Up to six unique display pages can be built for each processing engine.

Livewire+™ AES67 and SMPTE ST 2110

Telos created its original Livewire technology back in 2001, but we also intentionally began the process of creating the AES67 standard by initiating the proposal of the standards project to the Audio Engineering Society in 2008.

Whereas AES67 strictly focuses on the audio stream format, Livewire can be thought of as a more complete network-based digital ecosystem for creating a "facility over IP" model that includes control, advertising, discovery, and GPIO contact closures over IP.

We are fully committed to embracing and implementing SMPTE ST 2110-associated protocols IS-04 and IS-05 for discovery and control, and ST2022-7 for redundant stream support.

Audience Measurement and Watermarking

The Telos Alliance currently supports a wide variety of watermarking tools and technologies not only through the Omnia brand for radio but via the Linear Acoustic and Minnetonka Audio brands for television. These include Nielsen watermarking for television, Kantar Media (formerly Civolution) watermarking for radio and television, and Verance watermarking for television.

Configuration and Control

In addition to the NfRemote client application, Omnia.9 PTN can be controlled and monitored via a comprehensive HTTP-based API. All functions available through the NfRemote application are available through the HTTP API. Omnia 9 PTN also supports Ember+ capability along with IS-05.D.

Changes with Your Application

With Omnia.9 PTN and the suite of Telos Alliance customised software solutions, installing and deployment across your network has never been easier. Install and deployment time is much less than compared to traditional hardware-based solutions when rolling out a network wide change or upgrade. As you outgrow your server's capability, the cost of a server upgrade is a fraction of the cost of upgrading hardware across the network.

Omnia µMPX® Standalone Software

Audio Processing Software

OVERVIEW

Omnia µMPX is a specially designed audio codec that is able to transport high-quality Multiplexed FM signals (including RDS data) over a relatively small 320kbps data pipe. This specialized codec is purpose-built for FM radio. By reducing data requirements, high-quality multiplexed audio can be economically routed from an audio processor, over IP, and directly into an exciter. Lower bandwidth IP connections and narrow band STL channels can now be put into play to transport MPX signals.

OmniaSST was the first processor to include integrated µMPX capabilities. Omnia.9 now offers optional µMPX encoding capability.

Standalone Windows Encoder and Decoder applications are now available, and you can demo the software by downloading the Omnia SST installer. Unlicensed versions serve as a great "work bench" demo of the actual applications, but until a license is purchased, these versions play periodic tones and are not suitable for air. Refer to the SST manual µMPX section starting on page 59 for more information on configuration. Please note that to order a license you will need to generate a license key from the exact machine you plan to run the software on.

Omnia® MPX Node

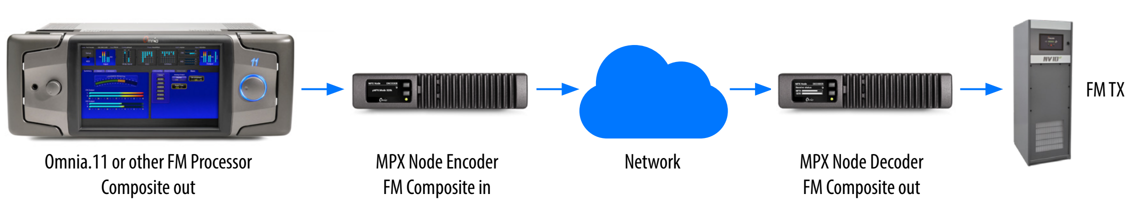
FM Multiplex / RDS over IP at data rates as low as 320kbps.

OVERVIEW

The Telos Alliance® revolutionized the modern broadcast facility with the invention of Audio over IP. We proudly extend the AoIP revolution we began in the studio out to the transmitter with the Omnia MPX Node.

Like its namesake—the classic Axia® xNode™—Omnia MPX Node is a building-block technology that helps stations leverage the growing power and capability of data networking. The Omnia MPX Node is the first purpose-built hardware codec capable of sending or receiving full FM signals at data rates as low as 320 kbps utilizing the Omnia µMPX™ algorithm, ideal for networks with limited capacity (including IP radios). MPX Node makes peak-controlled L/R baseband, stereo pilot, and RDS data routable from a studio to one or many FM transmitters.

By transporting an FM composite signal rather than left/right audio, broadcasters can keep their on-air processing and RDS encoding at the studio, then deliver a transmission-ready, peak controlled FM multiplex signal directly to an FM transmitter without the need for transmitter-side peak limiting or stereo generation. The MPX Node is available as either an encoder or decoder, and a pair of units creates an end-to-end system.



FEATURES

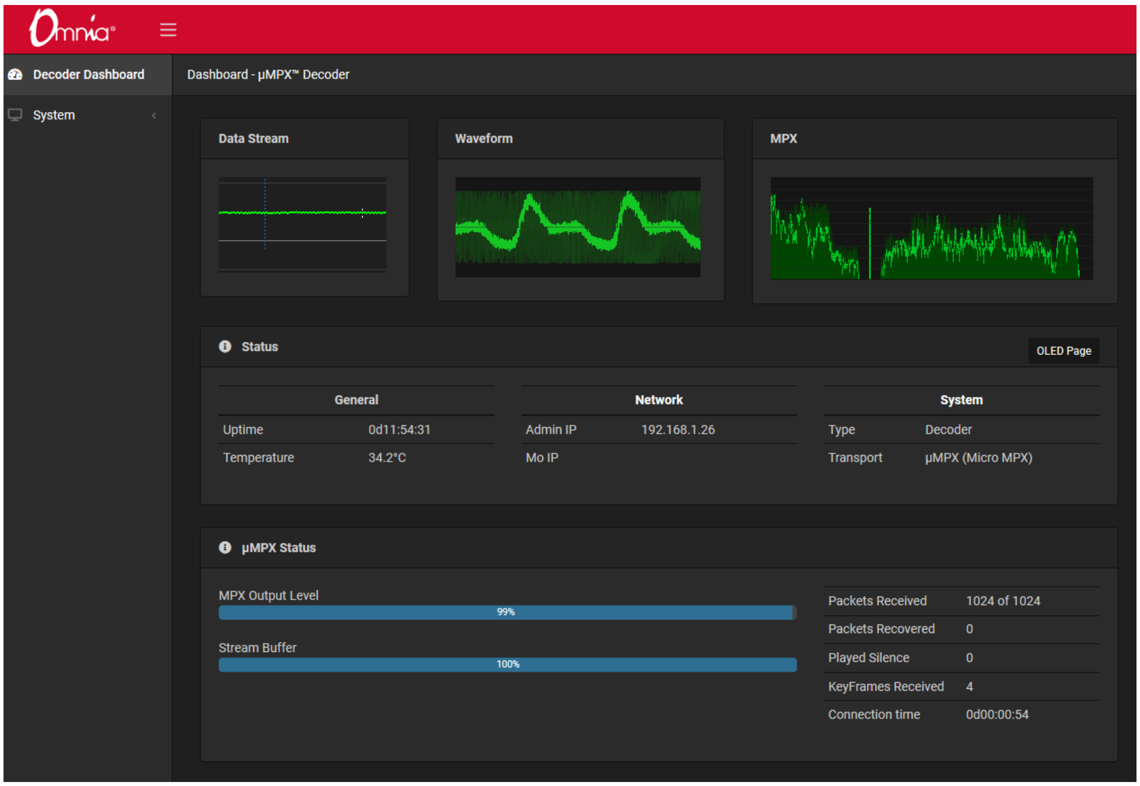
- MPX Node is available as either an Encoder or Decoder
- Omnia µMPX algorithm can transport FM composite (MPX) and RDS/RBDS at selecta variable rates from 320 kbps up to 576 kbps —perfect for IP radios and DSL IP circuits and other lower bandwidth connections
- Built-in, configurable error correction
- Processor agnostic design: Encoder can be fed by any brand of FM audio processor
- Omnia.9 or OmniaSST™ processing software can serve as the Encoder front end
- MPX Node Decoder outputs a standard analog MPX signal to feed any FM transmitter
- Full FM signals (including RDS/RBDS) can be sent over IP networks
- 5 In x 5 Out GPIO
- A single Encoder can transmit the same signal to multiple decoders
- Separate Ethernet ports support independent, dual networks
- Redundant networking allows the same signal be sent/received over two ISP's
- Simple HTML.5-based web GUI allows you to control MPX Node from your desktop, tablet, or smartphone
- Dual bank firmware update design, for secure upgrading
- Integrated test signal generator
- Cool-running, fanless operation
- Half RU, xNode form factor

IN DEPTH

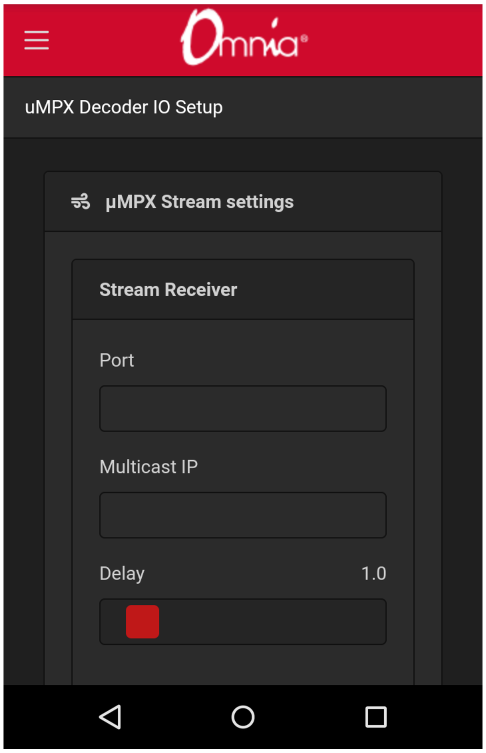
Advanced GUI

The Omnia MPX Node's HTML5-based GUI provides powerful, scalable, and secure control of the unit on all modern browser devices, from laptops to tablets to smartphones. Password-protected, continuously updated displays provide real-time metering and instant status of all parameters and settings.

PC Browser Control



Scaleable Control on a Smartphone Display

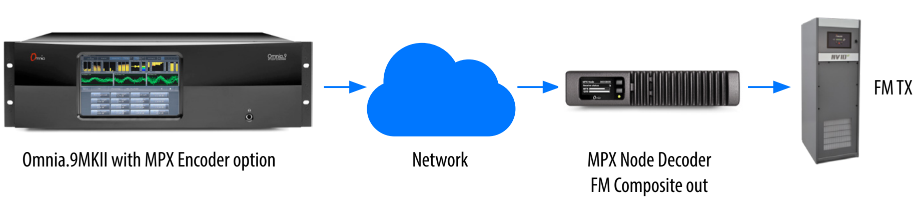


APPLICATIONS

Your Processor Is the STL

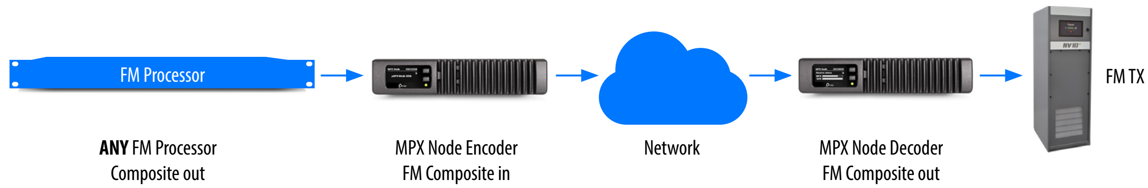
Imagine if your FM processor was also the front end of your transmission link. With an Omnia.9 at the studio, a stable IP link, and an MPX Node Decoder, you have a complete STL!

In this scenario, your Omnia.9 FM processor (with μMPX license installed) can be located at the studio. With IP addresses correctly set, Omnia.9 routes MPX/RDS data to one or many Omnia MPX Node Decoder(s) at one or more transmitter sites. The MPX Node (Decoder) then feeds your FM transmitter with a standard analog MPX signal. The Omnia MPX Node preserves all the complex peak limiting, stereo generation, and RDS (if provided), as if your FM processor were at the transmitter. The result is less equipment, fewer steps, and fewer points of failure between your processed signal and your transmitter.



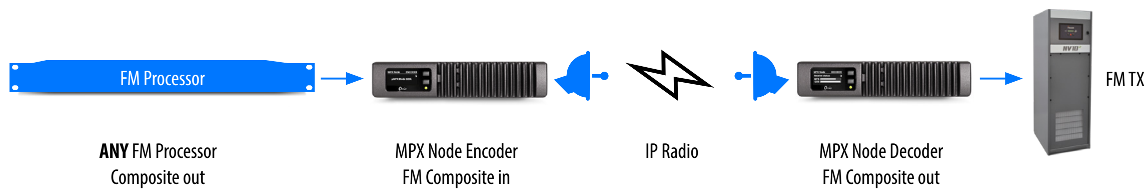
Works with Any Audio Processor

If you don't have an Omnia.9, no worries. A pair of MPX Nodes is all you need to transport the MPX signal (and RDS data) from your FM processor to your transmitter. MPX Node works with any FM processor with a stereo generator to transport a composite (MPX) signal over IP.



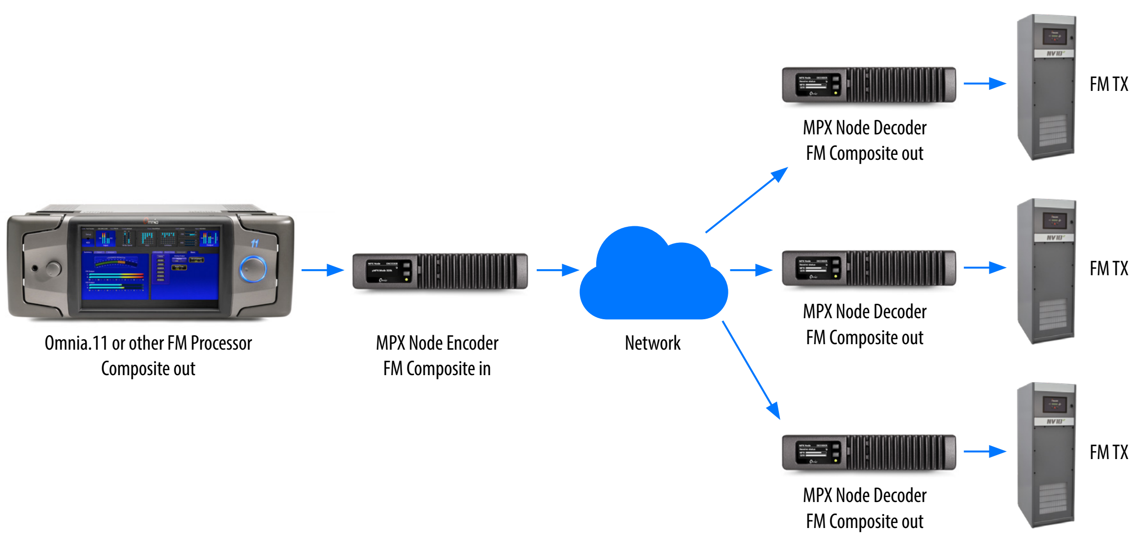
Backup STL or Translator Link

The Omnia MPX Node makes a perfect backup link to your main studio or translator. With µMPX, lower bandwidth connections, such as IP radios can be employed to get a composite signal from studio to transmitter, without the need for further peak control or stereo generation on the transmitter end.



Multi-Frequency Networks or Multiple Sites

With UDP (unicast) distribution, a single encoder can feed multiple decoders. Your FM signal can be routed point to multipoint over different IP connection types and carriers, including over the Internet. A base unit can send 2 streams. Up to 16 streams are supported with 2 and 4 stream licenses.



Multicast Support

With an optional license, Decoders can subscribe to multicast streams generated by MPX Node encoders, for private networks supporting multicast traffic.

Redundant Network support

With 2 independent network ports and internal stream replication, duplicate streams can be sent over 2 separate IP links from an Encoder. In the event of packet loss on one carrier, data is seamlessly spliced in from the second link on the Decoder for full redundancy.

SPECIFICATIONS

- Analog composite FM in/out on (BNC connector)
- Composite connector impedance 5 or 75 ohm (switch selectable)
- Net 1 port provides 10/100/1000 Ethernet
- Net 2 port provides 10/100 Ethernet
- Both Network ports support MPX transmission, web control and redundant streaming
- 5 in x 5 out GPIO on a DB15 connector
- 100-240 VAC input 50-60 Hz external power supply with 12VDC, 3.4A output on locking connector
- 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
- Operating temp range 0 to +40 degrees C, <90% humidity
- Shipping weight 7 lbs. (3.2 kg.)

Omnia VOCO® 8

Up to Eight Individually Processed Mics Networkable Through an Entire Facility

OVERVIEW

VOCO 8 is more than just a simple mic processor--It is a complete solution for managing microphone audio throughout your entire facility with a comprehensive set of tools such as:

- "Dominate-It" where the host mic can always be the dominant voice.
- Multiband processing
- Studio grade mic preamps with phantom power
- Eight line-level inputs
- "Session Recall" for convenience
- Livewire+™ AES67 support

The Omnia VOCO can adapt to nearly any voice or microphone, with factory presets tuned for both male and female voices and various degrees of processing.

FEATURES

Processing

- De-Esser
- 3-band Noise gate
- 3-band Processing
- 4-Band EQ
- Brick Wall Limiter

Processing Chain Extra Features

- Low Pass / High filters
- Phase rotation
- Dominate-It (reduces the level of other microphones by a predefined level whenever microphones with the Dominate-It feature are active.)
- Dual mix buses
- Preset sharing between networked VOCO units
- Share one VOCO between multiple studios
- Session Recall
- Control via external automation systems with “Link & Share”
- 192 kHz Sampling Rate for Processing
- Processing latency as low as 3 ms

Inputs

Eight high quality microphone preamplifiers, each switchable to line level analog input, four stereo AES/EBU inputs (eight mono), Livewire+™ AES67 AoIP input

Outputs

Line level analog, AES/EBU, and Livewire+™ AES67 AoIP

Bus Mix

Mix multiple microphones onto one of two mix busses to simplify console channel allocation, or use channels independently.

IN DEPTH

A User Friendly Control Interface

- Control all Mics on one screen
- Change all settings from one single screen. No more opening and closing windows to go from one function to another.
- "Basic" mode and "Advanced" mode: "Basic", allows selection of sessions and presets without direct access to processing parameters. "Advanced" mode, gives full access to all processing parameters in a single interface. Input and output VU meters along with de-esser and gate LEDs give access to channel activity at a glance.
- Unlimited "Undo/Redo" tracks all changes and allows them to be saved as new presets.
- "Compare" function allows quick comparison against a "reference" preset.
- Omnia Remote Gateway software works with Microsoft Windows XP SP3, Windows 7 (32 and 64 bit), Server 2008 R2, and Debian Linux.

GUI #1: Studio Mode GUI

"Studio Mode" gives access to all processing parameters for all 8 channels in a unified interface for "power users" who want the ultimate in control.

GUI #2: Live Mode GUI

"Live Mode" allows quick selection of sessions and presets in just a few clicks for fast-paced live show environments without giving direct access to processing parameters.

HQSound 192 kHz

Omnia VOCO 8 features the HQSound 192 kHz algorithm. This high resolution algorithm provides a large range of gain control without the typical "pumping" or "smashed" sound that can occur with high levels of compression. The result is a robust microphone sound sure to please even the most critical ears.

Effective 3-band noise gate

In voice processing, to get an efficient noise gate on all voices with one preset is impossible. This is mainly due to differences in levels and consistency between voices. With VOCO 8 it is possible to create a preset for each talent. This is a key point for a perfect noise gate efficiency. Working in 3-band is a real advantage. In noise gate, bands are able to work independently or in a Master/Slave scenario. This helps to isolate noise coming from table and doors.

S.I.S – Sound Impact System

A part of the HQSound 192 algorithm, S.I.S preserves attacks automatically for maximum voice impact.

Preset Sharing

Another unique feature, preset sharing allows users to synchronize presets and all changes on an unlimited group of VOCO 8 units. Preset sharing will automatically update all VOCO 8 processors without the need to load presets on each unit manually. In addition, whenever a new preset for a specific host is created in one studio, it will automatically be available in all other studios.

Multi-Studio Management

The Omnia VOCO 8 can process up to 8 microphones. Thanks to Multi-Studio mode, the Omnia VOCO 8 can distribute these resources across several studios. For example, if you have two studios each with three microphones, a single VOCO 8 can be used to independently handle the microphones for each studio. Each studio can also save and recall its own sessions for rapid configuration.

Security

The VOCO 8 features a full set of security features to prevent unauthorized tampering with presets and configuration. Multiple users and access levels are supported across all studios.

SPECIFICATIONS

Mic Input

- 8 channels, XLR
- +48v phantom, switchable
- Source impedance: 150 Ohms
- Input impedance: 4000 Ohms
- Level Range: -75 dBu to -20 dBu

Line Level Input

- ¼" (6.33mm)
- Level: +4dBu or -10dBu

Digital Input

- Quantity: 4 stereo (2 channels per AES/EBU input)
- Standard: AES/EBU
- Sampling Rate: 32 to 192 kHz - 24 bits
- DB-25 using Tascam® format

Livewire+™ AES67 Input

- Quantity: 8
- Type: Livewire (Standard or Live stream) & AES67
- Level: Adjustable in Omnia VOCO user Software
- Connector: Ethernet 100BASE-T

AES/EBU Sync

- Internal digital sync from high precision clock source.
- External reference supported from any AES/EBU input port.

Digital Output

- Quantity: 4 stereo (2 channels per AES/EBU output)
- Standard: AES/EBU
- DB-25 using Tascam format

Livewire+™ AES67 Output

- Quantity: 8
- Type: Livewire (Standard or Live stream) & AES67
- Level: Adjustable in Omnia VOCO user Software
- Connector: Ethernet 100BASE-T

GPI Interface

- Connector: Standard DB-15

Audio Performance

- Processing delay: 3 ms
- Frequency response: 10Hz - 22 kHz +/-0.2dB
- Distortion: <0.2% THD

Compatible Operating System for Remote Control Software

- Omnia Remote Gateway software works with Microsoft Windows XP SP3, Windows 7 (32 and 64 bit), Server 2008 R2, and Debian Linux.

Omnia VOCO to Client Communication Interface

- TCP/IP: Client (Remote via Ethernet)
- Link & Share: 100% of parameter are accessible through telnet protocol

Power Requirements

- 100-264 VAC, 47-63Hz autosensing, dual PSU

Power Connector

- Dual IEC male, detachable 3-wire power cords supplied

Regulatory

North America: FCC and CE tested and compliant, power supply is UL approved.

Europe: Complies with the European Union Directive 2002/95/EC on the restriction of the use of certain hazardous substances in electrical and electronic equipment (RoHS), as amended by Commission Decisions 2005/618/EC, 2005/717/ EC, 2005/747/EC (RoHS Directive), and WEEE.Ethernet - Industry standard EMI-suppressed RJ-45 connector.

Axia® Quasar™

The New Star in Axia IP Consoles

OVERVIEW

With cosmic precision, otherworldly sound, and real star power, Axia's new Quasar sixth-generation mixing console draws upon the Telos Alliance's rich history as the inventor of AoIP for broadcast with more than 9,700 AoIP consoles on-air worldwide. Axia has channeled all that experience into this new flagship console, consolidating its native AoIP architecture and refining it for the ultimate user experience with limitless production possibilities for radio and specialized TV applications.

Sleek, Ergonomic Design

Quasar features a sleek new look and extremely high-quality components, rugged enough for a lifetime of uninterrupted use. Designed based on extensive global customer feedback and ergonomic studies, Quasar has an easy-to-operate touchscreen user interface (no external display required) that operators can also access remotely via any HTML5 device. The absence of an overbridge makes for easy desk installation, and the console is fanless and modular, with redundant load-sharing power supply units.

Customizable and Easy to Use

Quasar makes the operator's job dramatically easier, including new Source profiles (for source-associated logic automation), automatic mix-minus, and automixing an all channels. Extensive metering is built into the surface right where it needs to be—on every channel display and next to each fader, as well as on the monitor module. Users can customize their Quasar surface thanks to user-assignable buttons in the Master touchscreen module and in every channel strip.

For TV, production, or high-end radio applications, the powerful new Quasar Engine delivers 64 stereo input channels—all with robust DSP processing—and loudness metering on all outputs. Four programmable Layers allow the user to control all channels, including DSP, even on smaller surfaces.

Mature, Reliable AoIP Technology

Quasar gives operators confidence with world-renowned Axia audio quality and reliability. The Engine's native AoIP processing, based on a server-class hardware platform, ensures high-performance audio. The console's sixth-generation technology is mature and sophisticated, offering extreme reliability, with system modularity minimizing single points of failure.

IN DEPTH

Sixth-Generation AoIP Console from Axia

Quasar is Axia's new, state-of-the-art broadcast mixing system. And it is built like a tank.

Beautiful and rugged, with durable and scratch-resistant work surfaces, high-resolution color TFT displays and RGB pushbuttons throughout, plus an industrial-grade 12.1-inch TFT IPS touchscreen. With its easy-to-use User Interface, advanced ergonomics, top-quality components, and massive feature set, Axia raises the bar with Quasar, once again setting new quality standards for native AoIP broadcast consoles.

Configurations

Available in sizes from 4 to 28 faders per frame, with support for up to 64 faders in multiple linked frames, the console offers a reduced footprint and directly connects to an Axia network with a single or dual redundant Ethernet cable. Frames are available in both tabletop or flush-mount versions and can be converted from one type to the other with a special kit.

Modules

Fader modules offer touch-sensitive controls and motorized faders as standard features, along with high-resolution bargraph metering for each fader, source-drive color-coding, and customizable hardware buttons on the entire surface. A variety of source profile types provide control of mic/line inputs, telephones, codecs and other devices. Enhanced, integrated features for phones and codecs include auto-assigned mix-minus on each channel, easy talkback for remote talent cueing, one-button off-air phone record mode, and integrated, touchscreen-based Telco line switching that can interface with Telos phone hybrids and VoIP/SIP talk-show systems.

Touchscreen UI

Quasar sports a Master Touchscreen module that presents a simple and intuitive GUI (Graphical User Interface) that is so familiar, you'll be acquainted with the operation of the console within minutes. Thanks to its touchscreen, Quasar does not require an external display to function, although an external monitor could be connected via the rear HDMI port to show a duplicate of the touchscreen interface.

Web UI

New Expert Source Profile controls allow power users a granular definition of custom logic associated with each source. The end user can program GPIO control, mix-minus routing, talkback, and other functions based upon console channel state. Flexible Record Mode gives complete control of monitors, meters, headphone feeds, program bus assignments and more. New Show Profiles allow up to 4,000 console "snapshots" with different settings, layouts, and defaults loaded instantly, customizing the board to each show requirement or talent preference, if desired. An HTML5 remote GUI is built into the Web UI, to allow remote control and operation of the console from any browser-enabled device.

Cooling & Power Supply

The Quasar control surface is completely fanless, with built-in industrial-grade Power Supply Units available in either single or redundant configuration. Up to four PSUs can be included in a single console frame, depending on frame size and configuration desired.

Control Surface Dimensions

Width: from 428mm / 16.85" (4 faders+Master) to 1360mm / 53.54" (28 faders+Master)
Depth: 580mm / 22.83"
Height: 110 95mm / 4.33 3.74" (table-top frame with rubber feet)

Axia Quasar Engine

Quasar is powered by the Quasar Engine, a native AoIP powerhouse with 64 Stereo channels, 4-band fully parametric EQ, powerful dynamics processing and automixer on every channel, four program buses and eight auxiliary buses. Four Surface Layers and a Virtual Mixer (VMix) with 16 independent 5-channel V-Mixers extend the mixing capacity of your Quasar console far beyond physical fader count. Support for AES67 is included, as well as Talent headphone processing and many other advanced features that make operation simpler and more intuitive. Redundant power is standard with this mix-engine platform. The Quasar Engine has forced fan cooling, while the Quasar console is completely fanless.

Build Quality

The Quasar system is designed and built to last a lifetime. All its components were carefully selected with very strict lifetime requirements. All parts subject to wear are industrial, automotive, or even avionics grade.

Leveraging the Power of Axia's AoIP Infrastructure

The Quasar connects to Axia's Livewire+ AES67 AoIP network, and takes advantage of its powerful distributed I/O architecture. The Livewire+ network allows detection, sharing and control of audio resources across multiple studios connected to the network, and its technology complies with the latest AES67 standards.

Surface Features

- Super-reliable 6th-generation surface from Axia
- Compact and sleek design, based on extensive ergonomic studies
- Designed for any size radio studios and specialized TV installations
- Built-in, modular fanless PSUs with redundant option
- Reduced fader pitch for higher fader density
- No overbridge for easier installation on work surfaces. No OLEDs
- Single or split-frame configurations available, table-top or flush-mount
- Table-top frames are convertible to flush-mount
- Color and finishes fit with lighting used in modern TV-camera-equipped studios
- Modules can be installed as standalone outside of frame for modern, nontraditional creative work spaces
- Easy-to-operate touchscreen-based UI. No external display required
- New, intuitive Web UI with integrated HTML5 remote control
- Ethernet-connected and self-contained surface modules
- All channel strip pushbuttons are user-assignable
- Up to 4 user-programmable Surface Layers
- 8 user-programmable Master buttons with capacitive touch-sense
- All encoders and faders are touch-sensitive
- Extensive metering built right into the surface, including fader bargraphs
- New Quasar Engine with 64 stereo input channels
- Fully redesigned DSP processing, available on all channels
- Automixer available on all channels
- Additional V-Mixer with 80 stereo inputs (independent from console)
- Highly flexible and automated audio workflows including auto mixing and auto mix minus
- Source and show profiles for ultimate customization without user intervention
- Customizable color strip on modules for easy identification of source groups
- Motorized faders

Axia Fusion®

Where Design and Technology Intersect.

AES67 Livewire+™

OVERVIEW

Since 2003, Axia® has become the name broadcasters think of first when they think of networked broadcast facilities. Thousands of radio and audio professionals have made Axia their first choice for powerful, flexible, easy-to-use mixing consoles.

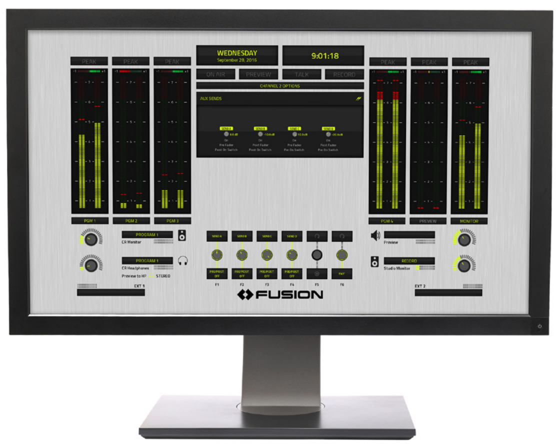
Fusion is the new Axia modular console packed with features and capabilities refined from over a decade's worth of IP-Audio experience. It's available in frame sizes to support consoles of 8 to 40 faders in single or multiple linked frames. Fusion may be powered by the Axia PowerStation® or StudioEngine DSP mixing engines, and connects to the Axia network with a single CAT-6 Ethernet cable, allowing the sharing of local audio devices (and their associated GPIO control) among multiple studios to maximize efficiency and reduce cost.

Fusion has four stereo Program buses, four Send buses, and two Return buses. A variety of module types are available, from fader-only modules to Call Controller modules with integrated multi-line controls for Telos® multi-line phone systems. Fusion also features unique Axia VMix (Virtual Mixer) channels, which allow combining up to 5 audio sources for presentation on a single console fader — further extending the flexibility and usefulness of the console.

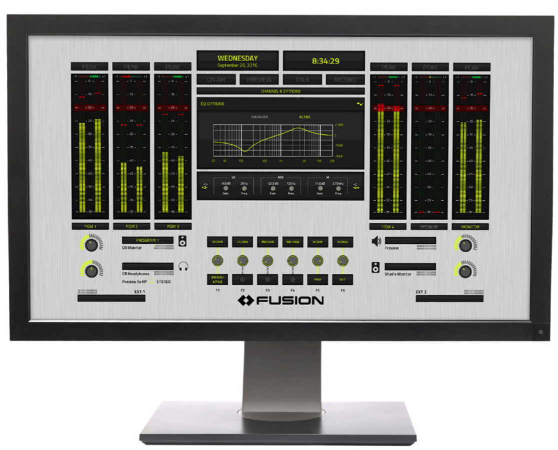
Other features include Auto-assigned, auto-generated mix-minus on each channel, easy individual or group talkback for remote talent cueing, one-button off-air phone record mode, and up to 99 Show Profiles console "snapshots" for set, save and recall of console layouts customized to the working style of individual shows or operators. Built-in digital EQ may be applied individually to all audio sources, as well as dynamic microphone processing from Omnia® for all mic sources.

Fusion Screen Shots:

The main view

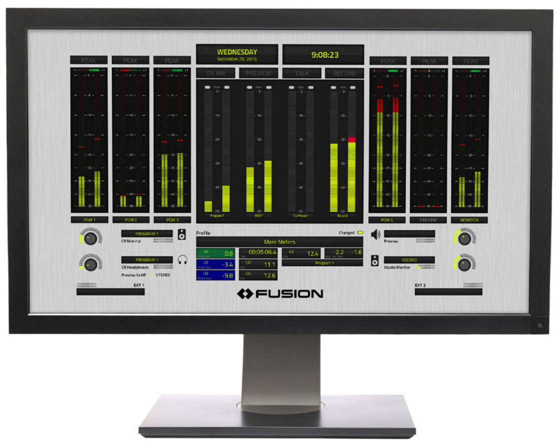


AUX SEND view with BBC PPM ballistics shown on meters



Channel EQ view with VU meter ballistics shown

Channel Mic processing view with VU meter ballistics shown



Loudness metering, EBU Digital ballistics shown on main meters, and 4 assignable meters

Show profile selection view with Nordic ballistics shown

Channel source profile selection view with EBU Digital ballistics shown

FEATURES

- From 8 to 40 fader channels, each with instant, unlimited access to any source. Assign any type of source to any channel.
- Rugged construction of extruded aluminum ensures rigidity and EM-tightness. Anodized aluminum work surface with laser-etched markings that can't be rubbed off ensures durability and good looks for life.
- Four main stereo outputs (Program-1 through Program- 4), plus four stereo Aux sends and two Aux returns.
- High-resolution OLED displays above each fader strip display selected source, full-time source and backfeed confidence metering, talkback status, pan/fade information and more.
- Integrated intercom capability includes built-in IFB for two-way communication to individual talent positions via headphone feeds and mics, plus a variety of optional drop-in intercom modules that connect to Axia IP Intercom whole-plant intercom systems.
- Flexible, intuitive Talkback system lets board ops talk to hosts, studio guests, external feeds — any source with an associated backfeed.
- Every channel has a stereo Preview ("cue") function, with a unique latching interlock system for fast, intuitive operation. Multiple channels may be assigned to Preview simultaneously.
- Reconfigurable monitor section with reassignable controls let operators instantly change monitored sources "on-the-fly."
- Software control of options such as EQ, mic dynamics, aux sends and returns, pan and balance and other features delivers maximum flexibility without panel clutter or intimidating controls.
- Built-in Omnia dynamics processing lets operators combine compression, de-essing and expansion with EQ to "sweeten" microphone sources.
- A unique, flexible Record Mode enables one-button setup of record mixes for phone bits or off-air interviews.

- Consolidated user display conveys meter, clock, timer and monitor source information at a single glance. Meter up to 6 buses at once by default, using VU or PPM-style ballistics — add another 4 meters for a total of 10 if desired.
- Precision timer and clock functions, including an event timer that can be triggered by pre-defined sources, a countdown timer with last-minute alerting and a time-of-day clock that can be synchronized to network time using NTP.
- Show Profiles set-save-recall feature allows users to instantly recall a customized personal profile, or a profile tailored to specific show types. Up to 99 Show Profiles can be saved for interview shows, music-intensive programming, call-in talk shows, etc.
- Console functions can be accessed remotely for configuration, management and diagnostic purposes using any standard Web browser.
- Built-in 5.1 discrete mixing capabilities for production use.
- Optional Telos phone control module provides direct, on-the-console line switching control of any Telos multi-line broadcast phone system.
- Numeric keypad (with # and * keys) lets operators quickly place calls with phone systems or codecs attached to the Axia network.
- Completely automatic, hands-free mix-minus generation for every Phone caller or remote Codec source.
- Built-in User Keys for can be programmed with Axia PathfinderPC routing control software to control profanity delay units (such as the 25-Seven® Program Delay Manager), or can be used to trigger routing salvos, scene changes or send GPIO commands.
- No audio passes directly through the Fusion control surface, keeping your programming safe from studio accidents. All mixing and processing is performed by the StudioEngine or PowerStation mixing engines.
- Axia's trademark long-life conductive-plastic faders with side-loading actuation defy dirt, grit and dust.
- Aircraft-grade switches with LED lighting have been tested to withstand millions of operations.
- Can be directly remote-controlled using Axia SoftSurface software for Windows.
- Fusion 3.1 software update adds AES67 support.

IN DEPTH

A decade of IP-Audio experience, packed into a single console.

Ask broadcasters who's the leader in IP-Audio, and chances are good they'll tell you "it's Axia." That's because we're the *originators* of studio networking for broadcast facilities; we produced our first console in 2002 (back when everyone else was saying "Audio over Ethernet? That'll never work!"), and we've been listening, learning and inventing ever since. That's why thousands of happy broadcasters worldwide have put Axia consoles to work for them.

Now, we'd like to introduce you to Fusion, the newest modular IP-Audio broadcast console from Axia.

Why did we name it Fusion? According to Webster's, "fusion" means "joining two or more things together to form a single entity." And that's just what we've done. We've taken everything we've learned in the past decade – about talent experiences, on-air mechanics, in-studio workflow and more – and combined those thoughts, ideas and observations into the smoothest, most intuitive, most indestructible networked console yet.

Fusion was created from what you, our clients, have taught us about today's fast-paced broadcast environment; an environment that interfaces with listeners both on the dial and on the 'Net. And our talented team (made up of scientists, engineers and even former air talent) built a console that has what it takes to support everything from the mile-a-minute call-in talk shows, to tight music-driven formats, to multi-talent morning shows — or anything else your programming department can imagine.

So, what things did we learn that made their way into Fusion? That "Powerful" doesn't have to mean "complicated," for one. You taught us that a broadcast console can have lots of capabilities, and still be easy to use. So our console designers looked at the way broadcasters accomplish complicated things, and figured out ways to make them simpler.

Like mix-minus: Fusion creates mix-minus for every codec and phone caller you put on the air. Automatically. With no extra buttons to push, bus assignments to make, or settings to change. It *just happens*.

Or, take recording off-air phone bits for later playback. 'Til now, it meant taking talent and callers from Program buses manually, assigning them to utility buses, manually starting a recorder...and don't forget creating the mix-minus! Fusion's one-touch Record Mode does all of that for you, managing bus assignments, creating the mix-minus, recording the conversation, even changing the monitor feed — then putting it all back the way it was when you've finished.

Then there's Fusion's built-in talent Mic processing that combines with its Show Profile set-save-recall system. Got a jock whose voice needs a little extra sweetening? Build a custom voice-processing mode for them using Omnia compression, limiting and de-essing tools, then save those settings in a personalized profile they can recall anytime they want.

What about reconfiguring the console from producing a music show to handling a live in-studio band performance with multiple mics and DI inputs? In the old days, you'd be pushing dozens of buttons to make new input choices, bus assignments, monitor settings and EQ tweaks. With Fusion, you can do it all with two clicks, bringing up a stored Show Profile snapshot that suits the job at hand.

In fact, Fusion is a power-user's dream. A unique Expert Source Profile mode lets you build custom audio inputs with completely customized GPIO functions, IFB backfeed and mix-minus, and Monitor and Program Bus assignments — all based on channel On/Off/Preview state. These powerful tools give you complete control of the behavior of audio sources – *on a per-show basis* – as they enter and leave the console, allowing automation of complex operations and helping operators run easier, more error-free shows.

Mixing capacity? Fusion has 4 Program buses, plus 4 Aux sends and 2 Aux returns, along with 16 five-channel "Virtual Mixers" that let you mix multiple audio inputs using software-controlled virtual faders.

You also told us you value durability. Axia consoles are known for their toughness, and Fusion is no exception. All work surfaces are made of heavy-duty, anodized machined aluminum panels. This ensures that Fusion will shrug off mistreatment by even the toughest jocks — there are no plastic overlays to crack and peel, and no paint to wear off. Fusion's markings are laser-etched, so they stay legible forever; they literally can't rub off!

We also put razor-sharp, high-resolution OLED displays on every fader strip. These are readable from nearly every angle, which aids talent during fast-paced show production. These OLEDs also display full-time confidence meters for each assigned source, further ensuring smooth, error-free shows and helping prevent dead air. You'll find even more information in Fusion's on-screen metering display, too.

Making the most of modern wide-screen monitor design, Fusion displays six stereo meters by default, making it perfect for main control rooms with multiple active program outputs. Meters are switchable and can display gain in VU, DIN or BBC-style PPM, plus EBU Digital and Nordic scales.

Finally, our designers have made Fusion the easiest Axia console, ever, to set up. Like any other AoIP device, it connects directly to the network via Ethernet, allowing unparalleled flexibility when placing consoles and connecting to mixing engines.

Take a good look at the bottom of a fader strip. Notice there's a "Talkback" key there? Every mic source on your Axia network – news booth, talent position, producer's station – can have an individual headphone backfeed. Touching the Talkback key associate with any mic source allows your board op to talk privately to just that talent position (and they can talk back to the CR, too, using their mic). Fusion operators can use even use this unique Talkback feature to communicate with phone callers, remote talent or other studios using the console mic, and can "button mash" to communicate with entire groups of locations at once.

With Fusion, controls for all of your studio devices are right on the console, where they're most useful. For instance: phone hybrid modules with dedicated faders give instant control of Telos talkshow systems; talent can dial, answer, screen and drop calls without ever taking their eyes off the console, which means smoother, easier on-air phone segments. IP Intercom modules let talent communicate with other places in the broadcast plant using the CR mic, and even take broadcast-quality intercom audio directly to air when desired.

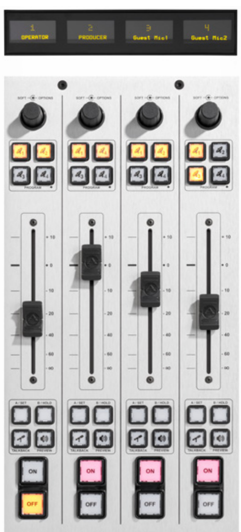
Like all Axia consoles, Fusion has a built-in, password-protected Web server for easy remote administration from your office, your boss' office, even your home office – anywhere there's a network connection. Fusion works with Axia SoftSurface virtual console software too, so talent can take direct remote control of the console.

Naturally, our designers put their legendary attention to detail to work on Fusion. Redundant power supply options with fanless convection cooling, hot-swappable modules, silky-smooth conductive-plastic faders with a side-loading design that foils dirt and other contaminants, razor-sharp OLED options displays, optical rotary encoders and, of course, avionics-grade switches with LED lighting (tested to withstand more than five million operations) are all part of Fusion's premium design. You can also add redundant power without additional IO with Axia Console Power Supply, which offers a single-cable connection to PowerStation Main, providing backup power with automatic switching. (Auto-sensing power supply, 90VAC to 240VAC, 50 Hz to 60 Hz. 250 Watts, 2RU.)

The only thing not premium about Fusion is...its price. For all of this power, flexibility and ease of use, Axia clients have told us they'd expect to pay much more. Luckily, you don't have to.

Fusion is fully customizable, of course, with a full options list of module types designed to suit your station's unique way of making great radio. There are integrated controls for phones, codecs and studio talkback, SmartSwitch modules with context-sensitive displays that enable one-touch router salvos, even motorized faders for remote control or integration with your delivery system.

4-Fader Module



The 4-Fader module is where you start to build your Fusion. Use it for any source: line, mic, hybrid, phone or codec source. Comes in standard and motor-ized-fader versions for use with automation systems or other moving-fader applications.

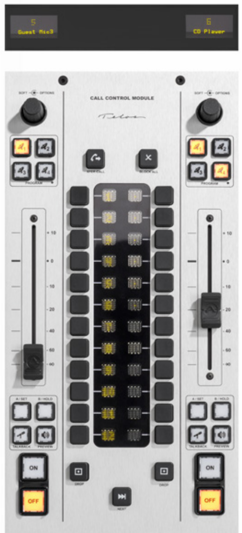
Monitor Modules

The other basic module you'll need is a Monitor module. You can choose between two types:

The Expert Monitor/Navigation module shown here has extended monitor, headphone and preview controls, a numeric entry/dialpad that can be used with Fusion phone modules, plus four programmable User Keys that can trigger GPIO commands like profanity delay controls or recording devices, or be used with Axia PathfinderPC software to issue routing salvos, initiate scene changes, etc.

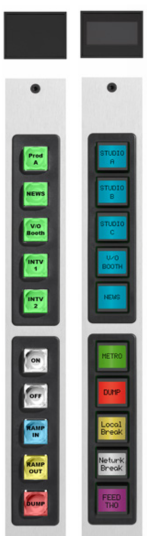
For studios where expert monitor controls are not needed, the Standard Monitor/Navigation module is a space-saving design that incorporates two faders in addition to the numeric entry/dial pad and basic Monitor/Headphone controls.

Call Controller Module



The Call Controller module has two faders plus integrated line switching controls with Status Symbols, for on-console control of advanced Telos broadcast phone systems. Available in standard and motorized-fader versions.

Switch Modules



Two available styles of programmable switch modules work with Axia PathfinderPC routing control tools. They make it easy to put custom routing salvos or simple machine logic right at talent's fingertips.

Economical 10-Button Film-Cap switch modules are perfect for giving talent access to often-used machine-control or GPIO-triggered routing commands. LED button backlights can be individually changed to any of 8 colors.

Need more complex control of routing functions? 10-button SmartSwitch modules feature dynamic, backlit LCD displays. Button functions, colors and even text can be programmed to change in response to user input using Axia PathfinderPC software. Construct custom routing salvos, cascading machine-logic command sets, or other complex routing operations.

IP Intercom Modules

Fusion consoles come equipped with a sophisticated Talkback system that allows board ops to communicate directly with remote talent via individual Talkback channels. But when larger facilities require even more powerful communication capabilities, these 10 and 20-station intercom modules, part of the Axia IP Intercom system, put broadcast intercom controls right in the console. Station presets and GPIO functions for both types of modules are programmed using any standard Web browser. Using these Intercom modules, Fusion operators can instantly talk to any other studio, control room, operations center – even PCs equipped with Axia SoftCom intercom software. And the audio is broadcast-quality, so putting an Intercom source on-air is easy and sounds great.

10 and 20-station OLED Intercom modules feature high-resolution programmable OLED displays that indicate assigned stations. The 10-Station Filmcap intercom module has 10 LED-lit film-cap buttons for economical on-console IP Intercom integration.

Fusion User Panels

Fusion Fader Panel

In-studio accessory panel mounts in tabletop or turret. Provides remote control of on/off functions; dedicated Mute and Talkback buttons give talent full control of their position. Volume/selection knob allows users to select their headphone monitor source; display readout confirms their choice. Motorized fader which is linked to a channel on the console. Connects to Fusion Interface Board CANbus using CAT-5 cable. 6"x 2", requires 3" mounting depth.

Fusion Mic Control / Headphone Selector Panel

In-studio accessory panel mounts in tabletop or turret. Provides remote control of mic on/off functions; dedicated Mute and Talkback buttons give talent full control of their position. Volume/selection knob allows users to select their headphone monitor source; display readout confirms their choice. Connects to Fusion Interface Board CANbus using CAT-5 cable. 6"x 2", requires 2" mounting depth.

Fusion Headphone Selector Panel

In-studio accessory panel with volume/selection knob mounts in tabletop or turret to allow users to select their headphone monitor source; display readout confirms their choice. Two preset buttons let talent quickly recall frequently-listened-to sources. Connects to Fusion Interface Board CANbus using CAT-5 cable. 6"x 2", requires 2" mounting depth.

Fusion Mic Control Panel

In-studio accessory panel for remote control of mic on/off functions. Mounts in tabletop or turret; includes dedicated Mute and Talkback buttons. Requires one free Axia GPIO port per panel. 6"x 2", requires 2" mounting depth.

Fusion Producer's Mic Control Panel

Special accessory panel for producers source profiles, which mounts in tabletop or turret to provide remote control of mic on/off functions. Includes dedicated Mute key. Two Talkback keys allow producers to talk to control room board op, studio guests, or codecs. Requires one free Axia GPIO port per panel. 6"x 2", requires 2" mounting depth.

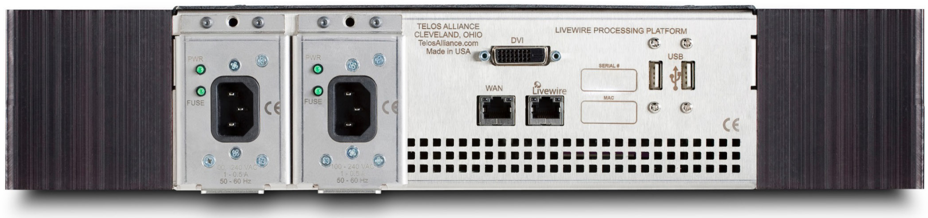
Fusion 4-Button LCD SmartSwitch Panel

Program these backlit LCD button panels using Axia PathfinderPC software to provide producers or talent with remote access to often-used machine control or software functions. In-button LCD display shows function readout. Dynamic programming allows specific assigned functions to change for each Show Profile or in response to user activation. Requires PathfinderPC (P/N 3001-0015). Easily connects to Fusion Interface board CANbus using CAT-5 cable. Mounts in tabletop or turret. 6"x 2", requires 2" mounting depth.

Mixing Engines

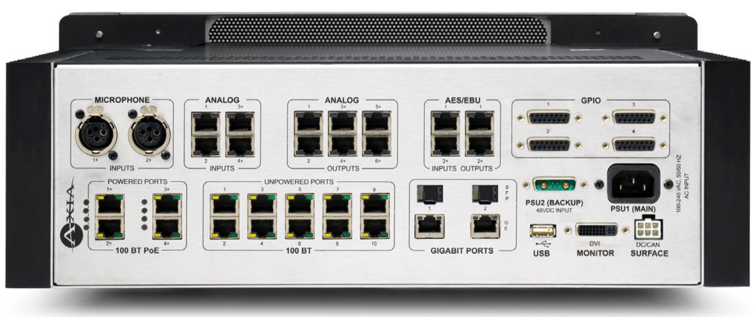
Fusion consoles were designed to give you maximum flexibility and configuration options. So instead of just one mixing engine, you've got your choice of two! Pair your Fusion control surface with a powerful Linux-based StudioEngine and separate xNode audio interfaces – or choose the PowerStation integrated console engine, an all-in-one powerhouse with audio I/O, DSP mixing engine and integrated zero-configuration network switch.

StudioEngine



Pair your Fusion with Axia StudioEngine, an extremely powerful mixing and processing device based on a blazingly-fast Intel processor. Each StudioEngine is fanless, has dual-redundant field-replaceable modular power supplies, and has so much CPU power it can outperform the very largest digital or router-based consoles. StudioEngine has multiple simultaneous inputs, outputs, mix-minus feeds, monitor signals, EQ and voice processing; it's the power behind state-of-the-art broadcast studios from New York to Tokyo.

PowerStation



PowerStation is an "integrated console engine", an all-in-one devices that makes it very easy to install Axia studios and Fusion consoles. Inside that ruggedly handsome case you'll find a super-powered DSP mixing engine, husky power supply sourced from telecom gear designed for harsh environments, plenty of built-in digital, analog and mic I/O, plus EQ, voice processing — and even a custom, built-for-broadcast Ethernet switch with Gigabit connectivity.

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
- Output Level: +4 dBu, nominal

Analog Line Inputs

- Input Impedance: >40 k Ohms, balanced
- Nominal Level 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 Ohms, balanced (XLR)
- Signal Format: AES-3 (AES/EBU)
- AES-3 Input Compliance: 24-bit with selectable sample rate conversion, 32 kHz to 96kHz input sample rate capable.
- AES-3 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
- Latency <3 ms, mic in to monitor out, including network and processor loop

Frequency Response

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

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, Stereo Separation and CMRR

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

Audio Processing

Equalizer

- Frequency Bands: 20Hz to 320Hz, 125Hz to 2KHz, 1.25KHz to 20KHz.
- Cut/Boost range on each band: -25dB to +15dB.
- Q-factor: Automatic - bandwidth varies based on amount of cut or boost.

Compressor

- Threshold: -30dB to 0dB Ratio: 1:1 to 16:1
- Post-processor Trim Level: Adjustable from -20dB to +20dB

Expander/Noise Gate

- Threshold: -50dB to 0dB Ratio: -30dB to 0dB

De-esser

- Threshold: -20dB to 0dB Ratio: 1:1 to 8:1

Axia Console Power Supply

- Add redundant power to PowerStation main without additional IO.
- Single-cable connection to PowerStation main provides backup power with automatic switching.
- Auto-sensing power supply, 90VAC to 240VAC, 50 Hz to 60 Hz.
- Power consumption: 250 Watts.

Power Supply AC Input, PowerStation Aux & Main

- Auto-sensing supply, 90VAC to 240VAC, 50 Hz to 60 Hz, IEC receptacle, internal fuse
- Power consumption: 500 Watts

Operating Temperatures

- -10 degrees C to +40 degrees C, <90% humidity, no condensation

Regulatory

North America: FCC and CE tested and compliant, power supply is UL approved.

Europe: Complies with the European Union Directive 2002/95/EC on the restriction of the use of certain hazardous substances in electrical and electronic equipment (RoHS), as amended by Commission Decisions 2005/618/EC, 2005/717/ EC, 2005/747/EC (RoHS Directive), and WEEE.

Axia® IP-Tablet Virtual Radio Software

This Is Virtual Radio

OVERVIEW

When you use Axia gear, you are part of a growing, innovation-driven ecosystem—not locked into one console company's vision of the future. In fact, part of Axia's innovation is thanks to the creativity of its more than 115 Livewire partners and its vast ecosystem. Witness the award-winning Axia IP-Tablet Virtual Radio software, designed by Livewire partner IP-Studio.

This one-of-a-kind software suite is a bold step toward the virtual radio studio of the future. It virtualizes the monitoring and control of your gear on a Windows PC or tablet (anything from a low-cost Asus model to a high-end Microsoft Surface) running any of the IP-Tablet Virtual Radio software modules. Sold individually, you can buy those software modules that are most relevant and beneficial to your studio setup, such as the Telos Systems VX or VX Prime phone systems, the Telos Z/IP ONE IP-audio codec, the Omnia.9 audio processor and Omnia VOCO 8 mic processor, Axia Fusion Console via modules that control the Axia PowerStation and StudioEngine, Axia DESQ, RAQ, Radius, and iQ consoles via modules that control the Axia QOR.32/16 engines, xNode control, and Axia Pathfinder buttons. Axia IP-Tablet also controls any HTML5-enabled gear! A Metadata Tools app allows you to pull metadata information from your automation system to display on the tablet screen.

Aside from virtualizing control, the IP-Tablet also allows you to manage user rights for device access, linking a user's profile to his or her needs. You can mount the Axia IP-Tablet right into your Fusion console with a beautifully machined, completely flush IP-Tablet Mount or use it on a freestanding tablet. Adding great value to your Axia console purchase, this the IP-Tablet Virtual Radio software puts your most-used console functions right at your fingertips, whether on a PC or a tablet.

IN DEPTH

The Axia IP-Tablet Virtual Radio software goes beyond virtualizing control of just your Axia consoles. It allows you to control a variety of Telos Alliance equipment, including your Omnia.9 audio processor, Omnia VOCO 8 mic processor, Axia xNodes, Axia Pathfinder, Telos Z/IP One IP-audio codec, Telos VX and VX Prime VoIP phone systems, and more. Read on for the full details on all the software modules available to help you make virtual radio in your facility a reality.

IP-Tablet Core Virtual Radio App

IP-Tablet Virtual Radio software gives you the ability to customize and consolidate control of Axia Fusion consoles, xNodes, Telos phone systems / codecs, and Omnia processors all on a single standard Windows touchscreen tablet. Purchase a IP-Tablet Virtual Radio Core app for each tablet you want to use in your studio environment.

IP-Tablet PowerStation Virtual Radio App

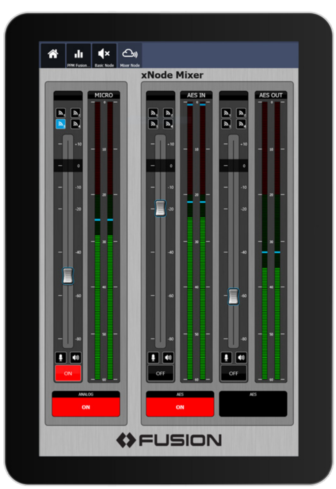
The IP-Tablet PowerStation app unlocks full control of an Axia Fusion console connected to the PowerStation. You can customize the layout and controls to fit the exact needs of each user. Choosing from one of the predesigned templates or the simple drag-and-drop design interface, you can draft a layout that is specific to your needs. You can design a screen containing VU Meters, Timers, Time and Date display, status of On-Air, Preview and Talkback, the ability to control faders and change fader sources, choose console shows, and have full Vmix control with faders. One PowerStation license will allow any number of tablets to connect to the PowerStation for control, remembering that each tablet still requires its own Core license.

IP-Tablet StudioEngine Virtual Radio App

Like the IP-Tablet PowerStation, the IP-Tablet StudioEngine app unlocks full control of an Axia Fusion console connected to the StudioEngine. You can customize the layout and controls to fit your exact needs. Choose from one of the predesigned templates or use the simple drag-and-drop design interface to draft a layout. You can design a screen containing VU Meters, Timers, Time and Date display, status of On-Air, Preview Talkback, the ability to control faders and change fader sources, choose console shows, and have full Vmix control with faders. One StudioEngine license will allow any number of tablets to connect to the StudioEngine for control, remembering that each tablet still requires its own Core license.

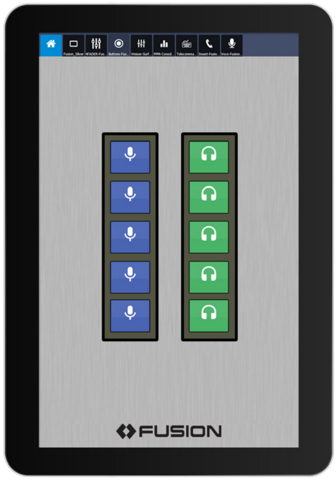
IP-Tablet QOR.32/16 Virtual Radio App

The IP-Tablet QOR software allows broadcasters to virtually control Axia iQ, DESQ, Radius, and RAQ consoles connected to a QOR.32 or QOR.16 engine. Broadcasters can customize the layout and controls to fit the exact needs of each user, choosing from one of the predesigned templates or customizing the layout through the simple drag-and-drop design interface. Users can design a screen containing VU Meters, Timers, Time and Date display, the ability to control faders and change fader sources, and even choose console shows. One QOR License will allow any number of tablets and PCs to connect to the QOR for control, remembering that each still requires its own Core license.



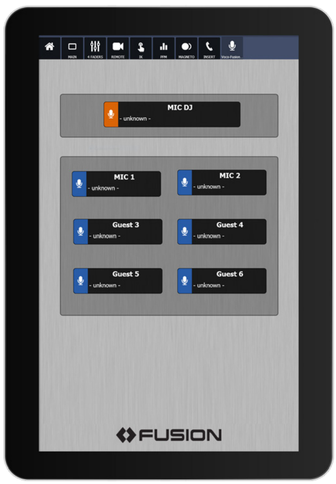
IP-Tablet xNodes Virtual Radio App

The IP-Tablet xNode Mixer app lets you access the internal mixer inside an xNode; change sources and routes; and control the levels of each with virtual faders. When you combine this software with an xNode, you get a fully customizable mini mixer.



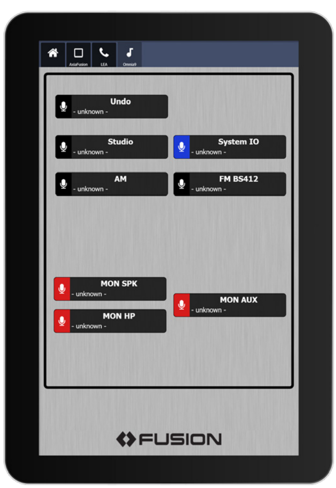
IP-Tablet Pathfinder Button Virtual Radio App

Design layouts of Pathfinder Button panels with this app. You can customize the size, shape, and color of the buttons and label them with custom text blocks, or choose from dozens of icons to serve as a legend. Requires Pathfinder PC, Pathfinder Pro or Pathfinder Core PRO.



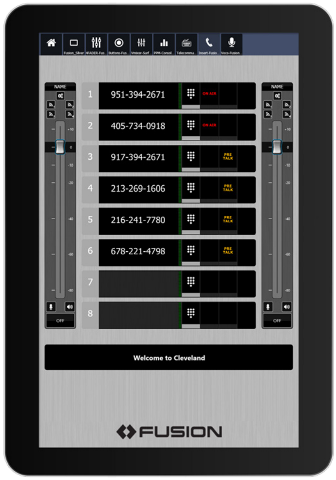
IP-Tablet VOCO 8 Virtual Radio App

Control the presets of an Omnia VOCO 8 mic processor with this app. You can design a screen mirroring the layout of your furniture to allow operators to easily change the Omnia VOCO 8 presets.



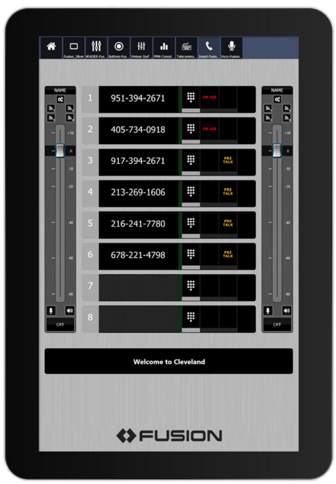
IP-Tablet Omnia.9 Virtual Radio App

Control the presets of an Omnia.9 processor with this software module. Easily switch processing presets right from your IP-Tablet without having to access the front panel of the Omnia.9 unit.



IP-Tablet VX Enterprise Virtual Radio App

Use the IP-Tablet to design a custom call management and control screen for your VX Enterprise phone system. You have control over answering a call, previewing a call, and switching a call on-air as well as caller ID information display.



IP-Tablet VX Prime+ Virtual Radio App

Use the IP-Tablet to design a custom call management and control screen for your VX Prime+ phone system. You have control over answering a call, previewing a call, and switching a call on-air as well as caller ID information display.

IP-Tablet Z/IP ONE Virtual Radio App

Use the IP-Tablet to control your Z/IP One IP-audio codec. You have the ability to establish a call and adjust the codec bit rates directly from the IP-Tablet interface screen.

IP-Tablet Metadata Tools Virtual Radio App

Use the IP-Tablet to pull metadata information from your automation system to display on the tablet screen. Great for consolidating critical information into a central location for your board operators and on-air talent. You can also display web URLs and scrolling messages on the tablet screen.

IP-Tablet Fusion Mount

Finally, you can mount your 10-inch Windows tablet inside your Fusion console using this beautifully machined aluminum panel. Occupies four spaces in a Fusion frame.

Axia® SoftSurface

Software for Axia® Fusion™ Consoles

OVERVIEW

SoftSurface Virtual Console software for Windows gives you powerful real-time control of your Axia Fusion mixing console from home, office, or anywhere an Internet connection is available. Take direct remote control of your console, or, match SoftSurface directly to an Axia mixing engine to create a "virtual console" without a physical mixing surface. SoftSurface makes an ideal companion for existing consoles and it's also the perfect audio mixing solution for limited-space locations.

FEATURES

- Gives full remote control of Fusion consoles paired with Axia StudioEngine mixing engines.
- Pair directly with an Axia StudioEngine or PowerStation® to create a standalone “soft” console without a physical mixing surface. Supports from 4 to 48 faders in this mode.
- NTP-capable on-screen time of day clock/calendar.
- On-screen count-up event timer.
- Supports up to four Show Profiles console “snapshots” for instant recall of frequently-used configurations.
- Control all four program buses and all auxiliary mix buses.
- Remote control of mic compression, de-essing and expansion capabilities.
- On-screen control of per-source three-band parametric EQ.
- Excellent IFB Talkback capabilities let operators talk to other studios, external remote feeds, phone callers or any other source with its own backfeed.
- Full control of GPIO functions.
- When paired with Axia consoles equipped with motorized faders, physical fader position automatically mirrors that of the “virtual” SoftSurface fader.

IN DEPTH

Control at the Click of a Mouse.

You asked for a way to remotely control your console. Axia heard you! Meet SoftSurface, the audio mixing application for Windows.

You can use SoftSurface two ways. As a remote control, it gives powerful real-time control of premium Axia mixing consoles, utilizing an Axia StudioEngine or PowerStation that's connected to a Livewire® network. It's perfect for remote diagnostics or off-site operation of a mixing console from remotes, transmitter sites — even from home, via an Internet gateway.

As a virtual console, SoftSurface combines with an Axia StudioEngine or PowerStation to create a "soft" mixing surface. It's perfect for those limited-space situations where there's no room for a real console. With SoftSurface, if you've got a Windows laptop and an IP connection, you're good to go.

Now, when we say SoftSurface lets you control a console, we don't mean a wimpy console. You get all the functionality and features of Axia's extremely popular Fusion modular mixing surface, the board at the heart of thousands of superb broadcast facilities around the world.

SoftSurface opens up new dimensions of creative applications for broadcasters. Remote broadcasts get easier: your talent can take a tablet with SoftSurface, a USB mic and a Telos® Z/IP ONE IP Codec into the field, link up with the Fusion console at your studio, and have its entire suite of capabilities at their fingertips — leave those CDs and MP3 players at home.

Or, pair SoftSurface with a StudioEngine or PowerStation mixing engine for a "virtual console" installation in personal studios, or areas where space restrictions don't permit a physical control surface.

The SoftSurface display is divided into a virtual mixing surface and a control section. The mixing section's onscreen width varies based on the number of channels you wish to display, while the context-sensitive control section is fixed in size, and navigation is via a series of intuitive tabs.

The Main Monitor tab provides control of the monitor speakers, operator headphones, external monitor speakers, and preview volume.

The Show Profiles tab reveals the profiles that have been configured and allows the user to select a show and load it. As few as one or as many as 99 Show Profiles are supported.

Monitor Options provides control over dimming values and control of the channel feeding the monitors and headphone.

The Meter Options tab provides control over the presentation of the meters, including choice of metering ballistics styles: VU, BBC and DIN-style PPM, EBU Digital and Nordic.

Auxiliary Send and Return provides control of the final AUX send mix, as well as two auxiliary returns, as defined in the Show Profile. The returns can be assigned to a program bus. These options are defined in a Show Profile and controlled as needed by the user. Individual sources can be assigned to feed one or all AUX Send buses from the Channel Options Aux Send screen.

EQ curves are adjustable for each audio source. EQ can be adjusted on the fly, or saved as part of a source's Source Profile and automatically recalled whenever that source is loaded to a fader.

Mic, Codec and Phone sources can be sweetened with Voice Dynamics from Omnia®. Expansion, Compression and De-Essing are part of the toolkit; Like EQ, dynamics settings can be adjusted at will or preset and saved with sources for automatic recall.

SoftSurface's signal processing toolkit is completed with comprehensive Pan, Summing and Phase control, available on a per-source basis.

SPECIFICATIONS

Hardware Requirements

- A PC with Microsoft™ Windows XP or higher operating system installed.
- A display screen with a minimum resolution of 1280x1024 pixels.
- A mouse, touch-screen, or other suitable pointing device.
- A 100BASE-T LAN connection with a static IP address.
- 20Mb free disk space.

Audio Processing

Equalizer

- Frequency Bands: 20Hz to 320Hz, 125Hz to 2KHz, 1.25KHz to 20KHz.
- Cut/Boost range on each band: -25dB to +15dB.
- Q-factor: Automatic - bandwidth varies based on amount of cut or boost.

Compressor

- Threshold: -30dB to 0dB Ratio: 1:1 to 16:1
- Post-processor Trim Level: Adjustable from -20dB to +20dB

Expander/Noise Gate

- Threshold: -50dB to 0dB Ratio: -30dB to 0dB

De-esser

- Threshold: -20dB to 0dB Ratio: 1:1 to 8:1

Axia® iQx AoIP Console

Look Below the Surface

AES67 Livewire+™

OVERVIEW

Appearances can be deceiving. Axia® iQx looks like a traditional console, but it's actually a surface and mix engine rolled into one —at a surprisingly affordable price. We dove deep with AES67, giving you access to every source, anywhere on the network. With nearly limitless connections, we can't even fathom the possibilities.

FEATURES

AoIP Features

- Standards-based console supports AES67 and SMPTE 2110-30.
- Configurable from 8 to 24 faders, each with instant access to any AoIP source.
- Works with any AES67, Livewire+™ AES67, or Livewire source.
- Assign any AoIP source to any channel, like having over 16 million patch points!

Console Features

- 24 built-in stereo three-band EQs.
- Channel-input confidence meters assure operator of audio presence before taking sources to air.
- Fader's context-sensitive Soft key can be used to activate talkback, start delivery system events, or perform other special functions.
- Stereo Preview ("cue") function for every fader with unique interlock system for fast cueing of multiple sources.
- Reconfigurable CR monitor section with direct selection of 4 program busses and re-assignable buttons that allow instant monitoring of external sources.
- Additional monitor section provides separate monitor volume, source selection, and talkback controls for an associated studio.
- Flexible built-in mix-minus and talkback system lets operator talk to guests and phone/codec sources, each with an associated, automatically created backfeed mix-minus.
- Precision event timer can be operated manually or triggered by starting preselected sources.
- Time-of-day clock synchronized to network PTP time or NTP, with time zone and daylight savings settings.
- Unlimited source profiles with four quick-recall snapshots (show profiles).
- Remote access for configuration, management, and diagnostics using standard Web browser. No proprietary software required.
- Auxiliary "V-Mix" remotely controllable 5-input virtual mixer, for whatever extra audio need you may have.

Telco Features

- Unique Record Mode enables one-button setup of record mixes for phone bits or off-air interviews.
- Optional Telco Expansion Frame provides direct, on-the-console control of Telos multiline phone systems. High-resolution displays for instant call status information. Dump key for profanity delay.

Design Features

- Smooth, long-life 100mm conductive-plastic faders resist dirt and contamination.
- High-resolution displays provide responsive, readable VU or PPM metering styles. Displays can be switched to display 2, 3, or 4 meters at once.
- Alphanumeric displays below each fader show current audio source and other features without panel clutter or intimidating controls.
- Up to 2 additional iQ side frames can be joined to produce a single large control surface or operated separately to suit studio design.
- Tabletop mounting.
- Fan-free, convection-cooled operation, quiet enough for on-air mics.

IN DEPTH

Surface Meet Engine...Engine, Surface

Traditionally, surfaces need to be connected to a separate mix engine, the so-called "brains" of the system. Axia iQx combines the mix engine and surface into one unit. That's one less component and connection you need to worry about when building your studio, allowing you to get up and running quickly and easily.

Standards-Based for Easy Connectivity

Built from the ground up as an AoIP console, iQx is AES67-compliant and stands fully capable of supporting emerging standards including SMPTE 2110-30. There's no limit to the number of sources and connections you can access on the network. Well, OK, 16 million...but who's counting?

Don't Pay for I/O You Don't Need!

If you already have an existing AoIP network, you may not even need to add additional I/O. You can connect your iQx and access any audio source, anywhere on the network, from anywhere in the world. Have iQ consoles or QOR engines on your network? Drop in an iQx and share resources without having to add additional equipment. By piggybacking off existing resources like an Axia QOR engine, a station on a budget can suddenly build an additional studio for far less. This is a first and changes the game.

Affordable Yet Impeccable

Not only does iQx allow you to utilize existing network resources without breaking the budget in ancillary I/O, it is very affordable, bridging the gap between a lower-end $2K console and a higher-end, $8K model, while retaining all the craftsmanship, build quality, and innovation that Axia is known for.

Instant Studio

Create an instant studio by adding a Telos Alliance xNode to support local microphones, headphones, and any other sources you might want, and you're ready to rock in a flash. Add additional xNodes or an xSwitch to expand I/O and network switching capabilities as needed. While iQx is sophisticated inside, it's familiar on the outside so there's no steep learning curve.

Plug-and-Play Installation

Configuration is a no-brainer thanks to a built-in web GUI—no proprietary application required. And while some locations where consoles are installed can be tight on rack space, iQx doesn't require any thanks to the integrated engine. The console can be placed tabletop, without the need to modify furniture. All while offering network and power redundancy expected by world-class broadcast facilities.

Ready for the Future, Even If You Aren't

iQx is great for nontraditional studio setups too. Need a temporary studio for a special event or to add or move a studio quickly? Mixing sports, remote talent, or audio that isn't located in your studio? iQx is your console, letting you bring in sources from all over the world. Control is not just a local thing either; an iQx can be controlled from the remote site, so no in-studio operator is required. With iQx, you have access to everything, everywhere. While we can't predict the future, we know change is in store, and iQx is flexible enough to support you today and tomorrow.

Get the Benefits of an AoIP Network

New to AoIP? In addition to its own unique advantages, iQx is an AoIP console at its heart, giving you all the benefits that come with an AoIP network. More flexibility, easier and faster installs, cost efficiency, a decentralized system without a single point of failure, and the ability to upgrade from analog one studio at a time are all upsides of AoIP.

Once a source is on an AoIP audio network, it can be made available to any device on that same network. In the analog days, this could only be accomplished with costly routing switchers or stacks of distribution amps. Now, any station with an AoIP network has this capability built in.

Finally, with no analog audio physically inside an AoIP console, you reduce or even eliminate interference that can compromise sound. With AoIP, stations enjoy much cleaner audio throughout the studio facility.

SPECIFICATIONS

Inputs/Outputs

- Four main stereo program bus outputs, plus bus outputs for Record, Phone, CR Monitor, CR Monitor Direct, CR Headphones, Preview, Talkback to CR, Talkback to External, Studio Guest HP, Studio Monitor, Studio Talent HP.
- Automatic mix-minus provided for any source input.
- 5 stereo input to 1 stereo output, auxiliary "V-Mix" mixer.
- 2 network ports, 1 for link to the AoIP network and 1 for a local I/O.
- Fanless power supply standard; second redundant power supply optional.

Dimensions

- Unit weight: 12 lbs / 5.5 kg.
- Shipping weight: 25 lbs / 11.4 kg.
- W: 20.4"(518.5mm), L: 18.2"(463.3mm), H: 4.4"(111.4mm)

Axia® iQ
The Smarter IP Console

OVERVIEW

The Axia® iQ console system can be used to build custom consoles of sizes from 8 to 24 faders. A basic system consists of one iQ 8-Fader Main Frame and one QOR.32 integrated console engine, a DSP-based mixing engine which also incorporates analog and digital audio I/O, GPIO and a custom, zero-configuration Ethernet switch. Faders and control capabilities can be expanded by adding one or more iQ Expansion Frames (up to a maximum of 3 frames per console installation). iQ console frames may be placed on top of desk surface, or mounted drop-in style. Multiple frames may be physically joined if desired.

iQ operates as a standalone console, but can also connect to Axia networks. The iQ mixing surface plugs into the QOR.32 engine using a single cable. Setup couldn't be simpler: connect the iQ control surface to the QOR.32, add audio inputs using CAT-5, perform some fast Web-based configuration, and your iQ system is ready to broadcast. It really is that simple!

iQ features 3 dedicated stereo Program buses, plus a stereo Utility bus that can be used for phone calls, off-air recording, or as a fourth Program bus. Automatic mix-minus is provided on each fader, plus talkback functions, one-button off-air Record Mode, Show Profile functions for instant recall of up to 4 pre-defined console "snapshots", high-resolution OLED program meters switchable between VU and PPM metering styles, OLED option and source name displays on each fader strip, Studio and Control Room monitor controls. Add redundant power to your iQ system with the Axia Console Power Supply. This single-cable connection to the QOR.32 console engine provides backup power with automatic switching.

FEATURES

- Configurable from 8 to 24 faders, each with instant access to any source.
- Proven surface-and-core architecture separates control from mixing processes. No audio passes directly through iQ; all mixing and processing is performed in the QOR.32 Integrated Console Engine – so studio "accidents" don't turn into off-air events.
- Software upgrade for QOR.32 integrated console makes iQ AoIP console AES67-compliant.
- Assign any type of source to any channel with a twist of the Options knob.
- Four main stereo outputs (Program-1 through Program-4).
- Built-in three-band per-source EQ.
- Alpha-numeric OLED displays below each fader always show the current audio source, and, when the Options knob is pressed, offer fast adjustment of fader gain trim, voice EQ, pan and balance, phase correction and other features without panel clutter or intimidating controls.
- Channel-input confidence meters assure talent of audio presence before taking sources to air.
- Each fader's context-sensitive Soft key can be used to activate talkback, start delivery system events, or perform other special functions.
- Every fader has a stereo Preview ("cue") function, with a unique interlock system for fast cuing of multiple sources.
- Smooth, long-life 100mm. conductive-plastic faders resist dirt and contamination.
- Reconfigurable CR monitor section with direct-selection of Program buses and reassignable buttons that allow instant monitoring of external sources.
- An additional monitor section provides monitor volume, source selection and talkback controls for an associated air studio.
- Flexible talkback system lets board op talk to studio guests or any Phone or Codec source with an associated backfeed.
- Up to 8 automatic mix-minuses may be used simultaneously for phones, remote talent, etc.
- Unique Record Mode enables one-button setup of record mixes for phone bits or off-air interviews.
- High-resolution OLED displays provide responsive, readable VU or PPM metering styles. Displays can be switched to display 2, 3 or 4 meters at once.
- Precision event timer that can be operated manually or triggered by starting pre-selected sources.
- Time-of-day clock can be synchronized to network time using NTP.
- Four custom Show Profile "snapshots" can be saved to instantly recall frequently-used console setups – useful to quickly prepare for interview segments, music-intensive programming, call-in talk shows, etc.
- All functions can be accessed remotely for configuration, management and diagnostic purposes using any standard Web browser.
- Multiple iQ frames can be joined to produce a single, large control surface, or operated separately if desired to suit studio design.

- Optional Telco Expansion frame provides direct, on-the-console control of matching Telos® iQ6 six-line telephone system, or other Telos talkshow systems. High-resolution OLED displays use exclusive Telos Status Symbols for instant call status information. Includes a Dump key to trigger user-supplied profanity delay unit using GPIO closures.
- Easy-to-deploy QOR.32 integrated console engine includes console CPU and power supply, DSP mixing engine, custom Ethernet switch with 6 Livewire® ports and 2 Gigabit ports for studio networking, 16 analog inputs and 8 analog outputs, 2 AES inputs and 2 AES outputs, 4 Mic inputs with switchable Phantom power, and 8 GPIO ports for machine control. I/O can be expanded using Telos Alliance® xNodes.
- Integrated zero-configuration network switch is custom-designed for broadcasting — no switch setup required.
- QOR.32's built-in Ethernet switch supports Simple Networking, allowing up to 4 iQ consoles to be daisy-chained without the need for a separate core switch.
- Fan-free, convection-cooled power supply for noiseless in-studio operation.
- Optional backup Axia Console Power Supply with automatic failover for complete peace of mind.
- Configurable network gateway allows loading of networked as well as local audio sources while simultaneously exporting audio streams for network use elsewhere. Gateway can be configured for 12-in, 4-out or 8-in, 8-out modes.

IN DEPTH

Control at the Click of a Mouse. Easy Installation. Fast Configuration. Intuitive Operation.

For today's broadcast engineer, there aren't enough hours in the day. You're looking for a console that makes the most of your resources. One that installs quickly, with a minimum of fuss. One that works smart, with features that help talent to do smoother, more error-free shows. One that's perfectly happy in a standalone studio — but that also connects quickly and easily to a larger studio network.

iQ is the console you're looking for. More than just a pretty face, iQ is a broadcast console with mixing engine, analog and AES audio I/O, Livewire audio connections, machine-control logic and a zero-configuration built-for-broadcast Ethernet switch, all rolled into one easy-to-deploy package. Connect the iQ control surface to the QOR.32 integrated console engine with just one cable. Then add audio inputs using CAT-5, perform some fast Web-based configuration and, presto! your new iQ console is ready to broadcast. Optional Axia Console Backup Power Supply adds redundant power to your iQ system for complete peace of mind.

Thanks to all those built-in goodies, iQ is the perfect self-contained, standalone console for an individual studio. But should you wish to expand and network with other studios, iQ can grow with you. Simple Networking lets you daisy-chain up to four QOR.32 engines without the need for an external Ethernet switch. You can add iQ expansion frames to create consoles as large as 24 faders. Other optional frames add control for Telos telephone systems and GPIO routing functions to the console.

More smart stuff: iQ remembers. Four Show Profile memory positions let you set, save and recall snapshots of console settings for later use. High-resolution Organic LED meters (bright, high-resolution displays that are bright and legible, even under direct lighting) offer switchable VU or PPM metering styles, and the ability to meter two, three, or all four buses at once.

There are also OLED displays on every fader that provide source assignments, pan & balance settings, fader options and more — which means no additional computer monitors or mice to clutter up your studio. The display can also work with the Soft Keys just below to trigger GPIO events, step automation events, and adjust source input options.

iQ saves your studio furniture, too. Its desktop design lets you place it atop any solid surface — no templates to decipher or countertops to cut (unless you really want to). Since iQ only requires a single cable to connect control surface to mixing engine, even cable access holes can be small and unobtrusive. And iQ lets you choose between freestanding or contiguous console designs: you can easily join iQ expansion frames into one unit, or leave them separate to deploy a split-console design.

Like all Axia consoles, iQ is over-engineered for long life. It's built with sturdy, premium materials, to withstand even the beatings a weekend overnight jock can give. It's got sturdy, machined aluminum frame construction, LED button lighting, long-life conductive-plastic faders, and anodized – not painted! – surfaces with laser-etched markings that can't ever rub off. But the most clever thing about iQ might just be its price. A 16-fader iQ costs about half what you'd expect to pay for a console with all these features. Now that's pretty smart, don't you think?

iQ System Components

Like all Axia systems, iQ is customizable and scalable. The QOR.32 integrated console engine contains the console's mix engine, CPU, power supply and 32 audio I/O connections, and supports console sizes from 8 to 24 faders. Start with an eight-fader iQ Main Frame, then add expansion frames with more faders and capabilities to tailor iQ to your studio's needs. Gigabit Ethernet lets you connect to a larger Axia network; Simple Networking lets you daisy-chain up to four QOR.32 without the need for an external Ethernet switch.

iQ Main Frame

The heart of your iQ console; can be installed as a standalone console or connected to an Axia studio network. Has three dedicated stereo Program buses, plus a stereo utility bus that can be used for phone calls, off-air recording, or as a fourth Program bus, eight faders, automatic per-fader mix-minus, high-rez OLED program meters and channel displays, Studio and Control Room monitor controls and an integrated Talkback system. For bigger consoles, add one or two iQ expansion frames to build boards of up to 24 faders. Flexible mounting system allows desktop, drop-in and even rack-mounted operation.

8-Fader Expansion Frame

The iQ 8-Fader frame doubles the size of your iQ instantly. It's simple to expand the capacity of iQ consoles, even after they've been in service, so you can easily grow your iQ system; expansion frames plug right into the QOR.32 integrated console engine. Like all iQ frames, the 8-Fader expansion comes equipped with Axia's rugged, anodized machined-aluminum surface, conductive-plastic faders, aircraft-quality switches and LED button lighting. Can be physically joined to Main Frame or left separate.

6-Fader Expansion Frame with User Keys

Put machine control and GPIO-triggered routing commands at your operators' fingertips with this iQ expansion frame. In addition to the six additional faders, 10 User Keys can be software-mapped to control audio delivery systems, send contact closures or route GPIO commands to studio devices.

6-Fader Telco Expansion Frame

Puts integrated phone system control right where it belongs: on the console, to help eliminate distractions and errors. Along with six silky-smooth conductive-plastic faders, this frame includes on-the-board hybrid controls for the matching Telos iQ6 six-line telephone hybrid (it works with other Telos phone systems, too). The learning curve is low: exclusive Telos Status Symbols readouts on sharp-as-a-tack OLED displays, along with familiar twin hybrid controls, make easy work of busy call-in segments.

iQ6 6-Line Telco Gateway

The iQ6 broadcast phone system was custom-designed by the phone experts at Telos specifically for iQ consoles. It works with the hybrid controls built into your iQ's Telco expansion frame (and with Telos VSet6 phone controllers). Connect it to the QOR.32 console engine with a CAT-6 cable, plug in your phone lines, and start taking calls.

QOR.32 Integrated Console Engine

The QOR.32 integrated console engine is a DSP-based mixing engine with onboard I/O, GPIO, console power supply and custom-built, configuration-free Ethernet switch. You'll find plenty of I/O, including 4 mic inputs with selectable Phantom power, 16 analog inputs, 2 AES/EBU inputs, 8 Analog outputs, 2 AES/EBU outputs, 8 GPIO machine-control logic ports (each with 5 opto-isolated inputs and 5 outputs), and that powerful integrated Ethernet switch with 6 Livewire 100BASE-T ports (4 with PoE), 2 Gigabit ports (RJ-45 & SFP), and 4 CANBus ports for console expansion (maximum cable length 40 feet). Sure, that's plenty of I/O, but if you need more you can instantly add it just by plugging in Telos Alliance xNodes. QOR.32 is convection-cooled for utterly silent, fan-free operation.

Axia Console Backup Power Supply

Add redundant power to your iQ system! Single-cable connection to QOR.32 console engine provides backup power with automatic switching. Auto-sensing power supply, 90VAC to 240VAC, 50 Hz to 60 Hz. 250 Watts. Rackmount, 2RU.

SPECIFICATIONS

QOR.32 Connections

- Microphone Inputs: 4x balanced XLR-F, with selectable Phantom power
- Analog Inputs: 16x RJ-45, StudioHub+ standard.
- Analog Outputs: 8x RJ-45, StudioHub+ standard.
- AES/EBU Inputs: 2x RJ-45, StudioHub+ standard.
- AES/EBU Outputs: 2x RJ-45, StudioHub+ standard.
- GPIO: 8x DB-15
- Livewire:
    - 4x 100BASE-T with PoE, RJ-45
    - 2x 100BASE-T, RJ-45
    - 2x 1000BASE-T, RJ-45
    - 2x Gigabit, SFP (Small Form Pluggable)
- Console Frame Connections: 3x, 6-pin “latch and lock” style
- Accessory Connections: 1x, 6-pin “latch and lock” style

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
- Output Level: +4 dBu, nominal

Analog Line Inputs

- Input Impedance: 20 k Ohms
- Nominal Level 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: AES-3 (AES/EBU)
- AES-3 Input Compliance: 24-bit with selectable sample rate conversion, 20 kHz to 216kHz input sample rate capable.
- AES-3 Output Compliance: 24-bit
- Digital Reference: Internal (network timebase) or external reference 48 kHz, +/- 2 ppm
- Internal Sampling Rate: 48 kHz
- Output Sample Rate: 48 kHz
- A/D Conversions: 24-bit, Delta-Sigma, 256x oversampling
- D/A Conversions: 24-bit, Delta-Sigma, 256x oversampling
- Latency <3 ms, mic in to monitor out, including network and processor loop

Frequency Response

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

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: 125 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, Stereo Separation And CMRR

- Analog Line channel to channel isolation: 90 dB isolation minimum, 20 Hz to 20 kH
- Microphone channel to channel isolation: 80 dB isolation minimum, 20 Hz to 20 kHz
- Analog Line Stereo separation: 85 dB isolation minimum, 20Hz to 20 kHz
- Analog Line Input CMRR: >50 dB, 20 Hz to 20 kHz Microphone Input CMRR: >50 dB, 20 Hz to 20 kHz

Audio Processing

- Mic Equalizer (applicable to up to 6 faders)
- Frequency Bands: 20Hz to 320Hz, 125Hz to 2KHz, 1.25KHz to 20KHz.
- Cut/Boost range on each band: -25dB to +15dB.
- Q-factor: Automatic - bandwidth varies based on amount of cut or boost.

Power Supply AC Input, QOR.32 with iQ Console

- Auto-sensing supply, 90VAC to 240VAC, 50 Hz to 60 Hz, IEC receptacle, internal fuse
- Power consumption: 100 Watts

Operating Temperatures

- -10 degrees C to +40 degrees C, <90% humidity, no condensation

Dimensions

- iQ Main Frame 20.5” x 19” x 4.5” (desktop to meter bridge)
- iQ Expansion Frames 17.5” x 18.25” x 3” (desktop to tallest control)

Regulatory

North America: FCC and CE tested and compliant, power supply is UL approved.

Europe: Complies with the European Union Directive 2002/95/EC on the restriction of the use of certain hazardous substances in electrical and electronic equipment (RoHS), as amended by Commission Decisions 2005/618/EC, 2005/717/ EC, 2005/747/EC (RoHS Directive), and WEEE.

Axia® Radius

Throw Your Budget A Curve

OVERVIEW

Radius is an all-in-one console system designed for small standalone or networked studios, where no more than eight faders are needed. Like all Axia® consoles, it's easy to deploy: each Radius control surface is powered by a burly QOR.16 integrated console engine with DSP-powered mixing engine, analog and digital audio I/O, custom Ethernet switch and GPIO ports. The Radius surface connects to the QOR.16 engine with a single CANBus cable.

Radius includes 4 stereo Program buses — 3 dedicated Program, Audition and Utility mixing outputs; the fourth a stereo Utility bus for recording phone callers or other off-air bits. The fourth bus may also be used as an additional Program bus. Automatic mix-minus is provided on each fader, plus talkback functions, one-button off-air Record Mode, Show Profile instant recall of up to 4 pre-defined console "snapshots", LED bar-graph program meters switchable between VU and PPM meter styles, high-resolution OLED option displays on each fader, and Studio and Control Room monitor controls. Radius can be placed on top of desk surfaces, mounted drop-in style, or rack-mounted using included hardware.

FEATURES

- 8 faders, each with instant access to any source. Assign any type of source to any channel with a simple twist of the Options knob.
- Proven surface-and-core architecture separates control from mixing processes. No audio passes directly through Radius; all mixing and processing is performed in the QOR.16 Integrated Console Engine – so studio "accidents" don't turn into off-air events.
- Four main stereo outputs (Program-1 through Program-4).
- Software upgrade for QOR.16 integrated console makes Radius AoIP console AES67-compliant.
- Built-in three-band per-source EQ.
- Alpha-numeric OLED displays below each fader always show the current audio source, and, when the Options knob is pressed, offer fast adjustment of fader gain trim, voice EQ, pan and balance, phase correction and other features without panel clutter or intimidating controls.
- Channel-input confidence meters assure talent of audio presence before taking sources to air.
- Each fader's context-sensitive Soft key can be used to activate talkback, start delivery system events, or perform other special functions.
- Every fader has a stereo Preview ("cue") function, with a unique interlock system for fast cuing of multiple sources.
- Smooth, long-life 100mm. conductive-plastic faders resist dirt and contamination.
- Reconfigurable CR monitor section with direct-selection of Program buses and reassignable buttons that allow instant monitoring of external sources.
- An additional monitor section provides monitor volume, source selection and talkback controls for an associated air studio.
- Flexible talkback system lets board op talk to studio guests or any Phone or Codec source with an associated backfeed.
- Up to 8 automatic mix-minuses may be used simultaneously for phones, remote talent, etc.
- Unique Record Mode enables one-button setup of record mixes for phone bits or off-air interviews.
- Bright, readable bar-graph displays provide responsive, readable VU or PPM metering styles. Switchable displays allow metering any Program bus or monitor selection.
- Meter-bridge display includes a precision event timer that may be operated manually or triggered by starting preselected sources, and a time-of-day clock that can be synchronized to network time using NTP.
- Four custom Show Profile "snapshots" can be saved to instantly recall frequently-used console setups – useful to quickly prepare for interview segments, music-intensive programming, call-in talk shows, etc.

- All functions can be accessed remotely for configuration, management and diagnostic purposes using any standard Web browser.
- Radius surface is field-convertible to rack-mounted operation. Special faders provide smooth operation, yet hold their positions in vertical orientation.
- Easy-to-deploy QOR.16 integrated console engine includes console CPU and power supply, DSP mixing engine, custom Ethernet switch with 6 Livewire® ports and 2 Gigabit ports for studio networking, 8 analog inputs and 4 analog outputs, 1 AES input and 1 AES output, 2 Mic inputs with switchable Phantom power, and 4 GPIO ports for machine control. I/O can be expanded using Telos Alliance® xNodes.
- Integrated zero-configuration network switch is custom-designed for broadcasting — no switch setup required.
- QOR.16's built-in Ethernet switch supports Simple Networking, allowing up to 4 iQ consoles to be daisy-chained without the need for a separate core switch.
- Fan-free, convection-cooled power supply for noiseless in-studio operation.
- Configurable network gateway allows loading of networked as well as local audio sources while simultaneously exporting audio streams for network use elsewhere. Gateway can be configured for 12-in, 4-out or 8-in, 8-out modes.

IN DEPTH

Spend Less. Get More.

"You get what you pay for," as the saying goes. But sometimes, you actually get less. For example, you've probably noticed how "affordable" radio consoles are usually missing important features and capabilities. Trying to do a radio show with a board like that is like trying to open a can with a spoon: you might succeed eventually, but you sure won't enjoy it.

At Axia, we're broadcasters too, through and through. And we believe that having a reasonable equipment budget shouldn't mean being forced to settle for something less than you deserve. We've decided you should get more than you pay for — much more. Which is why we designed Radius, the IP console that proves you can have your cake and eat it, too. While some console companies try to see how much they can take out of a console to meet a price point, Radius was designed in exactly the opposite way: we challenged ourselves to see just how many features and capabilities we could pack in, while still meeting your budget requirements.

Radius is the easiest AoIP console ever. Just connect the 8-fader mixing surface to the QOR.16 integrated console engine, plug in your sources and power, and you're ready to make great radio. Because it's so compact, Radius is the perfect standalone console, but Gigabit ports on its QOR.16 integrated console engine let you connect it to other studios too. Radius' network gateway lets you load up to 12 audio sources from anywhere in your Livewire network, while simultaneously sending your locally produced streams back out to the net.

Radius is loaded with features you'd expect to pay much more for. You'll find three stereo Program buses, and a stereo utility bus that can be used for recording phone calls and off-air bits (or as a fourth Program bus). Automatic mix-minus for every phone caller and remote talent means never having to fiddle with making a manual backfeed. Bright multi-segment LED meters are switchable between VU and PPM styles. High-resolution OLED displays for each fader show source assignments, audio options and more. And Show Profiles that you can program to instantly load talent's most frequently-used console configurations.

Like all Axia consoles, Radius is built for long-lasting reliability, ready to stand up to anything your operators throw at it, with an EM-tight steel frame, anodized machined aluminum work surface with etched markings that can never rub off, silky-smooth conductive-plastic faders, aircraft-quality switches and rotary controls, and integrated clock/event timer. There are even monitor source and volume controls for an associated studio — something you'd expect to find only in bigger consoles costing much more.

There are also OLED displays on every fader that provide source assignments, pan & balance settings, fader options and more — which means no additional computer monitors or mice to clutter up your studio. The display can also work with the Soft Keys just below to trigger GPIO events, step automation events, and adjust source input options.

Like its big brother, iQ, Radius is designed to sit atop any solid surface — no templates to decipher or countertops to cut (unless you really want to). A single cable connects it to the QOR.16 mixing engine. Also like iQ, Radius is built with premium materials like a machined aluminum frame construction, LED button lighting, long-life conductive-plastic faders, and anodized – not painted! – surfaces with laser-etched markings that can't ever rub off.

Audio I/O, GPIO, console CPU, super-duty power supply, and even a network switch are all built into the QOR.16. Just plug in your mics, CD players, codecs, profanity delays, whatever. There are 16 audio I/O ports: two Mic inputs with switchable Phantom power, eight analog inputs and four analog outputs, and one AES/EBU input and output. QOR.16 also has four GPIO logic ports for machine control of studio peripherals, six 100BASE-T ports for Livewire devices, and two Gigabit ports with SFP for connection to the outside world. For more I/O, just add Telos Alliance xNode interfaces. And you can daisy chain as many as four QOR engines without the need for an external Ethernet switch, making installation even more economical.

SPECIFICATIONS

QOR.16 Connections

- Microphone Inputs: 2x balanced XLR-F, with selectable Phantom power
- Analog Inputs: 8x RJ-45, StudioHub+ standard.
- Analog Outputs: 4x RJ-45, StudioHub+ standard.
- AES/EBU Inputs: 1x RJ-45, StudioHub+ standard.
- AES/EBU Outputs: 1x RJ-45, StudioHub+ standard.
- GPIO: 4x DB-15
- Livewire:
    - 4x 100BASE-T with PoE, RJ-45
    - 2x 100BASE-T, RJ-45
    - 2x 1000BASE-T, RJ-45
    - 2x Gigabit, SFP (Small Form Pluggable)
- Console Frame Connections: 1x, 6-pin “latch and lock” style
- Accessory Connections: 1x, 6-pin “latch and lock” style

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
- Output Level: +4 dBu, nominal

Analog Line Inputs

- Input Impedance: 20 k Ohms
- Nominal Level 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: AES-3 (AES/EBU)
- AES-3 Input Compliance: 24-bit with selectable sample rate conversion, 20 kHz to 216kHz input sample rate capable.
- AES-3 Output Compliance: 24-bit
- Digital Reference: Internal (network timebase) or external reference 48 kHz, +/- 2 ppm
- Internal Sampling Rate: 48 kHz
- Output Sample Rate: 48 kHz
- A/D Conversions: 24-bit, Delta-Sigma, 256x oversampling
- D/A Conversions: 24-bit, Delta-Sigma, 256x oversampling
- Latency <3 ms, mic in to monitor out, including network and processor loop

Frequency Response

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

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: 125 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, Stereo Separation And CMRR

- Analog Line channel to channel isolation: 90 dB isolation minimum, 20 Hz to 20 kH
- Microphone channel to channel isolation: 80 dB isolation minimum, 20 Hz to 20 kHz
- Analog Line Stereo separation: 85 dB isolation minimum, 20Hz to 20 kHz
- Analog Line Input CMRR: >50 dB, 20 Hz to 20 kHz
- Microphone Input CMRR: >50 dB, 20 Hz to 20 kHz

Audio Processing

- Mic Equalizer (applicable to up to 6 faders)
- Frequency Bands: 20Hz to 320Hz, 125Hz to 2KHz, 1.25KHz to 20KHz.
- Cut/Boost range on each band: -25dB to +15dB.
- Q-factor: Automatic - bandwidth varies based on amount of cut or boost.

Power Supply AC Input, QOR.16 with Radius Console

- Auto-sensing supply, 90VAC to 240VAC, 50 Hz to 60 Hz, IEC receptacle, internal fuse
- Power consumption: 100 Watts

Operating Temperatures

- -10 degrees C to +40 degrees C, <90% humidity, no condensation

Dimensions

- 20.5" x 19" x 4.5" (desktop to meter bridge)

Regulatory

North America: FCC and CE tested and compliant, power supply is UL approved.

Europe: Complies with the European Union Directive 2002/95/EC on the restriction of the use of certain hazardous substances in electrical and electronic equipment (RoHS), as amended by Commission Decisions 2005/618/EC, 2005/717/ EC, 2005/747/EC (RoHS Directive), and WEEE.

Axia® RAQ

Rack-Mount IP Console

OVERVIEW

Six-fader Axia® RAQ console provides a convenient way to add a physical mixing surface nearly anywhere, no matter how space-limited. RAQ has six rotary faders with OLED channel options displays, two stereo mixing buses and Preview (cue) bus, a high-resolution OLED meter display with switchable VU / PPM ballistics, and monitor / headphone controls for auditioning of Program buses or two assignable External monitor source selections.

RAQ is built for heavy duty work. Aircraft-quality switches feature all-LED lighting. The anodized metal work surface features rub-proof, etched markings that can't rub off. Smooth, accurate rotary faders with push-on/push-off channel switches make fast work of audio control. And RAQ features Axia's famous fully-automatic mix-minus for phone callers and codec sources, too. Show Profiles give instant recall of up to 4 pre-defined console "snapshots".

RAQ is ideal for standalone installation, but networks with larger Axia networks too. A RAQ control surface and a QOR.16 integrated console engine constitute a complete RAQ system, but two RAQ consoles, or one RAQ and one DESQ console, may be paired with a single QOR.16 for cost-effective multi-console deployment.

FEATURES

- 6 faders, each with instant access to any source. Assign any type of source to any channel with a simple twist of the Options knob.
- Proven surface-and-core architecture separates control from mixing processes. No audio passes directly through RAQ; all mixing and processing is performed in the QOR.16 Integrated Console Engine – so studio “accidents” don’t turn into off-air events.
- Software upgrade for QOR.16 integrated console makes the RAQ AoIP console AES67-compliant.
- Two stereo mix buses and a Preview (cue) bus.
- Alpha-numeric OLED displays below each fader always show the current audio source with audio confidence meter, and, when the Options knob is pressed, offer fast adjustment of fader gain trim, EQ, pan and balance and other features without panel clutter or intimidating controls.
- Channel-input confidence meters assure talent of audio presence before taking sources to air.
- Built-in three-band per-source EQ.
- Each fader’s context-sensitive Soft key can be used to activate talkback, start delivery system events, or perform other special functions.
- Every channel strip has a stereo Preview (“cue”) function, with a unique interlock system for fast cuing of multiple sources.
- Reconfigurable CR monitor section with direct-selection of Program buses and reassignable buttons that allow instant monitoring of external sources.
- Four custom Show Profile “snapshots” can be saved to instantly recall frequently-used console setups – useful to quickly prepare for interview segments, music-intensive programming, call-in talk shows, etc.
- Automatic mix-minuses for phones, remote talent, etc.
- Bright OLED meter display provides responsive, readable VU or PPM metering styles. Switchable display allows metering either Program bus.
- All functions can be accessed remotely for configuration, management and diagnostic purposes using any standard Web browser.
- Network gateway enables loading networked sources while simultaneously exporting outputs back to the network.
- Easy-to-deploy QOR.16 integrated console engine includes console CPU and power supply, DSP mixing engine, custom Ethernet switch with 6 Livewire® ports and 2 Gigabit ports for studio networking, 8 analog inputs and 4 analog outputs, 1 AES input and 1 AES output, 2 Mic inputs with switchable Phantom power, and 4 GPIO ports for machine control. I/O can be expanded using Telos Alliance® xNodes.
- QOR.16’s integrated zero-configuration network switch is custom-designed for broadcasting — no switch setup required. Supports Simple Networking, allowing up to 4 QOR engines to be daisy-chained without the need for a separate core switch.
- Each QOR.16 can support two connected RAQ or DESQ consoles, or one of each.
- Fan-free, convection-cooled power supply for noiseless in-studio operation.

IN DEPTH

A big console for small spaces.

Not every studio requires a full-size mixing console. Not every studio is full-size, itself! But you still want the advantages of IP-Audio networking: the ability to send program audio to other studios, the ability to consume audio from satellite downlinks, remote codecs and phone hybrids, or to trigger routing scene changes from a user-mapped control panel. And you don't want a toylike plastic pro-audio mixer — you want a real broadcast console that fits into a rack or turret, or on a small desktop space. A console with a small footprint, but big capabilities.

RAQ is a compact, special-purpose IP console from Axia. It may be compact in stature, but it's big on features and performance. RAQ has "big board" capabilities you won't find in other consoles of this size — automatic per-fader mix-minus, built-in EQ for voice and codec sources, and the ability to instantly load new local or networked sources to any fader with just the turn of a knob. Which means RAQ easily out-classes mixers with similar form factors — and even ones that take up much more room.

RAQ is a six-channel mixer over-engineered the Axia way, with super-duty rotary faders, aluminum front-panel, high-resolution OLED displays for channel assignment and metering, heavy-duty switches with LED lighting, and four Show Profile snapshot locations you can use to store and instantly recall favorite console configurations. One touch, and presto! Talent's favorite sources are loaded, monitor source configured, and bus assignments made.

RAQ has two stereo mixing buses, plus a Preview (cue) bus, which makes it the perfect rack-mount utility mixer, whether in the studio, in an OB van, or in a road case. It fits in just 4 RU of space, so you can place it anywhere you need a full-featured, rack-mounted mixer: News booths, editors' workstations, voice-over booths, dubbing stations, even small remote studios or club installations.

RAQ also features something else you won't find on other compact consoles: a full-featured Monitor section. Along with headphone and Preview volume controls, you'll also find a selector that lets you hear either Program 1, Program 2, or one of two External sources —helpful for monitoring off-air feeds, a processed headphone chain, or another studio. And you can finally say goodbye to Dymo labels and masking tape: each channel has an OLED display to show exactly what source is loaded.

Audio I/O, GPIO, console CPU, super-duty power supply, and even a network switch are all built into the QOR.16. Just plug in your mics, CD players, codecs, profanity delays, whatever. There are 16 audio I/O ports: two Mic inputs with switchable Phantom power, eight analog inputs and four analog outputs, and one AES/EBU input and output. QOR.16 also has four GPIO logic ports for machine control of studio peripherals, six 100BASE-T ports for Livewire devices, and two Gigabit ports with SFP for connection to the outside world. For more I/O, just add Telos Alliance xNode interfaces. And you can daisy chain as many as four QOR engines without the need for an external Ethernet switch, making installation even more economical.

And here's the kicker: one QOR.16 can power two RAQ mixers — or a RAQ and a DESQ (RAQ's six-fader, desktop-mount cousin)! Despite all these features, RAQ is so cost-effective, broadcasters are coming up with creative, new uses for them. We figured folks would use them for news booths, dubbing stations and guest performance mixers, but audio pros are also telling us they'd be ideal for broadcast remote kits, mobile trucks, for shipboard broadcasting, or as personal mixers. What else could you use them for? The possibilities are endless...

SPECIFICATIONS

QOR.16 Connections

- Microphone Inputs: 2x balanced XLR-F, with selectable Phantom power
- Analog Inputs: 8x RJ-45, StudioHub+ standard.
- Analog Outputs: 4x RJ-45, StudioHub+ standard.
- AES/EBU Inputs: 1x RJ-45, StudioHub+ standard.
- AES/EBU Outputs: 1x RJ-45, StudioHub+ standard.
- GPIO: 4x DB-15
- Livewire:
    - 4x 100BASE-T with PoE, RJ-45
    - 2x 100BASE-T, RJ-45
    - 2x 1000BASE-T, RJ-45
    - 2x Gigabit, SFP (Small Form Pluggable)
- Console Frame Connections: 1x, 6-pin "latch and lock" style
- Accessory Connections: 1x, 6-pin "latch and lock" style

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
- Output Level: +4 dBu, nominal

Analog Line Inputs

- Input Impedance: 20 k Ohms
- Nominal Level 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: AES-3 (AES/EBU)
- AES-3 Input Compliance: 24-bit with selectable sample rate conversion, 20 kHz to 216kHz input sample rate capable.
- AES-3 Output Compliance: 24-bit
- Digital Reference: Internal (network timebase) or external reference 48 kHz, +/- 2 ppm
- Internal Sampling Rate: 48 kHz
- Output Sample Rate: 48 kHz
- A/D Conversions: 24-bit, Delta-Sigma, 256x oversampling
- D/A Conversions: 24-bit, Delta-Sigma, 256x oversampling
- Latency <3 ms, mic in to monitor out, including network and processor loop

Frequency Response

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

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: 125 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, Stereo Separation And CMRR

- Analog Line channel to channel isolation: 90 dB isolation minimum, 20 Hz to 20 kH
- Microphone channel to channel isolation: 80 dB isolation minimum, 20 Hz to 20 kHz
- Analog Line Stereo separation: 85 dB isolation minimum, 20Hz to 20 kHz
- Analog Line Input CMRR: >50 dB, 20 Hz to 20 kHz
- Microphone Input CMRR: >50 dB, 20 Hz to 20 kHz

Audio Processing

- Mic Equalizer (applicable to up to 6 faders)
- Frequency Bands: 20Hz to 320Hz, 125Hz to 2KHz, 1.25KHz to 20KHz.
- Cut/Boost range on each band: -25dB to +15dB.
- Q-factor: Automatic - bandwidth varies based on amount of cut or boost.

Power Supply AC Input, QOR.16 with RAQ Console

- Auto-sensing supply, 90VAC to 240VAC, 50 Hz to 60 Hz, IEC receptacle, internal fuse
- Power consumption: 100 Watts

Operating Temperatures

- -10 degrees C to +40 degrees C, <90% humidity, no condensation

Dimensions

- W 19.0 in (48 cm), H 3RU, 6.97 in (177.0 cm), D 2.54 in (64.50 cm)

Regulatory

North America: FCC and CE tested and compliant, power supply is UL approved.

Europe: Complies with the European Union Directive 2002/95/EC on the restriction of the use of certain hazardous substances in electrical and electronic equipment (RoHS), as amended by Commission Decisions 2005/618/EC, 2005/717/ EC, 2005/747/EC (RoHS Directive), and WEEE.

Axia® DESQ

Compact Desktop IP Console

OVERVIEW

Six-fader, two-bus Axia DESQ console is a cost-effective, small-footprint console option perfect for small production studios, remote vehicles, content ingest stations, etc. DESQ has two stereo mixing buses and a Preview (cue) bus, six high-quality 100mm conductive-plastic faders for silky-smooth operation and long life, razor-sharp OLED channel options displays, an OLED meter display with switchable VU / PPM ballistics, and monitor / headphone controls for auditioning of Program buses or two assignable External monitor source selections.

There's also an OLED time-of-day clock + timer display, with auto / manual reset option. As with all Axia consoles, aircraft-quality switches feature all-LED lighting; the anodized work surface has rub-proof etched markings that can't rub off. Additional features include automatic mix-minus for phone callers and codec sources, EQ for voice sources, and Show Profile instant recall of up to four pre-defined console "snapshots".

DESQ requires no countertop cutout and takes only 16" square of desk space; it connects to the QOR.16 integrated console engine with a single power/control cable. DESQ is ideal for standalone installation, but works with larger Axia networks too. A DESQ control surface and a QOR.16 integrated console engine constitute a complete RAQ system, but two DESQ consoles, or one RAQ and one DESQ console, may be paired with a single QOR.16 for cost-effective multi-console deployment.

FEATURES

- 6 faders, each with instant access to any source. Assign any type of source to any channel with a simple twist of the Options knob.
- Proven surface-and-core architecture separates control from mixing processes. No audio passes directly through DESQ; all mixing and processing is performed in the QOR.16 Integrated Console Engine – so studio "accidents" don't turn into off-air events.
- Software upgrade for QOR.16 integrated console makes DESQ AoIP console AES67-compliant.
- Two stereo mix buses and a Preview (cue) bus.
- Alpha-numeric OLED displays below each fader always show the current audio source with audio confidence meter, and, when the Options knob is pressed, offer fast adjustment of fader gain trim, EQ, pan and balance and other features without panel clutter or intimidating controls.
- Channel-input confidence meters assure talent of audio presence before taking sources to air.
- Built-in three-band per-source EQ.
- Each fader's context-sensitive Soft key can be used to activate talkback, start delivery system events, or perform other special functions.
- Every channel strip has a stereo Preview ("cue") function, with a unique interlock system for fast cuing of multiple sources.
- Reconfigurable CR monitor section with direct-selection of Program buses and reassignable buttons that allow instant monitoring of external sources.
- Four custom Show Profile "snapshots" can be saved to instantly recall frequently-used console setups – useful to quickly prepare for interview segments, music-intensive programming, call-in talk shows, etc.
- Automatic mix-minuses for phones, remote talent, etc.
- Bright OLED meter display provides responsive, readable VU or PPM metering styles. Switchable display allows metering either Program bus.
- All functions can be accessed remotely for configuration, management and diagnostic purposes using any standard Web browser.
- Separate OLED clock/timer display features NTP-capable time-of-day clock and event timer that can be manually or automatically reset via source activation.
- Network gateway enables loading networked sources while simultaneously exporting outputs back to the network.
- Easy-to-deploy QOR.16 integrated console engine includes console CPU and power supply, DSP mixing engine, custom Ethernet switch with 6 Livewire ports and 2 Gigabit ports for studio networking, 8 analog inputs and 4 analog outputs, 1 AES input and 1 AES output, 2 Mic inputs with switchable Phantom power, and 4 GPIO ports for machine control. I/O can be expanded using Telos Alliance® xNodes.
- QOR.16's integrated zero-configuration network switch is custom-designed for broadcasting — no switch setup required. Supports Simple Networking, allowing up to 4 QOR engines to be daisy-chained without the need for a separate core switch.
- Each QOR.16 can support two connected RAQ or DESQ consoles, or one of each.
- Fan-free, convection-cooled power supply for noiseless in-studio operation.

IN DEPTH

A big console for small spaces.

Not every studio requires a full-size mixing console. Not every studio is full-size, itself! But you still want the advantages of IP-Audio networking: the ability to send program audio to other studios, the ability to consume audio from satellite downlinks, remote codecs and phone hybrids, or to trigger routing scene changes from a user-mapped control panel. And you don't want a toylike plastic pro-audio mixer — you want a real broadcast console that fits into a rack or turret, or on a small desktop space. A console with a small footprint, but big capabilities.

DESQ is a compact, special-purpose IP console from Axia. It may be compact in stature, but it's big on features and performance. DESQ has "big board" capabilities you won't find in other consoles of this size — automatic per-fader mix-minus, built-in EQ for voice and codec sources, and the ability to instantly load new local or networked sources to any fader with just the turn of a knob. Which means DESQ easily out-classes mixers with similar form factors — and even ones that take up much more room.

DESQ is a six-fader console in a form-factor that lets it fit just about anywhere there's a few inches of spare space: DESQ is only 16 inches (39.9 cm) square. It's built Axia-tough, with a machined-aluminum work surface that takes the rough stuff jocks can dish out. Our familiar 100 mm. conductive-plastic faders feel like silk under the fingertips, and you'll also find the avionics-grade switches with LED lighting that have become an Axia hallmark.

Other features include OLED channel and meter displays, four-source monitor section with two external locations that can be reassigned "on the fly", and an OLED time-of-day clock and event timer. Like its rackmount cousin, RAQ, DESQ also has four Show Profile console snapshot locations, and push-and-turn Options knobs at the top of each fader that give instant access to fader source assignments, pan/balance, and input gain trim.

Despite all these features, DESQ is so cost-effective, broadcasters are coming up with creative, new uses for them. Its big features and small footprint make DESQ the perfect console for interview studios, live performance spaces for on-air broadcast, news and feature production — whatever. Take it on road trip remotes, or to sporting events where multiple mics are required. Or put it in mobile units or ENG kits. Perfect for personal production studios, too.

Audio I/O, GPIO, console CPU, super-duty power supply, and even a network switch are all built into the QOR.16. Just plug in your mics, CD players, codecs, profanity delays, whatever. There are 16 audio I/O ports: two Mic inputs with switchable Phantom power, eight analog inputs and four analog outputs, and one AES/EBU input and output. QOR.16 also has four GPIO logic ports for machine control of studio peripherals, six 100Base-T ports for Livewire devices, and two Gigabit ports with SFP for connection to the outside world. For more I/O, just add Telos Alliance xNode interfaces. And you can daisy chain as many as four QOR engines without the need for an external Ethernet switch, making installation even more economical. And here's the kicker: one QOR.16 can power two DESQ mixers — or a DESQ and a RAQ (DESQ's six-fader, rackmount cousin)!

SPECIFICATIONS

QOR.16 Connections

- Microphone Inputs: 2x balanced XLR-F, with selectable Phantom power
- Analog Inputs: 8x RJ-45, StudioHub+ standard.
- Analog Outputs: 4x RJ-45, StudioHub+ standard.
- AES/EBU Inputs: 1x RJ-45, StudioHub+ standard.
- AES/EBU Outputs: 1x RJ-45, StudioHub+ standard.
- GPIO: 4x DB-15
- Livewire:
    - 4x 100Base-T with PoE, RJ-45
    - 2x 100Base-T, RJ-45
    - 2x 1000Base-T, RJ-45
    - 2x Gigabit, SFP (Small Form Pluggable)
- Console Frame Connections: 1x, 6-pin “latch and lock” style
- Accessory Connections: 1x, 6-pin “latch and lock” style

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
- Output Level: +4 dBu, nominal

Analog Line Inputs

- Input Impedance: 20 k Ohms
- Nominal Level 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: AES-3 (AES/EBU)
- AES-3 Input Compliance: 24-bit with selectable sample rate conversion, 20 kHz to 216kHz input sample rate capable.
- AES-3 Output Compliance: 24-bit
- Digital Reference: Internal (network timebase) or external reference 48 kHz, +/- 2 ppm
- Internal Sampling Rate: 48 kHz
- Output Sample Rate: 48 kHz
- A/D Conversions: 24-bit, Delta-Sigma, 256x oversampling
- D/A Conversions: 24-bit, Delta-Sigma, 256x oversampling
- Latency <3 ms, mic in to monitor out, including network and processor loop

Frequency Response

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

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: 125 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, Stereo Separation And CMRR

- Analog Line channel to channel isolation: 90 dB isolation minimum, 20 Hz to 20 kH
- Microphone channel to channel isolation: 80 dB isolation minimum, 20 Hz to 20 kHz
- Analog Line Stereo separation: 85 dB isolation minimum, 20Hz to 20 kHz
- Analog Line Input CMRR: >50 dB, 20 Hz to 20 kHz
- Microphone Input CMRR: >50 dB, 20 Hz to 20 kHz

Audio Processing

- Mic Equalizer (applicable to up to 6 faders)
- Frequency Bands: 20Hz to 320Hz, 125Hz to 2KHz, 1.25KHz to 20KHz.
- Cut/Boost range on each band: -25dB to +15dB.
- Q-factor: Automatic - bandwidth varies based on amount of cut or boost.

Power Supply AC Input, QOR.16 with DESQ Console

- Auto-sensing supply, 90VAC to 240VAC, 50 Hz to 60 Hz, IEC receptacle, internal fuse
- Power consumption: 100 Watts

Operating Temperatures

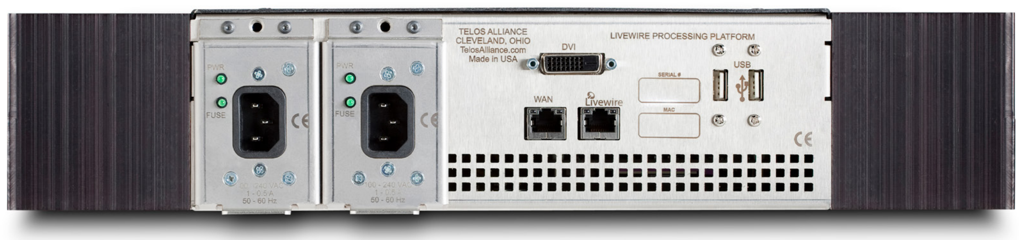
- -10 degrees C to +40 degrees C, <90% humidity, no condensation

Dimensions

- W 15.31in (38.9 cm), H 15.30 in (38.87 cm), D 2.79 in (7.1 cm)

Axia® StudioEngine

Bulletproof Power for 24/7 Operation.



OVERVIEW

The networked Axia StudioEngine provides bulletproof mixing console signal processing for Fusion™ mixing consoles. Each StudioEngine is equipped with multiple simultaneous inputs, outputs, mix-minus feeds, monitor signals, etc. and can provide EQ for multiple channels, voice dynamics, studio headphone processing and multiple VMix (Virtual Mixer) channels.

The StudioEngine is fanless for cool, silent in-studio deployment, and is equipped with Gigabit Ethernet ports for network connection, and dual-redundant, field-replaceable internal power supplies with automatic switching for complete peace of mind.

In addition to providing mixing for physical console surfaces, StudioEngine can be used in conjunction with Axia SoftSurface Virtual Console software to create a powerful “virtual console” of up to 48 faders that’s controlled with a standard Windows-based laptop — perfect for places where space constraints do not permit a physical mixing device.

FEATURES

- Fanless design with heavy machined heat-sinks is completely silent in-studio.
- Front-panel OLED display delivers monitors power status, operating temperature, and a fast manual setup option.
- Telecom grade dual-redundant power supplies are designed for maximum uptime under harsh conditions. Automatic, seamless switching.
- Field-replaceable, internally fused fault-protected power modules change out in under 1 minute.
- Dedicated Gigabit LAN interface for Livewire and System Administration
- Fusion 3.1 software update adds AES67 support
- AES67 Support.

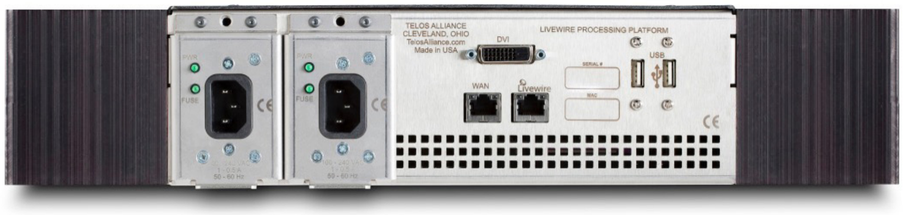
IN DEPTH

Mixing Power, a la carte.

Although Axia mixing consoles resemble traditional broadcast consoles, no audio is actually mixed by or even passes through their faders. Instead, think of Axia consoles as a "remote control" for Axia DSP-based mixing engines.

The rugged StudioEngine is a standalone mixing engine for use with Axia control surfaces; it has no audio I/O of its own, instead allowing you maximum flexibility in designing your audio network with a la carte I/O using Telos Alliance® xNode IP-Audio interfaces. In this way, you can construct bespoke systems to suit your specific needs.

The StudioEngine itself is an extremely powerful mixing device, based on a blazingly-fast Intel processor that can out-perform even the largest dedicated-DSP embedded designs. The StudioEngine 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). 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 equipped 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. for consoles as large as 40 faders. It can even provide EQ for and voice dynamics for multiple audio channels, as well as multiple VMix (virtual mixer) channels that allow combination of multiple audio channels on "virtual faders" that can then be mapped to a single physical fader. One StudioEngine supplies mixing power for even the largest Fusion.

Additionally, StudioEngine can be paired with Axia SoftSurface software to create a "virtual console" controlled by any computer with the Windows operating system — an ideal way of putting big mixing power into very small spaces.

SPECIFICATIONS

Power Supply AC Input

- Auto-sensing supply, 90VAC to 240VAC, 50 Hz to 60 Hz, IEC receptacle, internal fuse
- Power consumption: 100 Watts

Operating Temperatures

- -10 degrees C to +40 degrees C, <90% humidity, no condensation

Dimensions (HxWxD) and Weight

- 3.5 x 19 x 15 inches, 15 pounds

Network Interface

- 1x 1000BASE-T ports, standard RJ-45 connectors

Audio Processing

Equalizer

- Frequency Bands: 20Hz to 320Hz, 125Hz to 2KHz, 1.25KHz to 20KHz
- Cut/Boost range on each band: -25dB to +15dB
- Q-factor: Automatic - bandwidth varies based on amount of cut or boost

Compressor

- Threshold: -30dB to 0dB Ratio: 1:1 to 16:1
- Post-processor Trim Level: Adjustable from -20dB to +20dB

Expander/Noise Gate

- Threshold: -50dB to 0dB Ratio: -30dB to 0dB

De-esser

- Threshold: -20dB to 0dB Ratio: 1:1 to 8:1

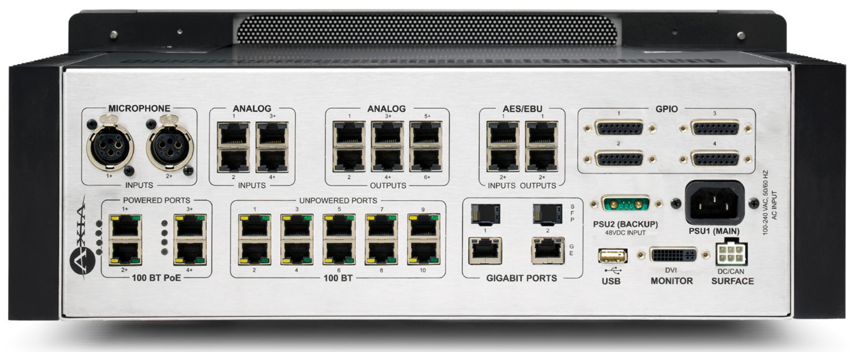
Regulatory

North America: FCC and CE tested and compliant, power supply is UL approved.

Europe: Complies with the European Union Directive 2002/95/EC on the restriction of the use of certain hazardous substances in electrical and electronic equipment (RoHS), as amended by Commission Decisions 2005/618/EC, 2005/717/ EC, 2005/747/EC (RoHS Directive), and WEEE.

Axia® PowerStation®

Integrated Console Engine



AES67 Livewire+

OVERVIEW

PowerStation is an all-in-one studio solution that combines audio I/O, a console power supply, mixing engine and built-for-broadcast network switch into one easy-to-deploy package. Each PowerStation Main provides 4 Analog inputs and 6 Analog outputs, 2 AES/EBU inputs and 2 AES/EBU outputs, 2 Microphone inputs with selectable Phantom power, 4 GPIO machine-control logic ports, each with 5 inputs and 5 outputs, an integrated network switch with 14 100BASE-T Ethernet ports and 2 1000BASE-T (Gigabit) ports with SFP, a heavy-duty Telecom-grade power supply with fanless convection cooling, and an industrial-grade CPU designed for harsh-environment reliability.

Use PowerStation Main with a Fusion mixing console as a standalone studio solution, or connect to other Axia equipment as part of a larger IP-Audio network. Simple Networking allows daisy-chain connection of up to 4 PowerStation-based studios without the use of an external network switch. Connecting a PowerStation Aux adds auto-switching redundant backup power and doubles audio I/O and GPIO capacity. I/O can also be easily expanded using Telos Alliance® xNodes.

FEATURES

- Fanless design with heavy machined heat-sinks is completely silent in-studio.
- Front-panel status display monitors power and network status.
- Telecom grade power supplies are designed for maximum uptime under harsh conditions.
- Add a PowerStation Aux to PowerStation Main for dual-redundant power supply with automatic, seamless switching.
- Add redundant power to PowerStation Main without adding additional IO with Axia Console Power Supply.
- Built-in, zero-configuration network switch with Gigabit and SFP for long-distance fiber connection.
- Large variety of built-in audio I/O boasts studio-grade audio performance specs.
- Add more I/O with PowerStation Aux, or a la carte using Telos Alliance xNodes.
- Fusion 3.1 software update adds AES67 support.
- AES67 Support

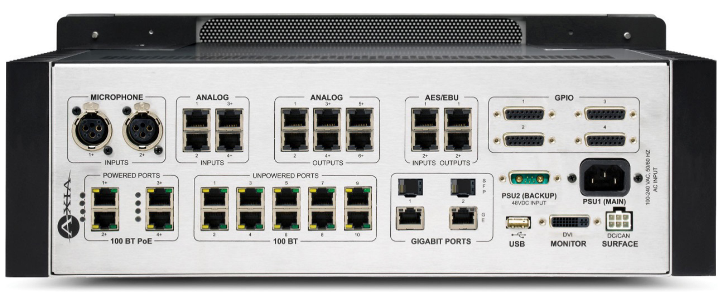
IN DEPTH

There's no such thing as too much uptime.

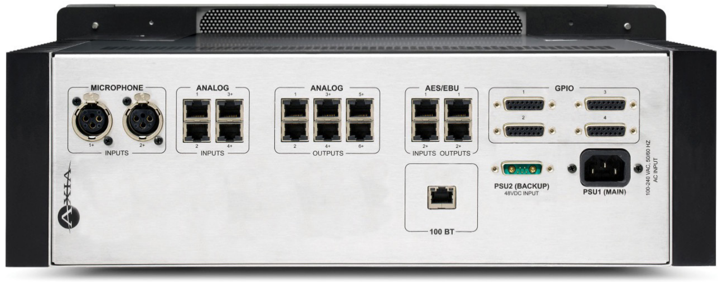
If you set out to build a console engine designed to power your studio 24 hours a day, 7 days a week, 52 weeks a year, you probably wouldn't skimp. You'd equip it with the most bulletproof, telecom-grade power supply you could find. You'd give it a redundant-power option, for even more peace of mind. You'd make it convection-cooled — no noisy cooling fans to assault your quiet studio. You'd give it plenty of I/O — analog, digital, Mic-level and GPIO logic. And then, the pièce de résistance: you'd equip it with a zero-configuration, built-for-broadcast Ethernet switch.

That's what we did when we designed PowerStation, the muscle behind our industry-leading Fusion mixing consoles. PowerStation is over-engineered to Axia standards, every part chosen for its ability to give constant, uninterrupted service. PowerStation combines four separate devices – a DSP mixing engine, a console CPU and power supply, audio I/O, GPIO and a custom, Axia-designed Ethernet switch – into a self-contained console engine that's engineered to ensure years of reliable, trouble-free service.

There are no compromises: PowerStation uses only best-of-the-best components, like studio-grade mic preamps and 24-bit, 256x oversampling A/D converters, a rigid, EM-tight chassis, an ultra-reliable DSP platform (not a common PC motherboard) and a hardened power supply designed for unfailing service, even in the harshest environments.



PowerStation Main is where you start. Inside is a bulletproof mixing engine capable of powering consoles of up to 40 faders. There's a massive fanless, convection-cooled power supply. There are two Mic inputs, four Analog inputs and six outputs, two AES/EBU inputs and two outputs, and four GPIO ports, each with five opto-isolated inputs and five opto-isolated outputs. There are 14 100BASE-T Ethernet ports with Livewire® for single-cable connection of Telos® phone systems, Omnia® audio processors and other Axia equipment, as well as gear from our huge list of Livewire partners. Two Gigabit ports with SFP enable connection to other studios via copper or fiber. Just connect it to your console (it only takes a single cable), plug in your audio devices, and perform some fast web-based configuration. Add power and you're on the air. It's that simple!



To beef up your PowerStation studio even further, there's PowerStation Aux. Connect it to the PowerStation main to instantly double mic, analog, AES and GPIO ports, and add a redundant backup power supply with auto-switchover. Most redundant supplies protect only the console, but with PowerStation, the mixing engine, audio I/O and network switch are protected as well. You can also add redundant power to PowerStation Main without additional IO with Axia Console Power Supply, which offers a single-cable connection to PowerStation Main, providing backup power with automatic switching. (Auto-sensing power supply, 90VAC to 240VAC, 50 Hz to 60 Hz. 250 Watts, 2RU.)

Best of all, there's that zero-configuration Ethernet switch that's built specifically to handle IP-Audio. No settings to tweak, no configuration code to upload – just plug it in and go. There are even two Gigabit ports with SFP, to connect to other studios via fiber or copper. You can even daisy-chain up to four PowerStation studios directly, for a self-contained network that doesn't require an external Ethernet switch. No other console company makes AoIP this easy.

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
- Output Level: +4 dBu, nominal

Analog Line Inputs

- Input Impedance: >40 k Ohms, balanced
- Nominal Level 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 Ohms, balanced (XLR)
- Signal Format: AES-3 (AES/EBU)
- AES-3 Input Compliance: 24-bit with selectable sample rate conversion, 32 kHz to 96kHz input sample rate capable.
- AES-3 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
- Latency <3 ms, mic in to monitor out, including network and processor loop

Frequency Response

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

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, Stereo Separation and CMRR

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

Audio Processing

Equalizer

- Frequency Bands: 20Hz to 320Hz, 125Hz to 2KHz, 1.25KHz to 20KHz.
- Cut/Boost range on each band: -25dB to +15dB.
- Q-factor: Automatic - bandwidth varies based on amount of cut or boost.

Compressor

- Threshold: -30dB to 0dB Ratio: 1:1 to 16:1
- Post-processor Trim Level: Adjustable from -20dB to +20dB

Expander/Noise Gate

- Threshold: -50dB to 0dB Ratio: -30dB to 0dB

De-esser

- Threshold: -20dB to 0dB Ratio: 1:1 to 8:1

Power Supply AC Input, PowerStation Aux & Main

- Auto-sensing supply, 90VAC to 240VAC, 50 Hz to 60 Hz, IEC receptacle, internal fuse
- Power consumption: 500 Watts

Axia Console Power Supply

- Add redundant power to PowerStation main without additional IO.
- Single-cable connection to PowerStation main provides backup power with automatic switching.
- Auto-sensing power supply, 90VAC to 240VAC, 50 Hz to 60 Hz.
- Power consumption: 250 Watts.

Operating Temperatures

- -10 degrees C to +40 degrees C, <90% humidity, no condensation

Dimensions (HxWxD) and Weight

- PowerStation Main/Aux: 7 x 19 x 15.5 inches (behind rail)
- Front panel extends 2.25 inches in front of rack rail
- PowerStation Main: 45 pounds
- PowerStation Aux: 40 pounds

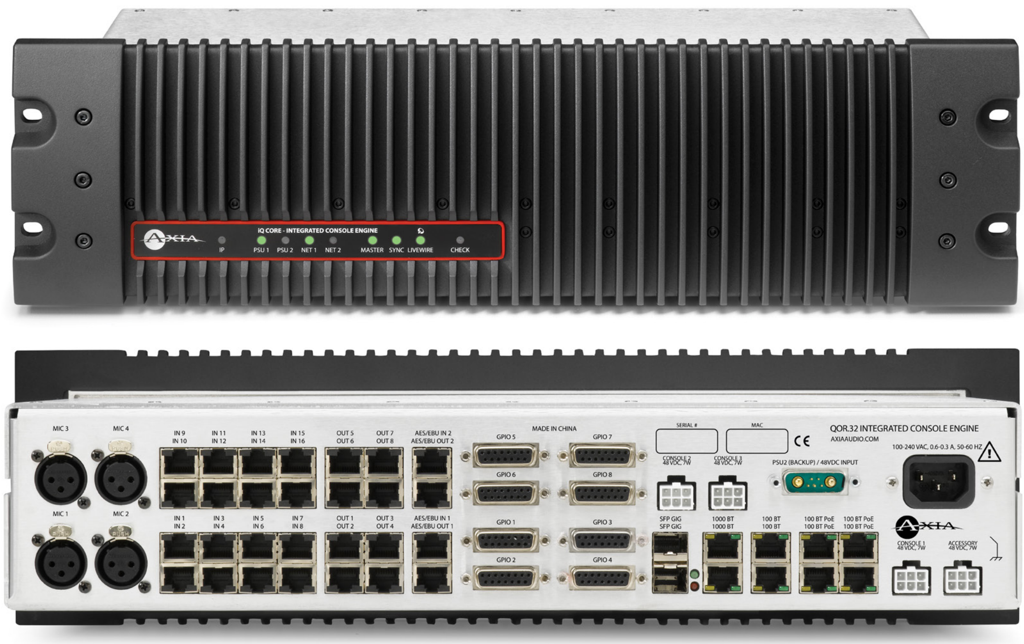
Regulatory

North America: FCC and CE tested and compliant, power supply is UL approved.

Europe: Complies with the European Union Directive 2002/95/EC on the restriction of the use of certain hazardous substances in electrical and electronic equipment (RoHS), as amended by Commission Decisions 2005/618/EC, 2005/717/ EC, 2005/747/EC (RoHS Directive), and WEEE.

Axia® QOR.32

Integrated Console Engine



OVERVIEW

QOR.32 is an Axia integrated console engine for iQ mixing consoles that combines audio I/O, a console power supply, mixing engine and built-for-broadcast network switch into one easy-to-deploy package. Each QOR.32 provides 16 Analog inputs and 8 Analog outputs, 2 AES/EBU inputs and 2 AES/EBU outputs, 4 Microphone inputs with selectable Phantom power, 8 GPIO machine-control logic ports, each with 5 inputs and 5 outputs, an integrated network switch with 6 Livewire® 100BASE-T Ethernet ports and 2 1000BASE-T (Gigabit) ports with SFP, a heavy-duty Telecom-grade power supply with fanless convection cooling, and an industrial-grade CPU designed for harsh-environment reliability.

Use QOR.32 with an Axia iQ mixing console as a standalone studio solution, or connect to other Axia equipment as part of a larger IP-Audio network. Simple Networking allows daisy-chain connection of up to 4 QOR-based studios without the use of an external network switch. Connecting a QOR Backup adds auto-switching redundant backup power. I/O can easily be expanded using Axia Audio Nodes.

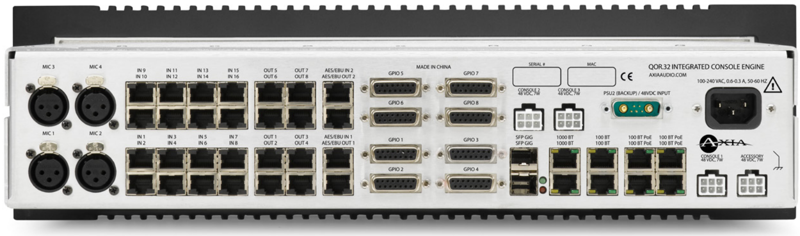
FEATURES

- Fanless design with heavy machined heat-sinks is completely silent in-studio.
- Front-panel LED display monitors power and network status.
- Telecom grade power supplies are designed for maximum uptime under harsh conditions.
- Add an Axia Console Power Supply Backup to QOR.32 for dual-redundant power supply with automatic, seamless switching.
- Built-in, zero-configuration network switch with Gigabit and SFP for long-distance fiber connection.
- Large variety of built-in audio I/O boasts studio-grade audio performance specs.
- Add more I/O a la carte using Telos Alliance® xNodes.
- Software upgrade adds AES67 support, allowing the QOR.32 integrated console engine to receive and transmit AES67 streams via Livewire+™ AES67.
- Automix allows operators to automatically and efficiently balance the levels of on-air-sources when more than one source is open at a time in a studio.

IN DEPTH

QOR.32 Integrated Console Engine

The QOR.32 integrated console engine is a DSP-based mixing engine with onboard I/O, GPIO, console power supply and custom-built, configuration-free Ethernet switch. You'll find plenty of I/O, including mic inputs with selectable Phantom power, analog and AES/EBU inputs and outputs, plenty of GPIO machine-control logic ports, and that powerful integrated Ethernet switch with Livewire ports to add local sources, and Gigabit ports for networking with the rest of your plant. That's a lot of I/O, but if you need more you can instantly add it just by plugging in Telos Alliance xNode audio interfaces. And QOR.32 is convection-cooled for utterly silent, fan-free operation.



Let's take a look around back, shall we? You'll find everything you need for an average, medium-sized studio: 4 mic inputs with selectable Phantom power, 16 stereo analog inputs and 8 stereo analog outputs, 2 AES/EBU inputs and 2 AES/EBU outputs, 8 GPIO machine-control logic ports (each with 5 opto-isolated inputs and 5 outputs).

There's Livewire I/O as well: the QOR.32 has an integrated Ethernet switch with 6 Livewire 100BASE-T ports. 4 of those ports have PoE (Power over Ethernet) that you can use to connect and power networked devices compatible with the IEEE 802.1af PoE standard (like our xNode audio interfaces, or Telos VSet phones). You'll also find 2 1000BASE-T Gigabit ports (RJ-45 & SFP) that you can use to connect to other studios. 4 CANBus ports provide for connection of up to 3 Axia iQ console frames, allowing construction of consoles up to 24 faders in size.

By the way, that zero-configuration Ethernet switch is built specifically to handle IP-Audio. No settings to tweak, no configuration code to upload – just plug it in and go. The built-in configurable network gateway allows loading sources from other studios, while simultaneously exporting audio streams for use elsewhere; the gateway can be configured for 12-in, 4-out or 8-in, 8-out modes. You can even daisy-chain up to four QOR-based studios directly, for a self-contained network that doesn't require an external Ethernet switch. No other console company makes AoIP this easy.

For installations that require redundant backup power, there's the Axia Console Power Supply. Connect it to the QOR.32 and you've added a redundant backup power supply with auto-switchover. Single-cable connection to QOR.32 console engine provides backup power with automatic switching. (Auto-sensing power supply, 90VAC to 240VAC, 50 Hz to 60 Hz. 250 Watts. Rackmount, 2RU.) Most redundant supplies protect only the console, but with Axia's integrated console engine concept, the mixing engine, local audio I/O and network switch are protected as well.

SPECIFICATIONS

QOR.32 Connections

- Microphone Inputs: 4x balanced XLR-F, with selectable Phantom power
- Analog Inputs: 16x RJ-45, StudioHub+ standard.
- Analog Outputs: 8x RJ-45, StudioHub+ standard.
- AES/EBU Inputs: 2x RJ-45, StudioHub+ standard.
- AES/EBU Outputs: 2x RJ-45, StudioHub+ standard.
- GPIO: 8x DB-15
- Livewire:
    - 4x 100BASE-T with PoE, RJ-45
    - 2x 100BASE-T, RJ-45
    - 2x 1000BASE-T, RJ-45
    - 2x Gigabit, SFP (Small Form Pluggable)
- Console Frame Connections: 3x, 6-pin "latch and lock" style
- Accessory Connections: 1x, 6-pin "latch and lock" style

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
- Output Level: +4 dBu, nominal

Analog Line Inputs

- Input Impedance: 20 k Ohms
- Nominal Level 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: AES-3 (AES/EBU)
- AES-3 Input Compliance: 24-bit with selectable sample rate conversion, 20 kHz to 216kHz input sample rate capable.
- AES-3 Output Compliance: 24-bit
- Digital Reference: Internal (network timebase) or external reference 48 kHz, +/- 2 ppm
- Internal Sampling Rate: 48 kHz
- Output Sample Rate: 48 kHz
- A/D Conversions: 24-bit, Delta-Sigma, 256x oversampling
- D/A Conversions: 24-bit, Delta-Sigma, 256x oversampling
- Latency <3 ms, mic in to monitor out, including network and processor loop

Frequency Response

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

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: 125 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, Stereo Separation And CMRR

- Analog Line channel to channel isolation: 90 dB isolation minimum, 20 Hz to 20 kH
- Microphone channel to channel isolation: 80 dB isolation minimum, 20 Hz to 20 kHz
- Analog Line Stereo separation: 85 dB isolation minimum, 20Hz to 20 kHz
- Analog Line Input CMRR: >50 dB, 20 Hz to 20 kHz
- Microphone Input CMRR: >50 dB, 20 Hz to 20 kHz

Audio Processing

- Mic Equalizer (applicable to up to 6 faders)
- Frequency Bands: 20Hz to 320Hz, 125Hz to 2KHz, 1.25KHz to 20KHz.
- Cut/Boost range on each band: -25dB to +15dB.
- Q-factor: Automatic - bandwidth varies based on amount of cut or boost.

Power Supply AC Input, QOR.32

- Auto-sensing supply, 90VAC to 240VAC, 50 Hz to 60 Hz, IEC receptacle, internal fuse
- Power consumption: 100 Watts

Axia Console Power Supply

- Auto-sensing supply, 90VAC to 240VAC, 50 Hz to 60 Hz, IEC receptacle, internal fuse
- Power consumption: 250 Watts

Operating Temperatures

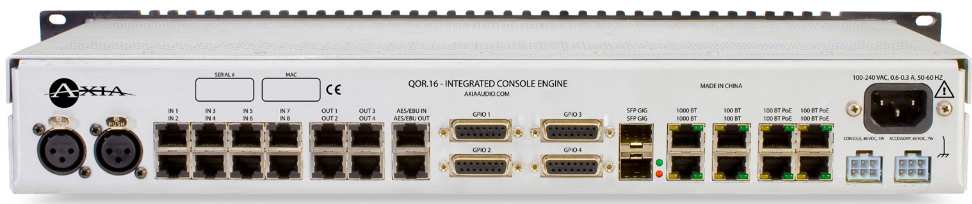
- -10 degrees C to +40 degrees C, <90% humidity, no condensation

Regulatory

North America: FCC and CE tested and compliant, power supply is UL approved.

Europe: Complies with the European Union Directive 2002/95/EC on the restriction of the use of certain hazardous substances in electrical and electronic equipment (RoHS), as amended by Commission Decisions 2005/618/EC, 2005/717/ EC, 2005/747/EC (RoHS Directive), and WEEE.

Axia® QOR.16
Integrated Console Engine



OVERVIEW

QOR.16 is an Axia integrated console engine for Radius, DESQ and RAQ mixing consoles. QOR.16 combines audio I/O, a console power supply, mixing engine and built-for-broadcast network switch into one easy-to-deploy package. Each QOR.16 provides 8 Analog inputs and 4 Analog outputs, 1 AES/EBU input and 1 AES/EBU output, 2 Microphone inputs with selectable Phantom power, 4 GPIO machine-control logic ports, each with 5 inputs and 5 outputs, an integrated network switch with 6 Livewire® 100BASE-T Ethernet ports and 2 1000BASE-T (Gigabit) ports with SFP, a heavy-duty Telecom-grade power supply with fanless convection cooling, and an industrial-grade CPU designed for harsh-environment reliability.

Use QOR.16 with a Radius, DESQ or RAQ mixing console as a standalone studio solution, or connect to other Axia equipment as part of a larger IP-Audio network. Simple Networking allows daisy-chain connection of up to 4 QOR-based studios without the use of an external network switch. I/O can easily be expanded using Telos Alliance® xNodes.

FEATURES

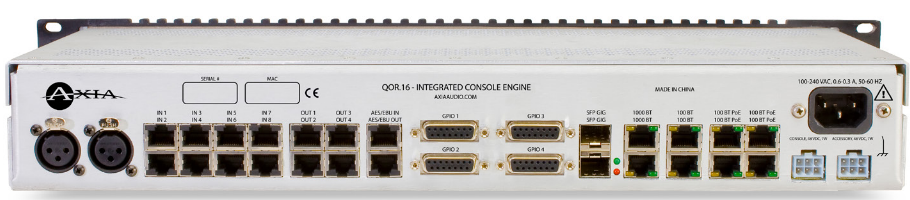
- Fanless design with heavy machined heat-sinks is completely silent in-studio.
- Front-panel LED display monitors power and network status.
- Telecom grade power supply is designed for maximum uptime under harsh conditions.
- PoE (Power over Ethernet) capability can supply power for PoE-compliant studio devices.
- Built-in, zero-configuration network switch with Gigabit and SFP for long-distance fiber connection.
- Large variety of built-in audio I/O boasts studio-grade audio performance specs.
- Add more I/O a la carte using Telos Alliance xNodes.
- Software upgrade adds AES67 support, allowing the QOR.16 integrated console engine to receive and transmit AES67 streams via Livewire+™ AES67.
- Automix allows operators to automatically and efficiently balance the levels of on-air-sources when more than one source is open at a time in a studio.

IN DEPTH

QOR.16 Integrated Console Engine

The QOR.16 integrated console engine is a DSP-based mixing engine with onboard I/O, GPIO, console power supply and custom-built, configuration-free Ethernet switch. It's the smaller brother of our QOR.32 integrated console engine, designed and built with the same high-grade components for deployment with Radius, DESQ and RAQ consoles in smaller studios where large amounts of I/O are not required.

QOR.16 comes with a wide variety of I/O, including mic inputs with selectable Phantom power, analog and AES/EBU inputs and outputs, plenty of GPIO machine-control logic ports, and that powerful integrated Ethernet switch with Livewire ports to add local sources, and Gigabit ports for networking with the rest of your plant. If more you I/O is needed, you can instantly add it just by plugging in Telos Alliance xNode audio interfaces. And QOR.16 is convection-cooled for utterly silent, fan-free operation.



QOR.16 has all the analog and digital inputs and outputs an average small studio requires: 2 mic inputs with selectable Phantom power, 8 stereo analog inputs and 4 stereo analog outputs, 1 AES/EBU input and 1 AES/EBU output, and 4 GPIO machine-control logic ports (each with 5 opto-isolated inputs and 5 outputs).

Of course, Livewire connections are built in. The QOR.16 has an integrated Ethernet switch with 6 Livewire 100BASE-T ports. 4 of those ports have PoE (Power over Ethernet) that you can use to connect and power networked devices compatible with the IEEE 802.1af PoE standard (like our xNode audio interfaces, or Telos VSet phones). You'll also find 2 1000BASE-T Gigabit ports (RJ-45 & SFP) that you can use to connect to other studios.

By the way, that zero-configuration Ethernet switch is built specifically to handle IP-Audio. No settings to tweak, no configuration code to upload – just plug it in and go. The built-in configurable network gateway allows loading sources from other studios, while simultaneously exporting audio streams for use elsewhere; the gateway can be configured for 12-in, 4-out or 8-in, 8-out modes. You can even daisy-chain up to four QOR-based studios directly, for a self-contained network that doesn't require an external Ethernet switch. No other console company makes AoIP this easy.

And here's a neat trick: if you're building audio workstations, news bullpens or ingest facilities, where small consoles like Axia DESQ or RAQ shine, a single QOR.16 can provide mixing power for two DESQ or RAQ mixers — or one of each! Just another way choosing Axia helps stretch your equipment budget.

SPECIFICATIONS

QOR.16 Connections

- Microphone Inputs: 2x balanced XLR-F, with selectable Phantom power
- Analog Inputs: 8x RJ-45, StudioHub+ standard.
- Analog Outputs: 4x RJ-45, StudioHub+ standard.
- AES/EBU Inputs: 1x RJ-45, StudioHub+ standard.
- AES/EBU Outputs: 1x RJ-45, StudioHub+ standard.
- GPIO: 4x DB-15
- Livewire:
    - 4x 100BASE-T with PoE, RJ-45
    - 2x 100BASE-T, RJ-45
    - 2x 1000BASE-T, RJ-45
    - 2x Gigabit, SFP (Small Form Pluggable)
- Console Frame Connections: 1x, 6-pin "latch and lock" style
- Accessory Connections: 1x, 6-pin "latch and lock" style

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
- Output Level: +4 dBu, nominal

Analog Line Inputs

- Input Impedance: 20 k Ohms
- Nominal Level 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: AES-3 (AES/EBU)
- AES-3 Input Compliance: 24-bit with selectable sample rate conversion, 20 kHz to 216kHz input sample rate capable.
- AES-3 Output Compliance: 24-bit
- Digital Reference: Internal (network timebase) or external reference 48 kHz, +/- 2 ppm
- Internal Sampling Rate: 48 kHz
- Output Sample Rate: 48 kHz
- A/D Conversions: 24-bit, Delta-Sigma, 256x oversampling
- D/A Conversions: 24-bit, Delta-Sigma, 256x oversampling
- Latency <3 ms, mic in to monitor out, including network and processor loop

Frequency Response

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

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: 125 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, Stereo Separation And CMRR

- Analog Line channel to channel isolation: 90 dB isolation minimum, 20 Hz to 20 kH
- Microphone channel to channel isolation: 80 dB isolation minimum, 20 Hz to 20 kHz
- Analog Line Stereo separation: 85 dB isolation minimum, 20Hz to 20 kHz
- Analog Line Input CMRR: >50 dB, 20 Hz to 20 kHz
- Microphone Input CMRR: >50 dB, 20 Hz to 20 kHz

Audio Processing

- Mic Equalizer (applicable to up to 6 faders)
- Frequency Bands: 20Hz to 320Hz, 125Hz to 2KHz, 1.25KHz to 20KHz.
- Cut/Boost range on each band: -25dB to +15dB.
- Q-factor: Automatic - bandwidth varies based on amount of cut or boost.

Power Supply AC Input, QOR.16 With Radius Console

- Auto-sensing supply, 90VAC to 240VAC, 50 Hz to 60 Hz, IEC receptacle, internal fuse
- Power consumption: 100 Watts

Operating Temperatures

- -10 degrees C to +40 degrees C, <90% humidity, no condensation

Regulatory

North America: FCC and CE tested and compliant, power supply is UL approved.

Europe: Complies with the European Union Directive 2002/95/EC on the restriction of the use of certain hazardous substances in electrical and electronic equipment (RoHS), as amended by Commission Decisions 2005/618/EC, 2005/717/ EC, 2005/747/EC (RoHS Directive), and WEEE.

Telos Alliance® xNode™ IP-Audio Interfaces

The Most Advanced AoIP Interfaces on the Planet.

OVERVIEW

The xNode lightweight, half-rack, high-performance IP-Audio interface from Telos Alliance is loaded with advanced features and capabilities. One-button configuration takes a new xNode from out-of-the-box to on-the-air in under one minute. They're fanless, which means they're noiseless too. Versatile mounting options let you deploy two xNodes in just 1RU of rack space, or on ceilings, walls, and under counters with an available wall-mount kit. xNodes have studio-grade audio performance specs. Redundant power options (using AC mains and Power-over-Ethernet) and dual-redundant network interfaces are included, both with automatic switching. And xNodes are fully AES67-compliant, so they work with all AES67 audio gear—now, and in the future. In fact, they are the first and only AoIP I/O devices that are Livewire+™ AES67, RAVENNA, and AES67 compliant. Every xNode not only supports RAVENNA audio stream interoperability, but also enables advertising/discovery of those streams natively, above and beyond AES67.

xNodes are available in Analog, AES/EBU, Microphone-level, Mixed-Signal, and GPIO versions to handle virtually any signal encountered in today's broadcast studio.

FEATURES

- Fanless design with cast-aluminum heat-sinks is completely silent in-studio. Front-panel heat sinks are cooled by ambient air, not "rack air," eliminating overheating worries.
- World's only fully AES67-compliant AoIP interface; xNodes are "universal translators" that support a huge installed base of Livewire+ hardware as well as audio streams from other AES67-compliant devices. Now featuring improved AES67 performance through reception optimization.
- First and only AoIP I/O device that is Livewire+™ AES67, RAVENNA, and AES67 compliant. Every xNode not only supports RAVENNA audio stream interoperability, but also enables advertising/discovery of those streams natively, above and beyond AES67.
- High-resolution front-panel multi-function OLED display meters inputs and outputs or GPIO status, gives software and other status information.
- Power-efficient: xNodes use just 14 Watts each.
- Exclusive redundant power plan uses AC and Power over Ethernet (IEEE 802.3af) supplied by compliant Ethernet switches. Multi-color front-panel LED glows green when AC mains power is used, red when PoE is used, and orange when both AC and PoE are connected.
- Exclusive redundant network connection: Dual NICs allow you to connect xNode to separate network branches for full audio pathway redundancy. Automatic failover activates backup connection should the primary be interrupted.
- 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.
- Synchronize your AES master clock to a designated xNode AES/EBU input to keep all of your AES streams synchronized to the house clock.
- xNodes use premium components, including rugged cast aluminum faceplates and heat sinks, high-resolution OLED displays, bulletproof power supplies designed for high-availability telecom applications, studio-quality SRCs with recording-studio specs.
- I/O connections via industry-standard RJ-45 audio connectors or high-density DB-25 connections, both available prefabricated and ready to attach in seconds.
- Versatile mounting options: Use freestanding, rack singly or side-by-side in 1 RU, or wall-mount using an optional surface-mount kit.
- Analog xNode inputs can be configured to supply four stereo audio channels, eight true mono channels, or 5.1 surround + stereo downmix. Outputs support the same variety of selections, easily selectable in software via the built-in web interface.
- On the Analog, AES/EBU, Mixed Signal, and Microphone xNodes, a fully configurable mixing matrix allows for mixing of both physical and network inputs, stream conversion, and a multitude of other unique solutions.
- SAP Support

IN DEPTH

The AoIP Interface that's twice as powerful. (But only half the size.)

One day, all audio equipment will be networked. Until then, there are xNodes, the world's first self-configuring, fully AES67-compliant AoIP interfaces.

xNodes give you an easy way to add non-networked audio devices to your studio network. They pack a lot of I/O into a very small space. And xNodes are so simple to set up, they nearly configure themselves.

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 loaded with features designed to ensure the uptime of your network. Dual Ethernet ports can provide redundant connections to separate network segments. Redundant power capability with automatic switchover enables xNodes to run on house mains or PoE (Power over Ethernet), letting the network switch itself supply power, and enabling easy single-cable setup in places where AC power isn't practical. Built-in Syslog servers with a configurable event filter and SNMP (Simple Network Management Protocol) support let you stay fully informed, should an xNode's power or connection status change.

The xNode Matrix Mixer feature is one of the most flexible and capable virtual mixers available. It lets users mix physical inputs (like mics and playback devices) with digital network sources (like stream inputs) to a single output. With the xNode Matrix Mixer feature, broadcasters can bypass the studio console during automated dayparts and send on-air mixes straight to the transmitter thus simplifying audio workflows. This one-of-a-kind solution offers the power and flexibility of a big studio mixer switching system in a compact ½ RU device!

xNodes are convenient, too. For example, a Microphone xNode placed in a studio can take 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 provide audio quality superior to any other AoIP interface. Not only are they capable of operating at a network sampling rate of 48kHz, 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 installation.

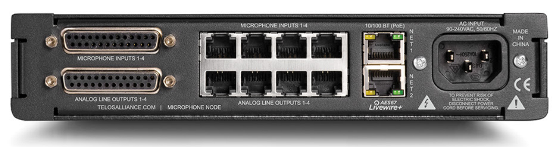
xNodes are versatile and cost-efficient. Since they're half the size of other AoIP interfaces, they cost less. And you can mix-and-match I/O as needed: Choose between analog, AES/EBU, or Mic-level inputs, without paying for ports you won't use. High-density GPIO xNodes let you easily provide logic and control for your audio source devices.

xNodes are easy to deploy, too. When you connect an xNode to your network, it automatically prompts you to give it an ID via the front-panel controls. Then, it 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.

xNodes are also fanless, so you can tuck one anywhere you need I/O without worrying about cooling fans or heat—they consume only 14 Watts of power! Two xNodes fit side-by-side in a single rack space using the included rack-mount kit. Or, mount them to walls, ceilings, or under countertops, with an optional surface-mount kit.

Five different xNodes provide analog and AES ins and outs, microphone inputs and GPIO logic ports, wherever you need them. No need for "home runs" to a central rack—one CAT-5 cable connection is all an xNode needs to interface multiple channels of bi-directional audio to your network.

Microphone xNode

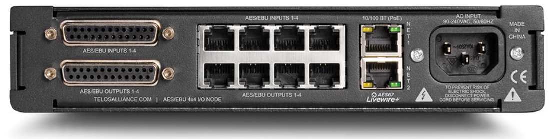


The Microphone xNode has four professional-grade microphone preamps with selectable Phantom power and software-adjustable gain. There are also four balanced analog line outputs to conveniently deliver headphone and studio monitor feeds back to your talent. Inputs and outputs are presented both on easy-to-install RJ-45s and high-density DB-25s, both of which connect to easily available 3rd-party breakout cables, to suit your connection preference.

Analog xNode

The 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. It can also accommodate 5.1 Surround inputs and outputs, each with an associated discrete Stereo mix. 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. Telos Alliance 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



Our AES/EBU xNode has 4 AES/EBU inputs and 4 AES/EBU outputs. Left and right input signals may be split and routed independently as mono signals. Stunning performance specs include 48 kHz sampling rate, 126dB of dynamic range, and <0.0003% THD. Sample rate conversion is available on all inputs; the unit can also be synchronized to a house clock to provide sync to your entire Axia network.

Mixed-Signal xNode

The 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.

SPECIFICATIONS

Microphone Preamplifiers

- Source Impedance: 150 Ohms
- Input Impedance: 4k 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: >40k 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
- 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 192kHz
- Output Sample Rate: 44.1 kHz or 48kHz
- 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 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, 50Hz to 60Hz, IEC Receptacle, Internal Fuse
- Power Consumption: 14 Watts

Operating Temperatures

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

Dimensions

- 8.5" (22 cm) wide; two may be mounted side-by-side in a standard 1RU rack space; 1.72" (4.4 cm) height, 11.75" (30 cm) depth

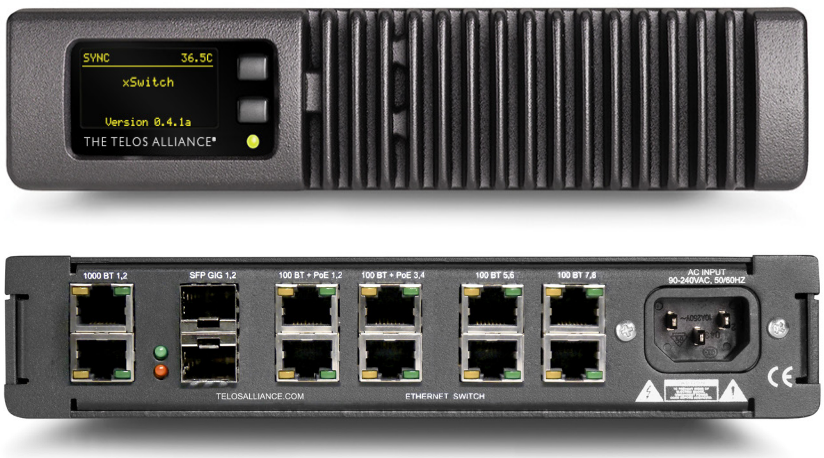
Regulatory

North America: FCC and CE tested and compliant, power supply is UL approved.

Europe: Complies with the European Union Directive 2002/95/EC on the restriction of the use of certain hazardous substances in electrical and electronic equipment (RoHS), as amended by Commission Decisions 2005/618/EC, 2005/717/ EC, 2005/747/EC (RoHS Directive), and WEEE.

Telos Alliance® xSwitch

The Network Switch Built for IP-Audio



OVERVIEW

xSwitch is the world's only zero-configuration Ethernet switch optimized for Livewire® IP-Audio applications. Fast setup requires only IP address assignment via front-panel OLED display or Axia® iProbe software. Features 8 10/100MBit Ethernet ports — 4 with Power-over-Ethernet to power Telos Alliance xNodes, Telos® VSet phones, and other networked devices compatible with the IEEE 802.1af PoE standard. Two Gigabit ports are provided for trunking, both with RJ-45 (copper) and SFP (fiber) connections; supports redundant copper/SFP Gigabit connections with auto-switching. Supports 2,000 Multicast groups and 2,000 ARP table entries (8x more than other small-form Ethernet switches). Web-based management interface uses built-in HTTP server. 9.5" x 11" half-rack form factor allows two xSwitches to be racked side-by-side, or placed in a rackmount with Telos Alliance xNode IP-Audio interfaces. Noiseless and fan-free; can be conveniently placed adjacent to your audio devices, rack-mounted using included hardware, or wall-mounted (with an accessory kit available separately).

FEATURES

- Fanless design with heavy cast-aluminum heat-sinks is completely silent in-studio. Front-panel heat sinks are cooled by ambient air, not "rack air," eliminating overheating worries.
- Friendly OLED front-panel display with port status, IP address, PoE status and operating temperature readouts - features not available in other switches in this class.
- One-button setup eliminates programming and saves hours of setup time.
- Functions as a core switch for standalone studios, or as an edge switch in larger facilities, or at your Ethernet-connected transmitter site.
- Allows Axia network admins to add network ports economically, a la carte, instead of 24 or 48 at a time.
- xSwitch supports IGMP (Internet Group Management Protocol) Version 2, used to manage Multicast group traffic (an essential part of Livewire's intelligent audio routing system).
- xSwitch can handle up to 2,000 Multicast groups, and 2,000 ARP table entries, meaning it can't run out of bandwidth. (Other 8-port switches support only 250 groups.)
- Superior support for low-latency media streams, using four-level hardware strict priority QoS — other switches have only one strict priority queue.
- Works with Axia iProbe network management software, allowing easy administration from your office PC or remotely via WAN connection.
- Part of Telos Alliance's xNode family, xSwitch can be used as a freestanding device or racked singly or side-by-side with other xSwitch or xNode devices.
- Premium components include rugged cast faceplate and heat sinks, high-resolution OLED displays, and bulletproof power supplies designed for high-availability telecom applications.

IN DEPTH

Your radio station needs programming.
Your network switch shouldn't.

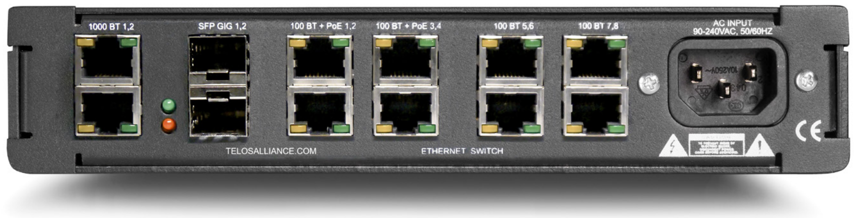
We invented Livewire in 2003, with the idea of saving money in broadcast studio construction by using off-the-shelf Ethernet switches to power networks that distribute broadcast-quality audio nearly anywhere — across the hall, across the building, or across town. Some said it would never work! But 10 years later, Axia is the #1 brand of IP consoles, networks and routing equipment to broadcasters worldwide. Maybe it's because Livewire IP-Audio is so flexible and easy to use that clients regularly tell us of days – even weeks – shaved off of studio installation time with components that simply click together using Cat-5 cables. Not to mention the money they've saved with Axia, compared to old-fashioned hard-wired studio builds.

But Axia fans told us there was one thing that could make Livewire even easier to install: A network switch that doesn't require setup or programming. So our engineers went to work. The result: xSwitch, the world's only zero-configuration network switch designed specifically for the needs of IP-Audio broadcasting.

xSwitch is different from any other Ethernet switch, because it's custom-tailored to the needs of Axia Livewire users. You see, third-party switches – even those certified for use with Axia – require programming to correctly configure them with the QoS settings Axia networks demand. Which generally means connecting a PC to the switch with a special cable, downloading a terminal emulation program, and entering lines of parameters and instructions.

Perhaps you've already got an Axia network installed (thank you!). Will an xSwitch work with the Axia gear you already have? Naturally! xNodes speak Livewire, the AoIP protocol that powers more than 50,000 networked pro audio devices at radio and TV stations around the world. One click to hook up, and they're ready to go.

xSwitch does away with switch programming. Our experts have already pre-configured xSwitch with all the instructions needed to run Livewire perfectly, flawlessly, out of the box. All you have to do is plug it in, perform a quick one-button setup, and start connecting Livewire devices. Easy, yes?



On the xSwitch's connection panel, you'll find two SFP (Small Form-Factor Pluggable) Gigabit ports, in addition to dual 1000BASE-T copper ports. Use the SFP ports for copper or fiber connections to your Livewire network. The adjacent 1000BT copper ports provide a dual-redundant network interface; if the primary network link is interrupted, the secondary backup connection is automatically activated. You'll also find 8 100-BaseT Livewire ports, 4 with PoE (Power over Ethernet) to power xNodes audio adapters, Telos VSet telephones, or any other network device that uses the IEEE 802.3af standard. Speaking of power, note the internal, auto-ranging power supply with professional IEC connector: you'll never find wall-warts powering Axia gear.

xSwitch is built using the chassis developed for our award-winning xNode family of AoIP audio adapters, the latest generation of half-rack, high-performance IP-Audio interfaces. They're fanless, which means they're noiseless too; you can put them in any studio. They have a versatile mounting arrangement that lets you deploy two xSwitches into just 1RU of rack space (or rack an xSwitch alongside an xNode). This allows you the flexibility to do things impossible before — like combine an xSwitch with xNodes to create a "Supernode". An xSwitch connecting 8 analog xNodes creates a 32x32 stereo router - or a 64x64 mono router - in the space of just 4RU. Great for making an audio snake, for adding I/O to that add-on studio on the next floor, or even as the heart of a standalone studio.

SPECIFICATIONS

Power Supply AC Input

- Auto-ranging supply, 95VAC to 240VAC, 1.0 A, 50 Hz to 60 Hz
- IEC receptacle, internal fuse
- Power consumption: 75 Watts (all PoE ports under load)

Power over Ethernet

- 15.4 W-per-port maximum, 61.6-W switch maximum

Environmental Ranges

- Operating temperature: 32° F to 104° F (0°C to 40°C),
- Relative humidity: <90% (noncondensing)

Physical Dimensions

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

Ethernet Switch Specifications

- 4 QoS levels
- VLANs supported: 1
- Hardware filter capacity: 8,000 (this is the total limit of MAC addresses + multicast group count supported).
- Supported protocols:
    - IPv4 hardware switching
    - IGMP version 2 snooping
    - IGMP snooping querier
    - DSCP (IP Type Of Service based priority)
    - 802.1p (Ethernet 802.1Q tag priority)
    - HTTP (WEB based management)

- Ports 100BT 1, 2, 3, 4: 100BASE-T Fast Ethernet (10/100MBit/s), Power-over-Ethernet source
- Ports 100BT 5, 6, 7, 8: 100BASE-T Fast Ethernet (10/100MBit/s)
- Ports GIG 1, 2: 100BASE-T Copper or SFP (Small Factor Pluggable Transceiver) module

IGMP Snooping Parameters

- Router present time out: 400s
- Query Response Interval: 10s

Connector Specifications

10/100/1000 Ports

The 10/100/1000 Ethernet ports use standard RJ-45 connectors.

Connecting to 100BASE-T-Compatible Devices

When connecting the ports to 100BASE-TX-compatible devices, you can use a two or four twisted-pair, Category 5e, straight-through cable.

Connecting to 1000BASE-T Devices

When connecting the ports to 1000BASE-T devices, you must use a four twisted-pair, Category 6, straight-through cable.

SFP Module Ports

The SFP module slot on a dual-purpose port uses SFP modules for fiber-optic and copper uplink ports. xSwitch works with the following supported SFP modules:

- Cisco Copper SFP Model:GLC-T=
- Cisco Copper SFP Model: SFP-GE-T=
- Cisco Multimode fiber model: GLC-SX-MMD=
- Cisco Multimode fiber model: GLC-SX-MM-RGD

Regulatory

North America: FCC and CE tested and compliant, power supply is UL approved.

Europe: Complies with the European Union Directive 2002/95/EC on the restriction of the use of certain hazardous substances in electrical and electronic equipment (RoHS), as amended by Commission Decisions 2005/618/EC, 2005/717/ EC, 2005/747/EC (RoHS Directive), and WEEE.

Axia® xSelector Router Panel

The Network Switch Built for IP-Audio

OVERVIEW

The Axia® xSelector combines the routing functions of an XY router control panel with the audio outputs of an Telos Alliance® xNode. In addition to analog, AES3 and headphone outputs, the Router Selector Node also features an analog and an AES3 input — ideal for production or news studios where operators both create and play audio streams. Six convenient "radio buttons" can be quickly programmed for instant access to favorite sources.

FEATURES

- Fanless design for silent in-studio operation.
- High-resolution front-panel multi-function OLED display meters inputs and outputs and provides audio source selection controls.
- Local I/O connections via industry-standard RJ-45 or XLR audio connectors,.
- In addition to being able to select audio streams from the Livewire® network for use locally, the xSelector features one stereo input and one stereo output, allowing fast network distribution of locally created streams from audio workstations or portable audio devices. Each xSelector can create 1 stereo Livewire stream, which becomes available to other devices on the Livewire network.
- Local I/O is presented on both AES digital and analog balanced inputs and outputs. The user can feed audio into either a balanced analog input or an AES input.
- Both the AES and analog outputs are active simultaneously; both outputs have the same audio present.
- Includes two GPIO closures presented on standard DB-15 connectors for machine control of associated devices.
- xSelector’s stereo outputs can be assigned to output either the locally-created audio stream, or a single stereo Livewire stream acquired from the network and easily selected from a list of available streams using the front panel OLED display.
- Six frequently-used streams can be assigned to the front panel “radio buttons” for instant access. Filmcap buttons can be labeled with names of assigned channels if desired.
- Dual Livewire 100BASE-T Ethernet ports for redundant connection to your Livewire audio network.
- Front panel headphone jack and volume control make xSelector a valuable addition to dubbing and ingest stations where minimal infrastructure is desired.
- Built-in HTTP server for easy remote control using any PC with a Web browser.

IN DEPTH

The production-room powerhouse.

The Axia xSelector looks a lot like a traditional XY router control panel, but it's much, much more. So much more, in fact, that you'll make xSelector a staple in your TOC, production rooms, news stations — anywhere your talent needs to both create and consume networked audio streams.

xSelector lets talent select from all available audio streams on the Livewire network, and route them to its local output (conveniently presented in both balanced analog and AES/EBU format). xSelector is easy to use: The front-panel LCD screen lists available network sources; talent uses the adjacent selector knob to browse sources and then pushes the knob to "take" the selected source, instantly routing that source to the local outputs for use with an audio workstation, a specific console input, recording device, etc.

Also on the front panel, six film-cap "radio buttons" provide instant access to frequently-used sources. There's also a stereo ¼" TRS jack with a volume control which supplies an internally-amplified audio output directly to talent headphones, making xSelector a perfect choice for small workstation environments by eliminating the need for an external headphone amp.

Around back, you'll find separate left and right balanced XLR and RJ-45 connections for the analog input and output, another set of XLR and RJ-45 connectors for the AES/EBU input and output, DB-15 connectors for the two GPIO machine-logic controls, and RJ-45s for the two redundant Livewire 100BASE-T Ethernet connections.

All this functionality makes xSelector the perfect choice for news booths or dubbing stations where only one active feed is required, or for intake stations that allow non-technical folks to easily move audio from external sources (like field recorders) into the Axia network.

SPECIFICATIONS

Connections:

AES/EBU

- 1x Stereo Input, presented on one XLR-F connection and one RJ-45 connection
- 1x Stereo Output, presented on one XLR-M connection and one RJ-45 connection

Analog

- 1x Stereo, presented on two XLR-F connections and one RJ-45 connection
- 1x Stereo, presented on two XLR-M connections and one RJ-45 connection

GPIO

- 2x DB-15, each with 5 opto-isolated inputs and 5 outputs

Network

- 2x 100BASE-T connections, presented on RJ-45

Audio:

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

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

Total Harmonic Distortion + Noise

- 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

Power Supply Ac Input

- Auto-ranging supply, 90VAC to 240VAC, 30 Hz to 60 Hz, IEC receptacle, internal fuse
- Power consumption: 35 Watts

Operating Temperatures

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

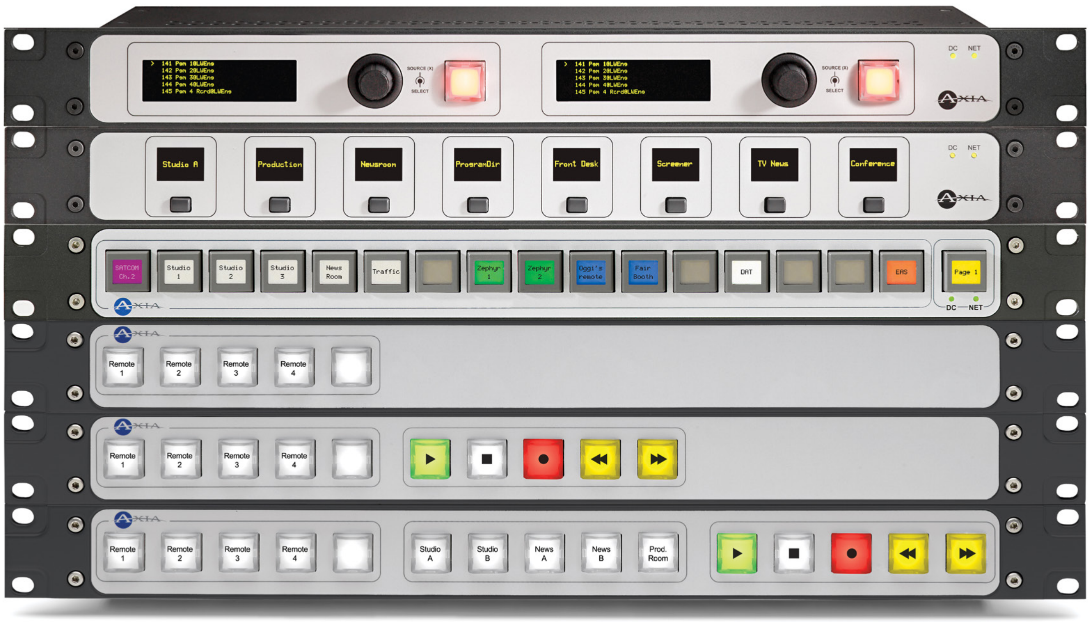
Regulatory

North America: FCC and CE tested and compliant, power supply is UL approved.

Europe: Complies with the European Union Directive 2002/95/EC on the restriction of the use of certain hazardous substances in electrical and electronic equipment (RoHS), as amended by Commission Decisions 2005/618/EC, 2005/717/ EC, 2005/747/EC (RoHS Directive), and WEEE.

Axia® Routing Control Panels

Fingertip Control, Just Where You Need It.



OVERVIEW

Axia makes it easy to build large, IP-based routing networks of up to 10,000 audio streams. We also make hardware tools to help your operators control all that networking power. Along with PathfinderPC and PathfinderPRO routing control software, Axia Routing Control Panels put fingertip control of inputs, outputs and routing scene changes anywhere you've got a rack space.

FEATURES

- Six accessory control panels for convenient talent / guest control of a variety of studio operations: routing scene changes, GPIO closures, XY control of inputs and outputs, etc.
- Slim panel design mounts in any 1RU rack space.
- Fanless, convection-cooled.

IN DEPTH

Take Control.

Axia control panels let you place routing power anywhere — in a studio turret, a TOC control panel or an equipment rack. These accessory control panels work with Axia's PathfinderPC and PathfinderPRO routing control hardware, allowing you to map routing commands – from simple contact closures to complex logic-driven events – to any button for fast execution.

Film-cap controllers with LED-backlit keys can be illuminated with a choice of colors; keycaps are film-legendable for quick function identification. SmartSwitch panels have dynamic, backlit LCD buttons that can change color and text with user activation. And the rack-mount 8-button SoftSwitch panel has high-resolution OLED buttons that can be loaded with user-created bitmaps for instant function identification. And the XY Routing Control Panel allows convenient on-the-fly routing of networked sources from anywhere in your facility; route any source to any output of your choosing with just a couple of knob twists.

17-Button LCD SmartSwitch Panel

The 17-button SmartSwitch Router Control Panel features backlit LCD buttons with dynamic text and color to provide 1-touch remote access to often-used machine-control or software functions. Multiple pages of button assignments can be programmed and recalled with just a touch; use PathfinderPC's Stacking Events Editor to map single commands or complex routing salvos to any button. Easy-to-use Web-based configuration pages can be accessed from any PC on the Livewire® network.

Film-Cap Switch Panels

Use these Film-Cap Router Control Panels when dynamic-text capabilities are not required; lighted aircraft-grade switches provide fast execution of router salvos, machine-control or software functions programmed using PathfinderPC Router Control software. 5-, 10- and 15-button rackmount models are perfect for use in a studio turret, TOC control panel or equipment rack. Place film labels under the clear button caps; set the LED backlights to any of 8 different colors.

OLED SoftSwitch Router Control Panel

The 8-Button OLED SoftSwitch provides high-visibility router control from any studio turret or equipment rack. Its eight bright, sharp OLED (Organic Light-Emitting Diode) readouts can display simple text or user-supplied monochrome bitmaps, and can be seen from nearly any angle — across the table, or across the room. Use PathfinderPC to program custom routing commands you can invoke instantly with the touch of a button.

XY Router Control Panel

XY Router Control Panel lets you route any source to any destination (any-to-any routing) with the click of a button. Choose your desired audio stream, select your network output and press "Take" to route audio. Perfect for TOC program stream selection, ingest stations where a multitude of incoming feeds need routing to air, or production rooms — anywhere you need many-to-one control of networked audio streams.

SPECIFICATIONS

General

- Rackmount package requires 1RU of free rack space.
- All Router Control Panels require 1 free 100BASE-T Ethernet port on a network switch for connection to the Axia network.
- 17-button SmartSwitch Panel requires PathfinderPC or PathfinderPRO software to program and execute conditional routing commands.
- FilmCap Button Panels require 1 free Axia GPIO port per each 5 buttons. PathfinderPC or PathfinderPRO software is not required for GPIO command of networked devices, but is required to program and execute routing commands.
- 8-Button OLED SoftSwitch requires PathfinderPC or PathfinderPRO software to program and execute conditional routing commands.

Regulatory

North America: FCC and CE tested and compliant, power supply is UL approved.

Europe: Complies with the European Union Directive 2002/95/EC on the restriction of the use of certain hazardous substances in electrical and electronic equipment (RoHS), as amended by Commission Decisions 2005/618/EC, 2005/717/ EC, 2005/747/EC (RoHS Directive), and WEEE.

Axia® Studio Control Panels

Give Your Talent The Power

Fusion Panels

OVERVIEW

Axia Studio Control Panels are a family of options panels designed for flush-mounting in desktop or turret cabinetry. They allow you to place control of headphone source selection, mic off/on control and even GPIO machine control at talent and guest desk positions, where they're most convenient.

FEATURES

- Six accessory control panels for convenient talent / guest control of frequently-changed options, including headphone and mic control, GPIO closures, routing scene changes tied to Axia PathfinderPC software, Talkback to CR board op or Guest postions.
- Easy RJ-45 connection to console CANBus control network.
- May be flush-mounted in any flat or vertical solid surface.
- All panels measure 6" x 2"; require 2" mounting depth.

IN DEPTH

Options are just a touch away.

Axia consoles are nearly synonymous with "flexibility." You can save show settings and recall them in an instant... customize backfeeds and routing salvos... share audio sources and control throughout your facility... and that's just the beginning. Axia helps you customize your studio too, with accessory control panels that work seamlessly with your consoles to give talent fast access to headphone, mic and select switching controls.

Mic Control Panel

The Mic Control panel gives talent or guests remote control of their mic channel. Press the Talkback key, and you open a comm channel to the board operator. There's a handy Mute key for those "frog-in-the-throat" moments, too. Works with all Axia consoles.

Producer's Mic Control Panel

Designed especially to suit the needs of busy talk show producers, the Producer's Mic Control panel provides control of microphone On/Off/Mute functions, and includes two special Talkback keys so producers can easily converse with studio remote talent. Works with all Axia consoles.

Headphone Selector Panel

The Headphone Selector panel lets talent control their own headphone feeds. Turn the knob and control the volume. Push the knob, scroll through the list of available sources, and push again to "take." Preset buttons are provided for instant access to two programmed sources. Works with Fusion™ consoles.

Mic Control / Headphone Selector Panel

Why choose when you can have it all? Combination Mic Control/Headphone Selector panel gives talent remote control of headphone source and volume, mic channel on/off, and includes Mute and Talkback functions. Works with Fusion consoles.

Four-Key SmartSwitch Button Panel

Four-key SmartSwitch has illuminated, dynamic LCD keys that can change text and backlight color based on conditional logic macros you construct in Pathfinder routing control software, using simple drop-down tools. Works with Fusion consoles.

SPECIFICATIONS

General

- Desktop panels require access to Axia CANBus control network via CAT-5 connection.
- Flush-desktop mounting style requires routed 6” x 2” cutout in countertop or work surface. 2” of space required behind each panel for adequate connector/cable clearance.
- Not all panels work with all Axia consoles. Consult Axia or your Axia representative for specific applications.

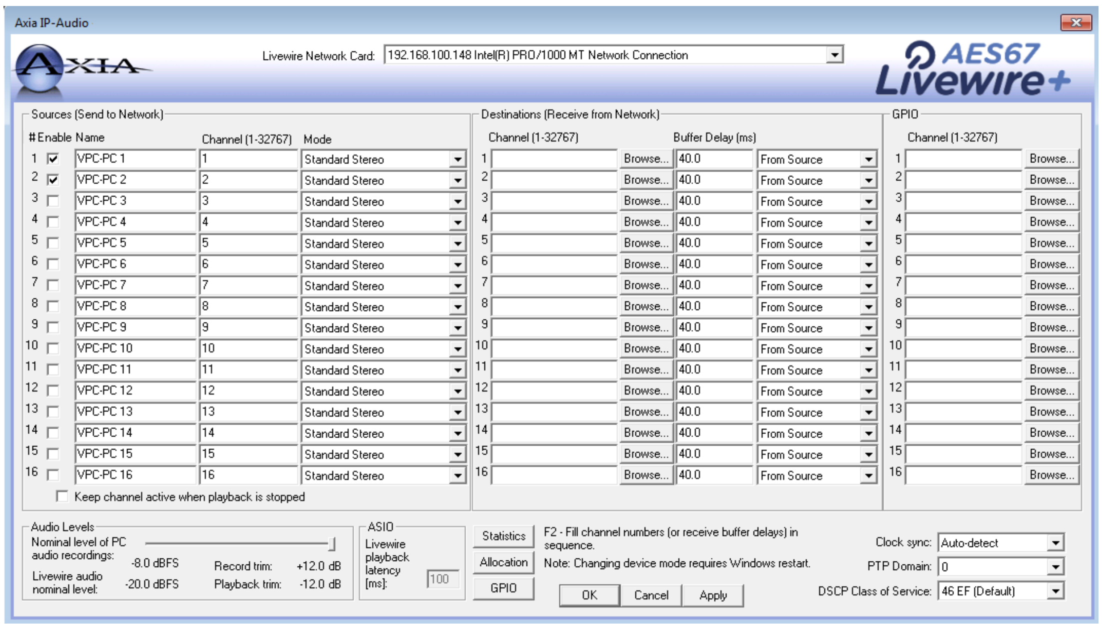
Regulatory

North America: FCC and CE tested and compliant, power supply is UL approved.

Europe: Complies with the European Union Directive 2002/95/EC on the restriction of the use of certain hazardous substances in electrical and electronic equipment (RoHS), as amended by Commission Decisions 2005/618/EC, 2005/717/ EC, 2005/747/EC (RoHS Directive), and WEEE.

Axia® Livewire+™ AES67 IP-Audio Driver

Pure Digital Audio from Networked PCs



AES67 Livewire+™

OVERVIEW

The Axia Livewire+ AES67 IP-Audio Driver is one of the first AES67-Compliant* IP Drivers. It lets you send and record single or multiple channels of stereo PC audio directly to and from Axia networks via Ethernet — no sound cards needed. Up to 24 channels of stereo audio can be sent simultaneously over a single CAT-5 Ethernet connection.

FEATURES

- AES67-compliant IP-audio driver.
- Sends audio sources to the Livewire® / Livewire+ AES67 network from PC/Windows audio applications such as multichannel delivery systems and other audio players.
- Receives audio from the Livewire / Livewire+ AES67 network to destinations on the PC/Windows system, such as audio recording applications.
- GPIO function conveys "button-press" data from the Livewire /Livewire+ AES67 network to destination applications; i.e., a console fader start button can command a PC/Windows-based audio player to start playback.
- The Axia Livewire+ AES67 IP-Audio Driver single-stream version emulates a standard sound card, with one stereo audio output device and one stereo audio input device. This version is suitable for typical two-channel (stereo) playback or recording applications.
- Axia Livewire+ AES67 IP-Audio Multichannel OEM versions emulate 4, 8 or 24 stereo channel sound cards (depending upon installed version), with one stereo audio output device and one stereo audio input device per "sound card." These versions are intended for more complex professional applications.
- Supports 5.1 surround audio streams as well as stereo, configurable on a per-stream basis.
- Windows version includes WDM and ASIO versions for maximum system flexibility.

IN DEPTH

Pristine PC Audio: No Sound Card Required

Way back when enormous cart machines still roamed the earth freely, we used XLR connectors to get recorded audio into the console. But when PCs replaced the cart machine, we continued to connect to their sound cards with plain-Jane XLRs and a thick bundle of discrete wires that can't carry logic, PAD or any of the useful information that PC playout systems provide. Why? With the Axia Livewire+ AES67 IP-Audio Driver, there's a better way, and it's now AES67-compliant. We've added the ability to sync to PTP, support to define multicast address outside of the Livewire range, and added SIP/Unicast support for RTP streaming.

The PC is the heart of the modern radio studio. And Axia makes it easy to connect and exchange pristine digital audio with it. The Axia IP-Audio Driver for Windows is a special Windows driver that feeds your digital audio directly from your PC's Ethernet port to the Livewire / Livewire+ AES67 network. Up to 24 stereo playback channels and 24 stereo record channels can be accessed using our multi-stream driver that's provided by your favorite digital delivery system provider; a single-play/single-record version is available for audio workstations.

The IP-Audio Driver also provides GPIO-like start/stop and other control functions over the same network. It's available with the latest versions of high-end Windows audio delivery and editing software applications such as those from BSI, Burli, DAVID Systems, Dalet, ENCO, iMediaTouch, Netia, RCS, WideOrbit, and Zenon Media (to name just a few) and for Linux-based Rivendell through Paravel Systems—more than 50 systems and counting.

The Windows version of the IP-Audio Driver is available to broadcasters directly from Axia in 1-Stream and 4-Stream versions, and from Axia Delivery System Partners in 8-Stream and 24-Stream versions. Linux versions are available from our partner, Paravel Systems. For a full listing of Axia Delivery System Partners, visit www.TelosAlliance.com/Partners.

SPECIFICATIONS

Microsoft Windows™ Operating System Requirements

- Windows 7 and Windows 7 Pro (32- and 64-bit editions)
- Windows 8
- Windows 10 (32- and 64-bit editions)
- Minimum hardware requirements specified for your Windows operating system are sufficient to run the Axia IP-Audio Driver.

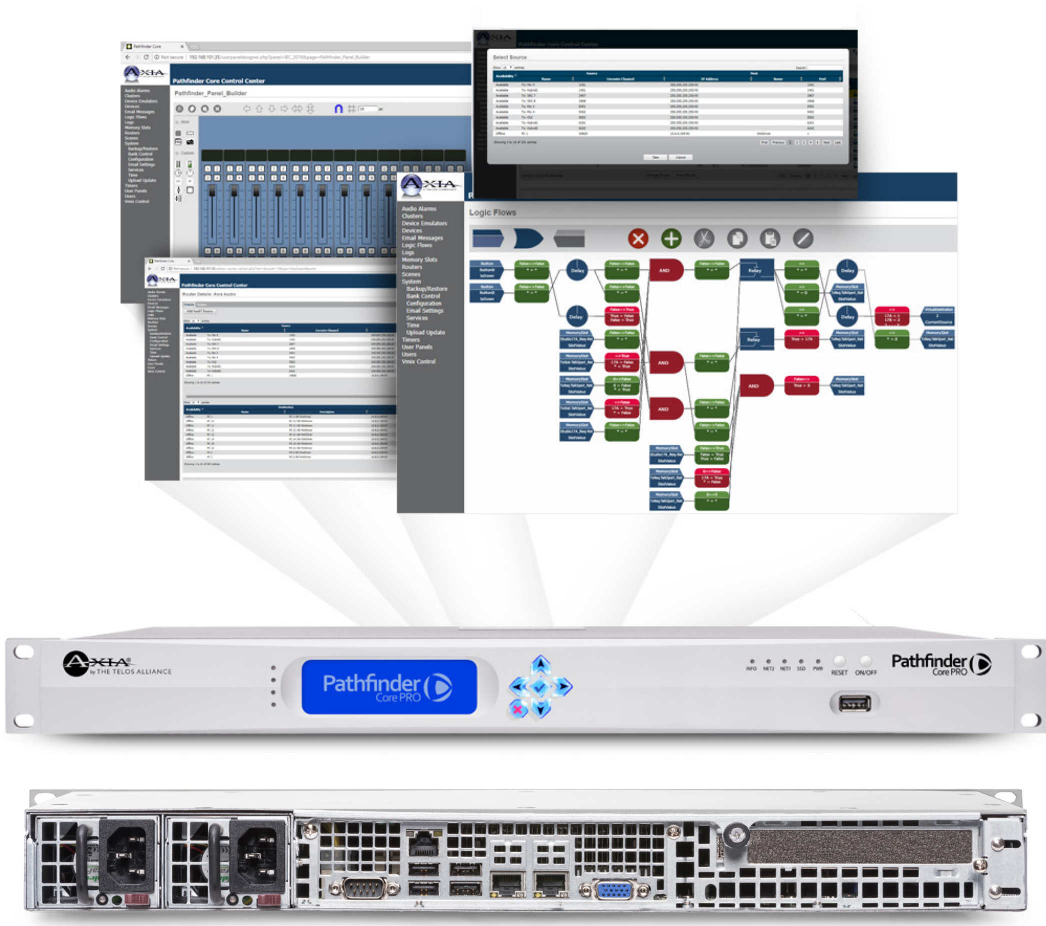
Linux Operating System Requirements

- The Axia IP-Audio Driver for Linux is sold exclusively through Paravel Systems. Please contact them at ParavelSystems.com/contact-us/

* One of our goals at the Telos Alliance is to further the adoption of AES67, the standard for Audio over IP designed to allow interoperability between various IP-based audio networking systems. And while these two words may sound the same, the differences between AES67 *compliance* and *compatibility* is huge. AES67, like all standards, can be minimally implemented. And when standards are minimally implemented, they minimally get the job done. Simply put, compliance means that every single aspect of the AES67 standard—like Unicast mode, for example—is met. Livewire+ AES67 is fully AES67-compliant.

Axia® Pathfinder Core™ PRO

Broadcast Controller & VM



OVERVIEW

See More. Do More. Control More.

With the third generation of Axia Pathfinder, the Pathfinder Core PRO Broadcast Controller and Pathfinder Core PRO VM, routing control has never been this simple...or this sophisticated.

Pathfinder Core PRO—either Appliance or VM—is a toolbox with powerful features that create efficient workflows and facility management. For one, it unifies your Axia device monitoring and control, letting you make system-wide changes from one central place—the Pathfinder Core PRO web interface. Pathfinder Core PRO simplifies routing in complex facilities, even those with thousands of audio sources and destinations, by giving the user easier, more intuitive control over audio workflows. This third generation of Pathfinder routing control lets you customize and command your Axia network with streamlined functionality, including Logic Flows—a flow-chart-style events system that makes events easier to create, adjust, and monitor in real-time. An efficient, intuitive web interface means easy configuration and monitoring from any device, and customizable user interfaces provide streamlined control over your entire Telos Alliance system. Finally, this Linux-based appliance or VM makes full use of the processor while freeing you from a Windows-based server.

Pathfinder Core PRO Broadcast Controller

Pathfinder Core PRO is a powerful and reliable 1RU appliance. The Pathfinder Core PRO Broadcast Controller is proven in years of on-air service. It brings enterprise server-class redundancy and multi-processor, multi-threading architecture to real-time broadcast operations. Faster response, deeper event logging, and ultra-reliable monitoring and control are available in the Pathfinder Core PRO Router Controller. Access to the Pathfinder Core PRO is browser-based, so it's PC platform-independent.

Pathfinder Core PRO VM

Broadcasters worldwide are migrating key infrastructure and processes to virtualized platforms. Whether in company data centers or third-party server farms, broadcasters in radio, television, and other digital media are finding significant benefits in equipment and process virtualization.

Axia Pathfinder Core PRO VM is the virtualized monitoring and control solution for Axia Livewire+™ AES67 facilities. Many IT departments have standardized on one or two server hardware platforms, and perhaps a single hypervisor solution. This standardization is good business practice with benefits for operations, maintenance, upgrades, and redundancy.

Now, Pathfinder Core PRO VM allows broadcasters to deploy Pathfinder monitoring and control on familiar, approved hardware. Moreover, thanks to Pathfinder's network-optimized monitoring and control protocols, Pathfinder Core PRO VM can be deployed locally or at an off-site data center. Indeed, just one instance of this VM-installable solution can monitor and control dozens of Axia broadcast facilities—over a small region or around the globe.

Today's broadcast IT and operations teams appreciate standardization, functionality, and redundancy. Pathfinder Core PRO VM may be installed in a selection of hypervisor environments, including VMware, Hyper-V, VirtualBox, Proxmox, and Stratus.

Thanks to installation and deployment options, the VM version of Pathfinder Core PRO lends itself to centralized deployment, allowing monitoring and control of routing and Axia equipment for a network of studios and remote Axia locations.

Pathfinder Core PRO VML

Joining the Axia Pathfinder Core PRO family is Pathfinder Core Pro VML. The VML delivers all the same features as VM, but has a lower starting price point as it comes with 300 connections (as opposed to VM's 1000 connections), making it perfect for a smaller system with just a studio or two. If the workflow expands, no problem. Simply apply add-on license points to the server and expand system control.

FEATURES (Broadcast Controller & VM)

- Reliable, redundant, system-wide routing control
- Use any modern browser on any platform to configure, monitor, and control
- Linux-based with a web user interface
- Provides route control and customized logic events
- Graphical interface with real-time state reporting
- Graphical logic gates for creation of complex logic flows
- Control protocol for third-party integration, including Device Emulators
- Automatic router table generation
- Hardware version offers dual Gigabit Ethernet ports and dual-redundant power supplies
- Virtual routing for customized views of key audio flows
- Customizable user panels
- Includes 1000 points for use as crosspoint or Logic Flow endpoints
- Add-on licenses for additional 500 points for use as crosspoints or Logic Flow endpoints available
- Add-on licenses for additional 100 points for use as crosspoints or Logic Flow endpoints also available
- VML comes with 300 points for use as crosspoints or Logic Flow endpoints

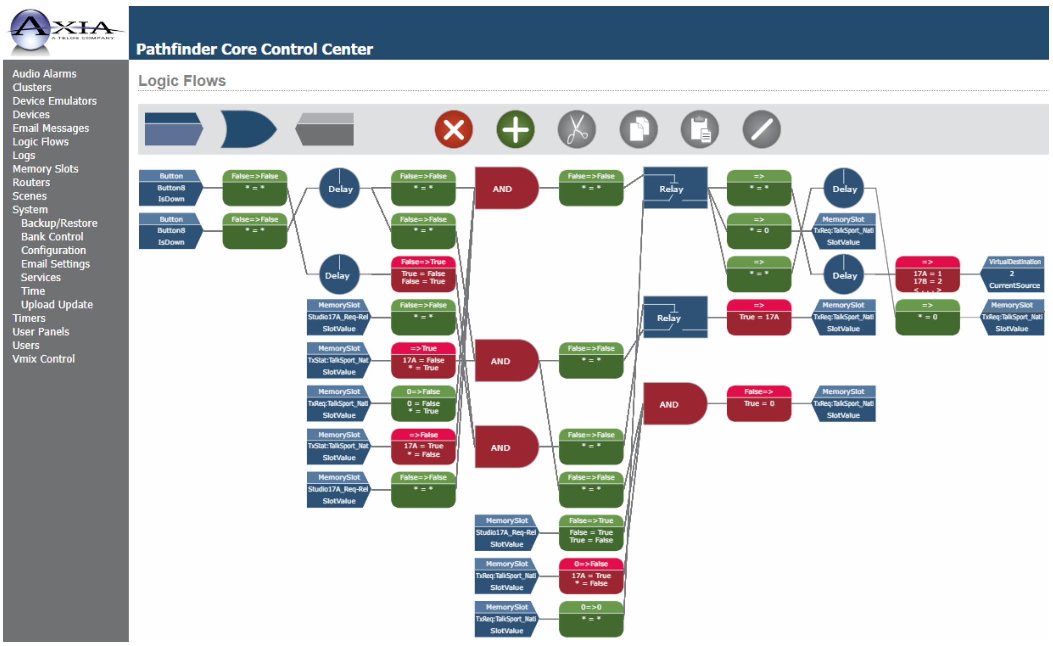
IN DEPTH

PC-Free

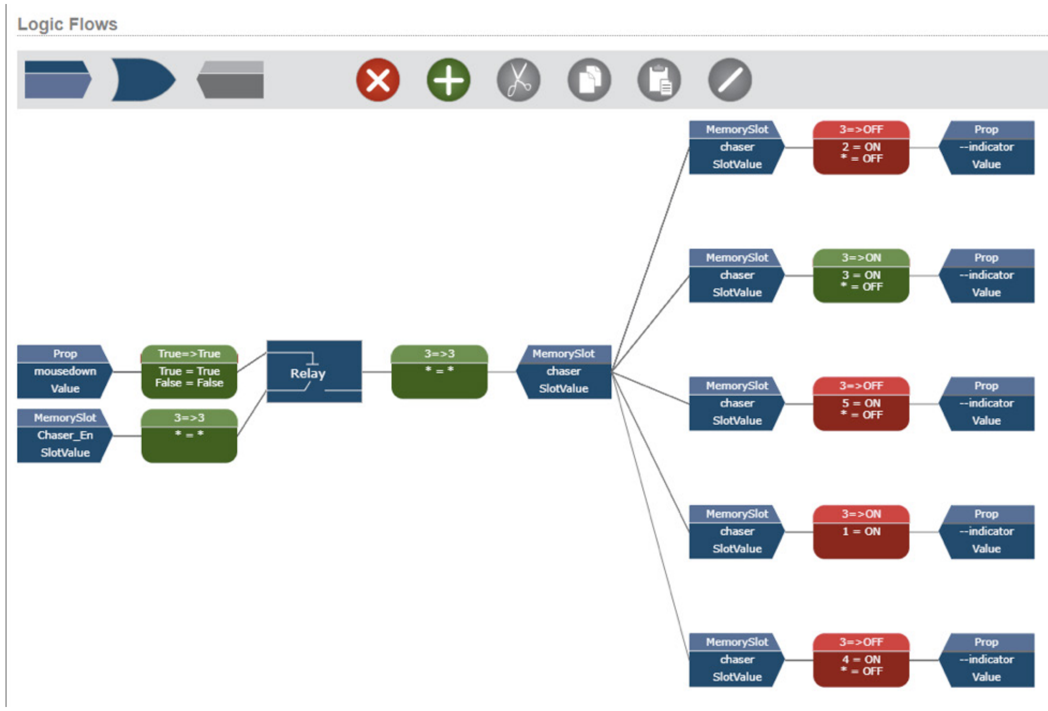
The Axia Pathfinder Core PRO is ideal for stations that want reliable, redundant, system-wide routing control that is purpose-built. The Linux-based network-attached appliance for routing control includes a web user interface to provide route control and customize logic events. Because it uses a web interface, Pathfinder Core PRO becomes platform-independent; users can use almost any personal computing device to interface with the appliance for reliable, 24/7 unattended operation. Customizable function panels are based on HTML5 and can be accessed from almost any computing device with access to the network.

Logic Flows

Pathfinder Core PRO features Logic Flows, an all-new events system that uses logic gates for the creation of complex logic and dramatically simplifies route changes. The Logic Flows event system features a graphical flow-chart style that allows you to see the connections and manipulate them easier, as well as see state changes in real-time via the web interface. For example, if an endpoint changes, you will see those changes live via color and/or textual change on the Logic Flow graphic. Logic Flows are intelligent enough to sense the connections between different flows and present those connections graphically with a much larger array of properties that are available for use. Finally, we have added the ability to clone entire groups of object settings with one simple flow, which becomes invaluable when creating redundant logical paths. These are dramatic improvements over previous versions.



Examples of PDM Logic Flows



Example of airchain switcher

Device Emulators Control Protocol for Third-Party Integration

Because there are a variety of automation systems that don't understand the Axia language, Pathfinder Core PRO Device Emulators allow the unit to "look like" another protocol, emulating third-party protocols and translating them into the Axia language. When Pathfinder Core PRO receives commands in a non-Axia protocol language, it then makes the requested changes to the Axia equipment, emulating a different kind of device by speaking that language. Additionally, the Device Emulators allow the user to define their commands to send and receive, so Pathfinder Core PRO can be used to control and receive commands from third-party equipment in that way as well. The user simply has to define the textual commands to watch for or send in the Device Emulator.

Easy Setup

Pathfinder Core PRO is fast, efficient, and simple to use: Just attach to your network, give it an IP address, and it automatically detects your Axia audio sources/destinations and GPIO ports. Pathfinder Core PRO searches for and finds all of the Axia equipment on the network, then automatically creates both an audio router and a GPIO router with all of the audio or GPIO sources and destinations that are discovered within that equipment, simplifying the initial configuration.

Pathfinder Core PRO also gives you peace of mind by providing distributed redundancy within your network. Multiple Pathfinder Core PRO units can be "clustered" for automatic redundant backup, and each fan-free unit has field-replaceable dual-redundant power supplies as well. As a dedicated hardware appliance, Pathfinder Core PRO gives you freedom from concerns about software compatibility, automatic OS patches, and computer hardware limitations.

Customizable User Panels

In addition to advanced routing control, Pathfinder Core PRO lets users create their user panels, giving them a unique and customized look and design for features like meters and buttons. This powerful drag-and-drop tool integrates routing, logic, and device control to create a user interface that intuitively matches your facility's unique workflow.

Virtual Routing

Aside from routing physical inputs and outputs, Pathfinder Core PRO lets you create virtual routers. For example, if a specific room in a facility doesn't need to see the entire network of inputs and outputs, but only the ones relevant to that room, then a Pathfinder administrator can create a virtual router that only includes the sources and destinations required by that room. In addition, virtual routers can be used to marry multiple sources and destinations together into routable packages. For example, you can create a virtual source that has both an actual audio source and an actual GPIO source as part of its package. Then when you create a route with the virtual router, the audio and GPIO get routed together with one route change. This capability extends even further by allowing any property known to the system to become a source or destination in a router.

Memory Slots

Memory Slots allow you to store—and make changes based on—custom pieces of information. Memory Slots have been expanded in Pathfinder Core PRO to provide easy latching capability of any property in the system, to follow the state of any property in the system, and to build custom text by combining values from multiple slots together. This makes the use of Memory Slots more powerful than ever.

Scene Changes

Scene changes are traditionally used for making a group of route changes at once. Pathfinder Core PRO extends this functionality by allowing any change over any device Pathfinder knows about to be included in a scene. Need to route a studio to air while turning on certain console channels, dialing a Z/IP ONE codec, and sending a start message to your automation system all at the same time? A scene can do that for you.

Crosspoints

With Pathfinder Core PRO, you get 1000 points used towards sources and endpoint change blocks (within Logic Flows). Need more? You can simply buy an extension license to add more logic or more crosspoints.

Note: There are a variety of crosspoints that don't count against the license, including GPIO and virtual-only ones. Only Axia audio sources whose streams are enabled count against the point license. Also, logic flows built within the panel designer do not count against the logic licenses.

FAQs

Is Pathfinder required to run my Axia network?

No, Axia networks are self-contained routing and mixing systems that don't require any external control. However, if you want to automate your routing switcher, with preset scene changes, conditional routing changes, or scheduled route changes, Pathfinder will satisfy your needs.

Why would I need routing automation?

Pathfinder Core PRO lets you consolidate control of your network operations, bringing all of your equipment together under one simple interface to create the most flexible workflows imaginable. It takes all of your Axia nodes and equipment and presents them as if they were a traditional single router, so you don't need to jump from place to place to see and manipulate your facility's routing infrastructure.

Can I trigger routing changes from studio consoles?

Yes. There are a variety of drop-in modules for our popular Axia Fusion consoles that you can use to change single routes or execute predefined salvos based on time-of-day or GPIO. Rack-mount panels make it convenient to map Pathfinder routing commands to hardware buttons located elsewhere, too—like your TOC, engineering office, or communications room.

Can Pathfinder sense dead air?

Yes, there's a Silence Detect function, including Silence, Audio Presence, and Clipping on Axia sources and destinations. With it, you can pick any audio stream in your network—say, your Program-1 output—and monitor it for signal loss. Once your "silence" condition is met, Pathfinder can take action by switching to a different audio input, flashing an alert to talent, or sending you an e-mail.

Can Pathfinder react to a system command? I'm thinking about EAS activation...

Sure. You can define Logic Flows, in which Pathfinder Core PRO monitors GPI channels for external commands and takes action when predefined conditions are met. So you could use a logic flow to watch the GPIO output of your EAS decoder, and switch your main program output, along with your HD channels, to be fed by the output of your EAS gear until the GPI is released—upon which audio inputs are returned to their normal sources.

Additionally, Pathfinder Core PRO provides generic Device Emulators that allow you to define custom commands to send and watch for on a tcp port. So your automation system can send any command you desire, and Pathfinder Core PRO can detect that command, act upon it with any action you see fit via a Logic Flow and reply with another custom command.

Finally, Pathfinder Core PRO provides an advanced control protocol that exposes everything you might want an outside system to control.

Does Pathfinder Core PRO support clustering, like PathfinderPRO software?

Yes. You can attach two Pathfinder Core PRO devices to your network for complete routing automation redundancy.

Does Pathfinder Core PRO support SNMP, AES70, or other protocols?

Pathfinder Core PRO supports Axia and other protocols to allow you to create highly customized workflows. As the industry is learning the importance of common tools, open protocols are becoming hot topics. Pathfinder Core PRO was designed with an open architecture to easily incorporate other protocols. Pathfinder Core PRO includes Device Emulators (protocol translators) with multi-protocol support that can be used to send and receive *any* data.

SPECIFICATIONS

Power Supply AC Input

- 100VAC to 240VAC, 50Hz to 60Hz, IEC receptacle, internal fuse
- Power consumption: 100 Watts, auto-ranging

Operating Temperatures

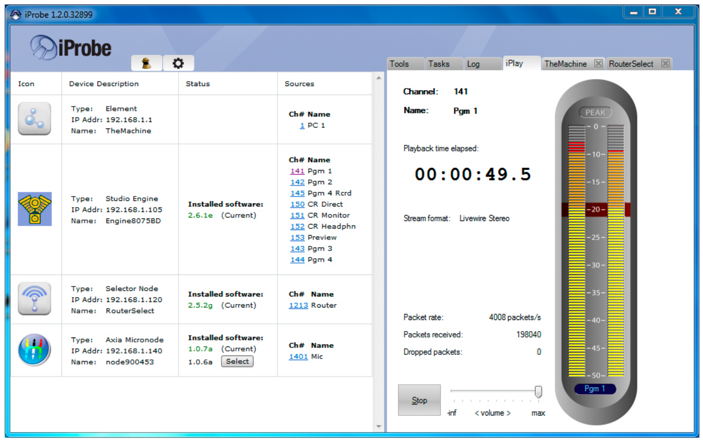
- 0 degrees C to +50 degrees C, <90% humidity, no condensation

Dimensions

- W19.00in (48.0 cm), H 1.75 in (4.40 cm), D 15.50 in (39.4 cm)
- 1RU

Axia® iProbe

IP-Audio Network Management Software



OVERVIEW

iProbe software helps with management, updating, and remote control of any Axia system. It has a powerful auto-documentation feature that generates configuration docs for every device, an Organizer that allows grouping networked audio devices into logical groups for easy management, facilitates uploading software to single or multiple devices, makes device configuration backups and more.

FEATURES

- Discovery: the ability to scan your Axia Livewire® network for Control Surfaces, Nodes (AES, Analog, GPIO), and Mixing Engines, as well as any Livewire devices from Axia Hardware Partners.
- Displays current firmware versions running on the all connected devices and gives you the ability to update firmware remotely, one device at a time or in logical groups of similar devices.
- Displays all the devices in your Livewire system and allows you to browse directly to a selected device's Web-based remote control interface. There is no need to type the device IP address into your browser.
- Complete configuration backup capabilities of individual devices, or all devices within your Livewire system.
- Integrated Syslog server for automated event logging.
- Auto-Documentation feature exports complete system data to a format of your choice for secure backup.
- Built-in iPlay module allows you to listen instantly to any channel on your network, and verify the levels of a given source.

IN DEPTH

Easy Network-Wide Backup, Update and Documentation

Axia iProbe is an intelligent network maintenance and diagnostics suite that consolidates managing, updating, and remotely-controlling your Axia system into one easy-to-use software application.

Axia networked audio devices are managed using a standard Web browser to view, configure, and administer each device. iProbe helps simplify this process by scanning and collecting all the information and presenting it a graphical interface. Along with this convenient central point of control, iProbe gives you powerful system tools like the Organizer, which performs advanced tasks such as gathering Livewire-enabled devices into logical groups for easy management and single-point administration of group settings. iProbe also helps with software version control, making it simple to upload software to single or multiple devices, back up device configuration, and more.

There's a powerful Auto-Documentation feature that queries and documents configuration settings for every networked Axia device — essential for administering large networks. Auto-Doc gives you the ability to export your Axia system data into an HTML format or text for printing a hard copy of your system configuration, or constructing a web page for future reference. You can also export to a tab-delimited text format for importing into other documents or spreadsheets. Exporting in XML format to other applications is also available.

And of course, there's a System Backup / Restore function that generates full-system backups which can be used to restore your Axia network from bare metal, should the need arise.

Even with all this power, iProbe is simple to use. An intuitive graphical interface lets you browse a list of similar devices, or click one-at-a-time on individual devices to make inspections or adjustments. You can even listen to individual sources with the integrated iPlay functions, and check on audio levels of any audio source.

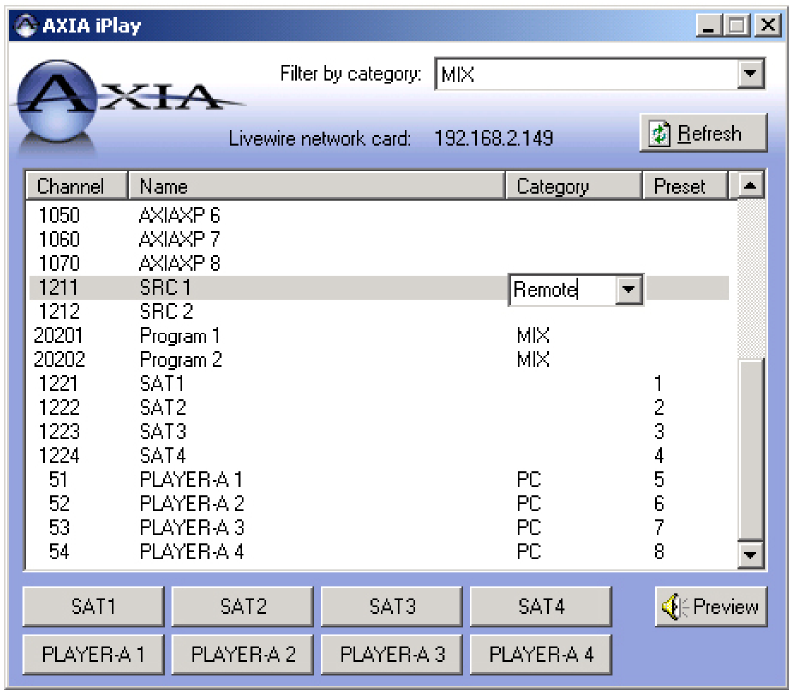
SPECIFICATIONS

System Requirements

- Windows 2000 Professional, Windows XP Professional, Windows Vista (32- and 64-bit editions), Windows 7 and Windows 7 Pro (32- and 64-bit editions), or Windows 8 operating system.
- 100BASE-T or higher wired network adapter.
- Internet access to enable device firmware downloads.

Axia® iPlay

Network Stream Player for Windows



OVERVIEW

Software-based IP-Audio monitoring program lets Windows PC users select and listen to any audio source available to their Axia network. Choose from a complete list of available streams; eight user-programmable preset buttons provide quick access to frequently-accessed channels. On-screen level display meters auditioned audio.

FEATURES

- Allows listening to any Axia network audio stream using standard PC sound card/speaker combo or headphones.
- Automatic detection of Livewire® audio sources from connected PCs.
- User friendly interface allows filtering and sorting Livewire channels, making it easy to navigate in big systems containing hundreds or thousands of Livewire channels.
- Preset buttons that allow quick access to eight pre-selected channels.
- Administrator can restrict access to a set list of audio channels using built-in Access Control Lists.

IN DEPTH

Turn Any PC Into a Listening Station.

Remember the days when giving your Sales Manager a listening station meant running cable through the ceiling, installing a selector panel in the office wall, and mounting speakers in the drop tiles? And then, he could only hear a limited number of the audio channels your plant produced.

Axia iPlay PC software does away with old-fashioned speaker wire and rotary selectors. It allows any Windows PC to listen to streamed audio directly from your Axia network — any streamed audio. Not just Program feeds, but satellite downlinks, remote hosts, news production studios, interview rooms, etc. iPlay lets you give PC monitoring capabilities to PDs, GMs and sales staff using their existing computers, with no special wiring required. Just connect their PC's NIC to your Axia network, install iPlay, and presto! Every PC is a listening station. There's even an on-screen level display that meters the audio you're listening to — great for use as confidence meters for PDs or production personnel.

Users can choose from a list of all available audio, but in big plants that can be a lot to sift through. Not to worry: they can pre-set their favorite channels on the eight user-programmable preset buttons to get quick access to the streams they listen to most. And of course, you can filter out raw mic channels or other selected audio streams to prevent unauthorized listening.

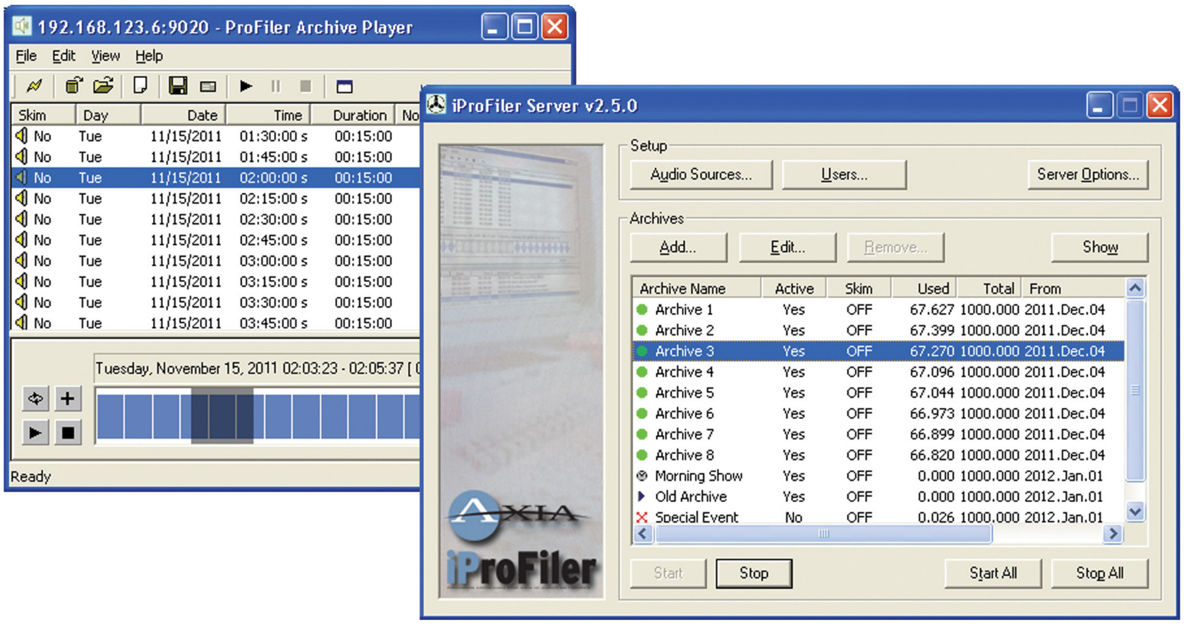
SPECIFICATIONS

System Requirements

- Windows 2000 Professional, Windows XP Professional, Windows Vista (32- and 64-bit editions), Windows 7 and Windows 7 Pro (32- and 64-bit editions), or Windows 8 operating system.
- 100BASE-T or higher wired network adapter.

Axia® iProFiler

Automated Program Archiving



OVERVIEW

Axia's popular iProFiler logging software lets you simultaneously capture up to 24 stereo audio channels to time-stamped MP3 audio logs directly from your Axia IP-Audio network — no audio cards required. Included software records, manages and plays back archived audio files. Recording software runs under Windows XP and later; playback software runs under Windows NT, Windows 2000, Windows 98 or Windows XP and later. Record mode can be set for logging, skimming, or combination of both. Logged audio may be auditioned remotely via LAN, WAN, or Internet.

FEATURES

- Simultaneously captures up to 24 channels of stereo audio.
- Directly records Axia digital audio streams — no sound card needed.
- Archived audio can be auditioned remotely via LAN, WAN or the Internet.
- iProFiler Live Player streams audio over any IP connection as it's being encoded. Great for consultants or group PDs listening remotely.
- NTP Time Sync synchronizes log file timestamps with your house NTP server (if equipped).
- Choose your skimming mode: Logging (continuous archival storage of program material), Skimming (records only when talent mic is open), or SmartSkimming (low-bitrate logging switches to a user-specified higher bitrate for quality captures when talent is on-mic).
- No "spool-up" time: iProFiler buffers incoming audio so that you never lose a word - no matter how late talent opens the mic.
- Recorded audio is time-stamped and stored in easy-to-search 15 minute blocks for fast retrieval.
- Standard MP3 file format allows logged audio to be played back on any media player application. Play files in iProFiler Archive Player to view detailed time-of-day data and user annotations.
- Easily select & export audio segments to WAV files for external editing.
- Choose any standard MP3 bit rate - from 16kbps - 320kbps - for the quality/drive space ratio that best suits your needs.
- Encoded program segments can also be set to upload automatically to an external drive, network share or FTP site.
- Remote monitoring application lets you "check up" on iProFiler remotely using a LAN or Internet connection; monitors disk space & audio presence.

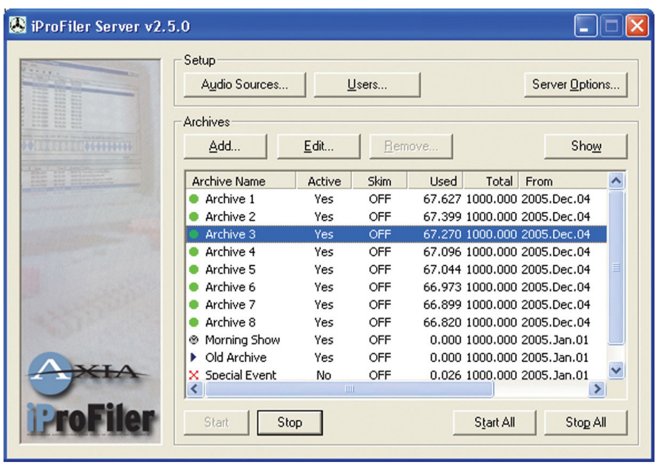
IN DEPTH

Multi-channel, Multi-stream audio archiving for Axia Audio networks.

Sooner or later, someone's going to ask for a hard copy of a specific broadcast. Whether it's a client looking for proof of play, a Group PD that wants airchecks, or a listener claiming your morning show did something naughty, you'll need a record of your broadcast programming.

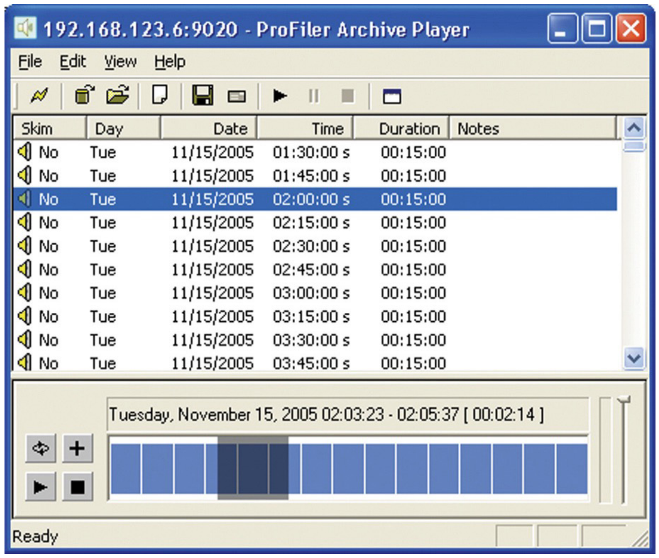
Be prepared with Axia iProFiler, the award-winning audio archiving software that integrates with Axia IP-Audio networks to capture up to 24 simultaneous channels of Livewire® Standard-stream stereo audio without sound cards. iProFiler uses the Axia IP-Audio Driver to exchange audio directly with Axia networks; just install iProFiler on a PC, connect the computer's NIC to the network with CAT-5, select the program streams you want to capture. It's as simple as that.

iProFiler's networked connectivity makes it the easiest logger to set up and operate, bar none. Just browse the audio streams available on your Axia network, select the ones you want to record, choose a bit rate for storage and off it goes.



And iProFiler is extremely flexible; you can continuously log program audio, automatically record telescoped talent airchecks, or record only what's broadcast when the mic isn't open. And iProFiler has "listen line" capability that lets you hear audio over your network (or the Internet) as it's being encoded - perfect for group PDs or consultants.

iProFiler's stored audio is networked, too. Any workstation or computer connected to your IP-Audio network can find and listen to time-stamped audio using a simple web-browser interface.



iProFiler gives you a choice of operating modes for each archived audio stream:

- Choose “Logging” for continuous archival storage of program material, indefinitely (dependent upon storage space) or on a timed-record basis.
- Choose “Skimming” to record audio only when talent’s mic is open, to capture live shows, call-in segments, talk shows or DJ bits. Program audio is pre-buffered so that there are no “up-cuts” upon record activation.
- Choose “SmartSkim” for a unique combination of skimming and logging. When talent mics are closed, ProFiler records audio in a low-bit rate logging mode, then switches to a higher bit rate for quality captures when talent is on-mic. All bit rates are user-selectable.

iProFiler is ideal for stations required by law to log program content, and since you can also listen to “live” audio over IP as it’s being logged, it’s great for Production Directors and morning show producers, program consultants or group PDs. Perfect for competitive monitoring, too — log other stations along with your own to fine-tune your formatics. An integrated audio browser lets your production crew tag segments and export them as WAV files for further editing, and logged shows can be automatically uploaded to FTP servers for storage or distribution.

SPECIFICATIONS

PC Hardware Minimum Requirements

- Pentium-IV, 2.4GHz processor or better with 512Mb RAM, 300 Gb free hard drive space, 100BASE-T NIC.

Operating System

- iProFiler Server: requires Windows XP or later. WAN/Internet connection required for remote monitoring.
- iProFiler Client: Requires Windows XP, Windows Vista, or Windows 7 or later.

Operating modes:

- Logging (continuous archival storage of program material)
- Skimming (records only when talent mic is open)
- SmartSkimming (low-bitrate logging switches to a user-specified higher bitrate for quality captures when talent is on-mic)
- Scheduled recording (date and time + length of program)

Audio Interface

- 100BASE-T or better Ethernet NIC with connection to Axia IP-Audio Network.
- Supports up to 24 stereo streams simultaneously.

Audio Specifications

- Storage Format: MP3.
- Comproession Algorithm: Genuine Fraunhofer IIS
- Bit Rates Available: 8 kbps to 320 kbps, in standard increments
- Pre-roll and Post-roll Skim delay: up to 10 seconds, user-definable

Axia® IP Intercom

Go Ahead: Talk Amongst Yourselves.

OVERVIEW

Axia® IP Intercom is the only broadcast intercom system that takes advantage of the ease and efficiency of proven IP-Audio technology. Using a standard Ethernet backbone, IP Intercom saves on cost, space, and installation time, and eliminates special plug-in cards altogether.

The advantages of IP and Ethernet – low cost, easy installation and maintenance, efficient infrastructure – are well known. Installing IP Intercom is as simple as clicking together Ethernet gear! And of course it's easily scalable: plug as many stations into your switch as you want and add on from there. There's no expensive, hard-wired, custom-cable multi-pair infrastructure to deal with.

If you don't have an Axia studio network, IP Intercom can still help you save money, increase efficiency, and decrease the hard-wired infrastructure hassle It's a stand-alone system with I/O that will accommodate multiple mixing consoles. But if you do have an Axia system, you'll get seamless console integration that gives your operators benefits other systems can't, like the ability to take broadcast quality intercom audio directly to air, and feed IFB audio directly to intercom callers.

The IP Intercom system includes a variety of desktop and rackmount stations, a software Intercom application that turns any PC into an intercom station, and drop-in modules for popular Axia mixing consoles.

FEATURES

- 100% digital system, end to end.
- Seamless integration between broadcast audio and communications channels. Full 20 Hz – 20 kHz audio response allows intercom channels to be taken to air with no degradation of sound quality
- Stand-alone rack-mount, desktop and integrated Axia console modules are available for a turnkey intercom installation.
- Program station presets and GPIO functions using any standard Web browser.
- Ethernet-based system has no central matrix or card-cage; is naturally scalable. Easily expand the number of intercom stations as your facility grows by simply plugging in new stations.
- Intercom keypad can also dial outside phone lines (using an optional telephone hybrid).
- Analog I/O presented on both XLR and StudioHub-compatible RJ-45 connectors.
- Front-panel locking connections accommodate popular mini-mics and headsets.
- Add PCs to the system with SoftCom Intercom Station for Windows.

IN DEPTH

Imagine a digital intercom system with no central matrix.

Actually, don't bother — we've built one. Axia IP Intercom saves on cost, space, and installation time, and eliminates special plug-in cards altogether. It's real plug and play that works every time — even when you need to add a station, or reconfigure the ones you've got.

Everybody knows the advantages of IP and Ethernet – low cost, easy installation and maintenance, efficient infrastructure. Thanks to its efficient Ethernet backbone, installing IP Intercom is a simple single-click connection. Of course it's easily scalable: plug as many stations into your switch as you want and add on from there. Then start talking! And if you move to a new location, you can just pick up the gear and take it with you — there's no expensive, hard-wired, custom-cable multi-pair infrastructure mess to deal with.

Don't have an Axia studio network? That's OK. You'll still save money and increase efficiency by choosing IP-Intercom; it's a stand-alone system with I/O that will accommodate multiple consoles. But if you do have an Axia system, you'll get seamless console integration that gives your operators benefits other systems can't. For instance, you can take broadcast-quality intercom audio directly to air. And you can feed IFB audio directly to intercom callers.

IP Intercom gives you unlimited full-bandwidth access to any studio, news or sports venue, office, hallway, broom closet or wherever. Talk and listen to individuals or groups hands-free, with no echo or feedback — IP Intercom features exclusive AEC advanced echo cancellation from Fraunhofer Labs (the inventors of MP3), so there's never any open-mic feedback during conversations. Ever.

IP Intercom system is completely digital. Other intercom systems try to make you think they're digital by piping their analog signals over CAT-5 cables, but the last thing you need during a breaking story or transmitter failure is hum and buzz getting between you and the guy you need to talk to. With IP Intercom, there isn't any.

So you've gotta be a genius to use it, right? Actually, anyone with an index finger can operate this system with ease. The web interface makes setup simple. Sharp, high-contrast OLED displays are easy to read from anywhere in the room. And our clever callback feature makes sure you'll never miss a call, no matter what you're doing. There are also functions that allow talent to mute calls from other stations, to make sure there's never an interruption on-air.

IP Intercom comes in several rack-mount and desktop styles, plus drop-in modules for Axia Fusion™ consoles. And our unique SoftCom software lets you turn any connected PC into an intercom station! Just mix and match to build a system customized to your needs.

Rackmount Stations

IC.20 Rackmount Station

The IC.20 intercom panel features 20 station presets for quick contact with frequently-called stations. Perfect for Master Control or TOC, the IC.20 includes a keypad and associated display for fast access to stations system-wide, plus group talk and auto-answer functions. Keypad can also dial outside phone lines (using an optional telephone hybrid). 2RU rackmount package features high-visibility 10-character OLED (organic LED) displays, built-in speaker, front- and rear-panel mic connections, 4-pin locking headset jack, analog I/O presented on both XLR and StudioHub-compatible RJ-45 connectors, GPIO connection for speaker mute/dim and external line-status tallies, and an Ethernet jack for single-cable network connection.

IC.10 10-Station Intercom Panel

The IC.10 is a 10-station version of the IC.20 we talked about earlier. It has 10 station presets with high-resolution OLED displays, a built-in speaker, front- and rear-panel mic connections, 4-pin locking headset jack, analog I/O on XLR and StudioHub-compatible RJ-45 connectors, GPIO connection for speaker mute/dim and external line-status tallies, and an Ethernet connection.

IC.1 10-Station Intercom Panel

The IC.1 is a cost-effective way to add intercom capabilities to any studio. It features 10 LED-backlit film-cap buttons that are easily labeled with station names; like other IP Intercom station, programming is via Web interface. IC.1 has a built-in speaker and front-panel 4-pin locking headset jack, front- and rear-panel mic inputs, analog I/O with XLR and RJ-45 connectors, GPIO speaker mute/ dim control. An Ethernet jack completes the connection complement.

Desktop Stations

IC.1D 20-Station Filmcap Intercom Panel

The IC.1D 20-station desktop intercom is perfect for producers, screeners, etc. IC.1D has 20 preset stations presented on LED-backlit button caps; an economical way to add intercom function to any space. 20 LED-backlit film-cap buttons can be labeled with station names and programmed using a built-in Web interface and any browser. The OLED callback window lets users identify and answer calls from remote stations that aren't programmed on a local "speed" key. IC.1D includes a built-in speaker and front-panel 4-pin locking headset jack. All it takes to add it to your intercom network is a single CAT-5 connected to the rear-panel Ethernet port; a built-in auto-sensing power supply eliminates nasty "wall warts."

IC.20D 20-Station OLED Intercom Pane

The IC.20D is the desktop version of the IC.20 rack-mount station we showed you earlier. The 20 station preset locations are equipped with high-resolution OLED displays; the OLED callback window and dialing pad let operators call any station not programmed to a preset location. Naturally there's a built-in speaker, front-panel 4-pin locking headset jack, front-panel mic input, an Ethernet port for fast hookup, and internal auto-sensing power supply.

Console Modules

You don't need to own an Axia console to use IP Intercom — rack-mount and desktop stations integrate with any broadcast mixer to route intercom traffic to air instantly — full-bandwidth, broadcast-quality audio, not tin-can-and-string noise. But if you do own an Axia Fusion mixing console, these drop-in modules make communications even easier by turning your board into an intercom station!

Built-in Talkback functions enable seamless communication between board ops, hosts and studio guests.

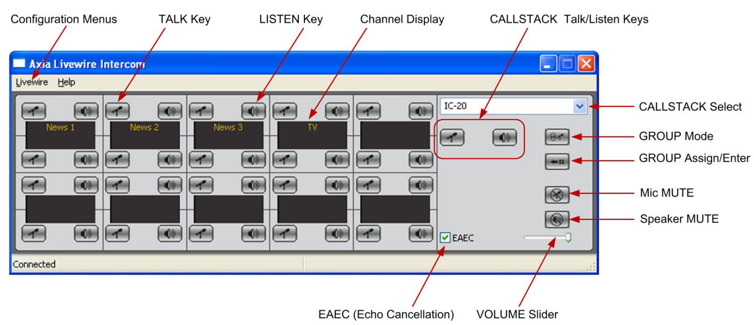
20 Station OLED Intercom Module

The 20-Station OLED intercom module requires two frame positions and provides access to 20 pre-programmed intercom stations. Individual talk and listen buttons are combined with high-resolution OLED displays for fast access to frequently-called stations; auto-answer functions are also provided. Mic audio is taken directly from the console operator's microphone; speaker audio is directed to the console's preview speaker. There's a dedicated listen volume control, individual mic and speaker mute keys and group talk functions; the overbridge display works with the console's monitor module numeric keypad to give direct access to any station systemwide. Station presets and GPIO functions are programmed using any standard Web browser. Available for Fusion consoles.

10 Station Film-Cap Intercom Module

This economical 10-Station Film-Cap intercom module features ten LED-backlit film-cap buttons for single-button calling of up to 10 preset stations. This module occupies one frame position, and also provides a dedicated listen volume control, speaker and mic mute buttons. It uses a single frame position. Available for Fusion consoles.

SoftCom IP Intercom for Windows



Axia Softcom Intercom for Windows makes any networked PC a part of your IP Intercom system! The easy user interface mimics the IC-20 control panel, with preset locations for 20 frequently-called stations. Auto-answer and hands-free functions are supported, and a drop-down station finder gives instant access to stations not pre-programmed. All your PC needs is a sound card with mic & speakers, and a 100BASE-T Ethernet connection to your Axia IP-Audio network. Purchase includes a site license for all PCs.

SPECIFICATIONS

Like all Axia products, IP Intercom uses only premium, studio-grade audio components to guarantee maximum performance.

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
- Output Level: +4 dBu, nominal

Analog Line Inputs

- Input Impedance: 20 k Ohms
- Nominal Level 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

Frequency Response

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

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: 125 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 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 Input CMRR: >60 dB, 20 Hz to 20 kHz
- Microphone Input CMRR: >55 dB, 20 Hz to 20 kHz

Power Supply AC Input, rackmount and desktop stations

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

Operating Temperatures

- -10 degrees C to +40 degrees C, <90% humidity, no condensation

Dimensions

- IC.20: 3.5 inches x 19 inches x 8.5 inches, 5 pounds
- IC.10X: 1.75 inches x 19 inches x 8.5 inches, 4 pounds
- IC.10: 1.75 inches x 19 inches x 8.5 inches, 4 pounds
- IC.1: 1.75 inches x 19 inches x 8.5 inches, 4 pounds
- IC.20D: 18.25 inches x 6 inches x 5.75 inches, 6 pounds
- IC.1D: 13.5 inches x 8.5 inches x 4.5 inches, 6 pounds

SoftCom PC Hardware Requirements

- Windows XP or higher
- 20MB free hard drive space
- 100BASE-T Ethernet connection to Axia network
- Sound card and mic/earphone headset

25-Seven® Voltair®

Ratings. Confidence.

OVERVIEW

It's all about your listeners. Your people, your programming, your gear… they're all focused on building and retaining your audience.

We don't have to tell you about the direct link between the size and composition of your audience—as measured and reported by your ratings—and your advertising revenue. That's why it's vital to you that every panelist in your market is accurately measured and that every station is playing on a level field.

Introducing Voltair, designed to give you greater confidence that every listener is counted when it counts the most.

Ratings, Programming & Technical Operations

Radio ratings have been called "a game of inches," where winners and losers are sometimes decided by the thinnest of margins. Station management teams have always carefully monitored their markets' listener data and taken action to maximize ratings and revenue.

With changes in rating survey methodologies in recent years, many program directors report making more dramatic changes than ever before. For example, dayparts have been moved, local breaks have been reduced, and programming clocks have become more rigid in response to the hard, quarter-hour boundaries of ratings credit. Likewise, the industry has seen changes in audio processing practices, airchain device order and other station engineering procedures—all in service to optimal performance of watermark-based ratings technologies.

Because ratings performance data is provided on a delayed basis, stations lack the means to conduct real-time analysis of audience response. Programmers have had limited insight into what efforts are of benefit and why. Hence, most station efforts to optimize their performance in ratings have been trial and error, with little insight into what may or may not be effective.

Understanding The Current Ratings Ecosystem

25-Seven® has been following the deployment of watermark-based rating methodologies since they were introduced. We've spoken with many program directors and engineers, getting their perspectives on the overall system architecture and the results of their optimization efforts.

After researching the publicly available data on the technology, our team of broadcast and audio experts uncovered the variables that contribute to watermark integrity. More importantly, we developed Voltair to provide you with the tools you need to monitor and analyze these variables, providing you with data to inform your technical and programming decisions.

What did we find?

- **Audio Content**—The spectral characteristics of your audio content—music, announcer voices, etc.—may negatively impact the robustness of your watermark encoding. Simply put, some audio content encodes well while other content does not.
- **Listener Environment**—A listener's device may not detect particular content because of their current acoustic environment. A song or voice may not, for example, decode as well in a car as it does in a bedroom.

To address these issues, we needed to account for the highly complex set of interactions among the encoding and decoding processes, the audio properties of content, and listeners' acoustic environments. We developed a totally new set of easy-to-use tools for Voltair to analyze and help you manage the consequences of these interactions.

How Voltair Can Benefit You

Operating transparently in your airchain, Voltair:

- Monitors and analyzes the robustness of watermark encoding across all program content.
- Offers visibility into how listening environments may influence watermark decoding, using models of acoustic spaces where listeners are wearing or carrying their devices.
- Includes advanced audio signal processing to enhance the detectability of the watermark codes within the context of your programming objectives.
- Empowers programmers to make informed decisions to address potential weaknesses in either encoding or decoding.

For example:

- You can compensate for changes in program material and listening environments during different dayparts and program types.
- You may choose to balance strong and weak program segments within each quarter-hour time segment to produce successful decoding and get the earned credit for the full segment.

Voltair also serves as an off-line tool to identify produced and live content with low encoding confidence. New programming elements—liners, promotions, etc.—may be created with greater confidence of strong encoding.

Voltair and Your Ratings

Your revenue relies on your ratings. The overriding purpose of Voltair is to increase your confidence that those ratings accurately reflect the habits of your listeners.

As you work to get credit for all the ratings you've earned, keep the following in mind:

- Even one listener device worth of data can make a measurable difference in your ratings.
- Insights into how your audio content is handled during the watermarking process can show you ways to improve the robustness of your encoding.
- Consideration of listening environments can guide changes that may improve the reliability of watermark decoding on listener devices.
- When you have confidence in the end-to-end watermark system performance of your stations' signals, you also have more confidence in the relationship between ratings and your programming decisions.

Voltair won't increase your actual listenership, but it will help you be more confident that listeners to your station participating in the watermark-based ratings process are correctly measured.

Regulatory

North America: FCC and CE tested and compliant, power supply is UL approved.

Europe: Complies with the European Union Directive 2002/95/EC on the restriction of the use of certain hazardous substances in electrical and electronic equipment (RoHS), as amended by Commission Decisions 2005/618/EC, 2005/717/ EC, 2005/747/EC (RoHS Directive), and WEEE.

25-Seven® TVC-15

Broadcast Watermark Analyzer, Monitor, and Adaptive Controller

OVERVIEW

TVC-15 Overview

Broadcasting is a numbers business. Your success depends on what kind of audience you attract and hold. Audience size and composition is measured primarily by reports from private ratings agencies, and for most broadcasters, there's a direct link between those reports and a station's revenue. In electronically measured markets, having good tools—ones that help you understand the entire electronic measurement ecosystem—is essential to your station's competitive picture. With TVC-15, for the first time ever, you can detect, monitor and analyze how well each element in your programming supports watermarking. Measurements happen in real time, right off the air, *without depending on or being connected to a particular encoder.* Every 400 milliseconds, TVC-15's tone verification codec analyzes the actual code symbols in any audio you feed it, whether yours, or your competitors'. It will work from any source, live or recorded. A front panel graph of your station's watermark density gives you a granular, moment-by-moment display; you can also download reports to look at encoding quality over hours, days and weeks.

And for stations with a Voltair watermark monitor and processor, you can use TVC-15 to automatically adjust enhancement levels in real-time. TVC-15's Intelligent Adaptive Enhancement [AE] closes the feedback loop, letting you dynamically control Voltair processing based on moment-by-moment analysis of your actual air signal, pushing enhancement when it is needed, while backing off when not. For more advanced watermark monitoring, TVC-15 lets you See What Counts!

IN DEPTH

Electronic Measurement and Your Ratings

Broadcasting is a numbers business. Your success depends on what kind of audience you attract and hold. Audience size and composition are measured primarily by reports from private ratings agencies, and for most broadcasters, there's a direct link between those reports and a station's revenue. The viability of your station's watermarks is constantly varying, depending on your programming, the panelists' environments, and other variables. Changes can happen as quickly as individual syllables in an announcer's voice, or traffic noises on the highway.

How Watermarking Works

Ratings agencies bases these reports on *listener panels*, where each panelist represents many people in a market. In electronically measured markets such as the top 48 markets in the USA, panelists wear portable devices called meters. These meters register unique digital codes broadcast by each cooperating station. Thousands of these codes can be created in the course of an hour. In theory, whenever a panelist hears a station—on their car or home receivers, in a store or restaurant, or even from a colleague's Internet computer speaker—the meter hears the station's code, and the ratings system registers the listening.

The codes themselves sound something like a fax signal, and aren't pleasant to the ear... so they're deliberately 'masked' under louder sounds in the programming, in a process called *watermarking*. Masking is a psychoacoustic phenomenon that keeps us from hearing certain combinations of sounds, even though electronic meters can still detect them. But there are more than a hundred possible digital code symbols used by the meter-based system, and each requires slightly different characteristics in the masking sound.

A proprietary watermarking encoder provided by the ratings agency sits in your air chain, and looks for masking opportunities where it can embed hidden codes. When it hears a potential mask for a current digital code symbol, it generates the symbol and mixes it with the programming. Unfortunately, masks are evanescent, appearing and disappearing as your content changes... sometimes, many times per second. So the number of codes you can broadcast is also constantly changing, depending on your programming. Some content is a lot better at supporting watermarks than others. Silence doesn't support them at all.

- Got a talk show with a musical introduction? *Chances are the intro will have more encoding opportunities than the talk.*
- Running a sports show or drama? *Scenes with just play-by-play or dialog probably won't be encoded as well as those with crowds or other busy backgrounds.*
- Playing a commercial or promo? *Our research indicates a sung jingle usually encodes better than a dry voice-over... even though the spoken words might be more important to the selling message.*

Furthermore, masking requires the code symbol to be significantly softer than the masking audio. As your content gets softer, the encoding hardware has to make the codes softer. Environmental noise around the listener can interfere with those softer codes, even if your listeners don't mind the noise: Humans are very good at tracking meaningful voice or music in a noisy environment. Meters, unfortunately, aren't as smart: It's possible that a watermark signal, sent by the encoder at levels where it wouldn't be annoying in a quiet environment, doesn't get detected by panelists' meters in the real, noisy world.

Bottom Line

The viability of your station's watermarks is constantly varying, depending on your programming, the panelists' environments, and other variables. Changes can happen as quickly as individual syllables in an announcer's voice, or traffic noises on the highway.

Having good tools—ones that help you understand the entire electronic measurement ecosystem—is essential to your station's competitive picture.

What can be done?

25-Seven put years of research and testing into the technical issues with watermarking, and our groundbreaking Voltair processor works with your station's encoder to enhance watermarking codes as they're being generated. Voltair's enhancement can be varied by the station to accommodate different programming styles, and controlled by station automation for different dayparts.

Many stations have found Voltair effective to help make their electronically derived ratings a better match for the audiences they know they've got, and more reliable during hard-to-encode programming. But to really manage this kind of problem, you have to be able to quantify it.

Both Voltair and hardware provided by ratings agencies include ways to meawsure how encodable a program stream is. Voltair can be particularly helpful, with a minute-by-minute front panel display of code reliability, techniques to reduce randomness when calibrating Enhancement settings against ratings reports, and optional downloadable history reports and Excel graphs of a station's coding activity.

But neither system can give you moment-by-moment measurements of how well each element in your programming supports watermarks.

And neither system takes this information to the next level, *actually adjusting enhancement levels in real-time* to compensate for the wide variety of sounds that keep a radio station interesting.

You need to understand the entire electronic rating system. You need tools that can quickly and precisely measure how it works. And you need efficient ways to apply this knowledge so it can optimize your station's product.

That's why we developed TVC-15.

SPECIFICATIONS

We are constantly working to improve our products. Specifications and features are subject to change without notice

ANALOG LINE INPUTS:

- Input Impedance: >40 k ohms, balanced
- Nominal Input Range: +4 dBu
- Input Headroom: 20 dB above nominal input
- A/D Conversions: 24-bit, Delta-Sigma, 256x oversampling

POWER SUPPLY AC INPUT

- Auto-ranging supply, 100VAC to 240VAC, 50 Hz to 60 Hz
- IEC receptacle, internal fuse, on/o switch
- Power consumption: 55 Wa s

OPERATING TEMPERATURES

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

DIMENSIONS AND WEIGHT

- Chassis Dimensions (ex protrusions): 19" (48.2 cm) wide 3.5" (8.9 cm) height 11.75" (30 cm) depth
- Chassis Weight: 14.5 lbs. (6.57 kg)
- Shipping Dimensions & Weight: Contact customer support

FEATURES & BENEFITS

Every 400 milliseconds — 150 times per minute — TVC-15's tone verification codec analyzes the actual code symbols in any audio you feed it.

- Raw symbol reliability is displayed on a constantly changing bar. The symbols that make up a complete station identification message are then processed through our proprietary algorithms.

A front panel graph of your station's watermark reliability updates every 400 milliseconds.

- That's fast enough to track individual program elements, or style changes in a song, or even the difference between a host and a call-in guest.

A front panel timer updates every time your station broadcasts a complete watermark message.

- It takes 4.8 seconds for the watermark system to assemble enough code symbols for full station identification. Under ideal circumstances1, TVC decodes a complete message every 4.8 seconds. Each time you do, the timer resets and appropriate message details are displayed.
- During periods of low masking (silence, spoken word, some music), the timer doesn't get as many chances to reset. It keeps counting, and changes color to alert you to the condition.

TVC-15 doesn't depend on a particular encoder, and doesn't have to be connected to it.

- You can connect TVC to an air monitor. Or to an Internet radio, a HD receiver, or any other way listeners are getting a signal with watermarking codes. Use any convenient analog source, and get an instant reading of how strong its codes are.
- TVC is switchable between encoding formats: Layer 1 (used for US radio) and Layer 2 (Canada and some other countries).
- You can equalize or distort the signal going to TVC to simulate low-quality radios. Or you can feed TVC from a microphone pointed to any radio or loudspeaker, in a quiet test room or noisy public space2.
- You can bias TVC's measurements using statistical noise simulation. Or you can record actual environmental noise and other possible interference, and mix it with the signal you're feeding TVC.
- You can feed it other stations' signals, to assure code reliability across a broadcast group... or even see how your competition is encoding. All this can happen in the privacy of your own local network, with nobody else able to see how you're making programming decisions.
- TVC's front panel and reports even identify when it sees different encoders, so you can scan multiple signal sources and sort them out later.

TVC-15 will work from any source, real-time or recorded.

- You can feed TVC recordings of your own or other stations' signals, whether they're from your program line, off-air monitors, or recordings from public spaces.
- You can use it offline with a spare encoder, to analyze program segments or production elements. TVC's fast response lets you compare different sub-elements within a program stream.
- You can use it with an automated switcher to cycle among various stations and program streams in your group to verify that encoders are working.
- Operation is completely flexible: Input can be switched between program sources or among different encoders without the need to recalibrate or reboot.

TVC-15 gives you downloadable reports and remote readouts.

- You can access TVC's password protected real-time display from any connected computer, even over the web. You'll know in an instant how well your programming is supporting watermark codes.
- You can download csv-formatted daily history reports of minute-by-minute actual code reliability, for custom analysis or for display in a program like Excel. Reports are private and you control who sees them.

And optionally, the big benefit for Voltair users:

TVC-15 can control your Voltair in real-time!

TVC-15's Intelligent Adaptive Enhancement [AE] closes the feedback loop, letting you dynamically control Voltair processing based on moment-by-moment analysis of your actual air signal.

You can take coding enhancement beyond simplistic "set and forget" or daypart setting strategies. TVC and Voltair work together like a continuous, intelligent automatic gain control on your hidden watermarks!

Have male and female hosts in a conversation? Got a call-in guest on a very compressed cell phone? Airing a stopset with jingles, dry announce, and produced sweepers? TVC-15 lets you compensate for all their different encoding requirements, continuously and with minimum annoyance to your listeners.

- Feed TVC-15 with your air signal, give it your Voltair's log-in address, and AE will constantly adjust your connected Voltair to provide just enough enhancement for the watermark confidence you want to achieve...while protecting the sound you want for your station, with minimum noticeable processing changes and artifacts.

Benefiting From TVC-15

Monitoring & Analysis of Station Encoding

It's vital to know that your watermarking system is working properly. Common wisdom in radio today is, "If you aren't encoding, you might as well be off the air."

But there aren't many ways to verify when you're encoding. The standard watermark encoder provides only a simple "no code" warning and basic error messages on an LCD. It won't alert you if a program stream is only marginally supporting watermarks. You might miss a lot of message opportunities before there's an alert.

The monitoring facility in Voltair is more powerful, sending initial warnings when 15 seconds have gone by without a valid message, and adding more warnings as the condition gets longer. Its front panel and optional downloadable reports give a minute-by-minute analysis of coding confidence, and let you simulate how various forms of environmental noise will affect it[5].

But both the standard encoder and Voltair's analysis can look only at the codes as they are being generated. Before those codes get to a listener, they'll often pass through a composite clipper or some data compression. Then they can be hit with transmitter issues or RF interference. In some installations, watermarking is also affected by airchain equalization or multiband compression.

You wouldn't consider your audio monitoring complete without a tuner, internet receiver, or some other form of real-world verification.

TVC-15 lets you do the same thing for your encoding.

You can feed TVC-15 with any audio signal, from a monitor receiver, a consumer radio that's flipping station-to-station, a field recording, a remote microphone, a router or patchbay... any source of analog audio.

On top of that, our sophisticated algorithms bring confidence analysis to levels that were never before possible with any system.

Near-instantaneous response:

- TVC-15's signal strength bar continuously responds to signal strength in the frequencies used by watermarking.
- It takes 400 ms for the encoder to create a valid code symbol, so TVC's front-panel graph updates that quickly: 150 times per minute. That's fast enough to indicate the differences when two on-air hosts have a conversation, or distinguish a sung jingle from a donut voice-over. The most recent two minutes of confidence measurements are displayed on a scrolling graph.
- A complete identification message requires 12 valid code symbols, carried on a combination of 10 different frequency channels. As soon as a valid ID is received, TVC's front-panel timer starts counting. If it takes too long for TVC to see a new valid message, the timer changes color.

More detailed information:

- A detailed 0 – 100% display of the likelihood each potential message will be received.
- Identification tags for each encoder. You can tell at a glance which of your streams—or your competitors'—is being analyzed.
- The timestamp encoded in each successful message. You can tell at a glance if an encoder's clock isn't accurate, a situation which can interfere with reliable ratings.

Complete remote access:

- TVC has a built-in, password-protected web server. You can log in with any connected browser, and assign different users the ability to either monitor TVC's readings, or remotely control its behavior.

Downloadable full reports:

- TVC's internal web server also lets you download a complete analysis of every signal TVC has received, available for any day it's been turned on. TVC reports are available as detailed files of each 4.8-second complete message analyzed over the course of a day, or as one-minute averages. They're in csv format, so you can analyze them with your own software, display them as an Excel spreadsheet, or compare them with station ratings reports. Reports are available only by password-protected log in: You control who sees the data.

Controlling Voltair Enhancement in Real-time

Voltair caused a revolution in station processing, enhancing watermarks so they'd have a better chance of being picked up by panelists' meters... even when a signal didn't support watermarking perfectly, or when a panelist was in a noisy environment. Voltair doesn't create 'phantom panelists' in the ratings system, but it helps make sure stations get credit for the listeners they really have. Unfortunately, too much enhancement can actually discourage listeners, breaking through the masking phenomenon, making watermark messages audible in the program stream. Listeners may hear this as extra noise or distortion. In extreme situations, they can be chased away.

It's a question of balance: You need enough enhancement to make codes reliable even during hard-to-encode program segments, or when there's a lot of environmental noise. But you don't want to annoy listeners. How much enhancement is *too much*? It depends on the program material, listening situation, and even listener expectations—the right enhancement for a news talk show might be too much for a high-quality acoustic music set.

Voltair includes tools including a "toggle test," to calibrate the amount of enhancement. It lets you add controlled amounts that can be correlated with ratings reports, so users can run their own tests. It also lets you preset three different Enhancement levels with GPIO control: You can have an "emergency watermark boost" button in master control, change enhancement when the host turns on his microphone, or have your station's automation system change enhancement for different dayparts.

But to get the highest level of control you'd need a trained operator, constantly monitoring your actual on-air signal with TVC, and continuously adjusting Voltair's enhancement for different air talents, audio sources, noise levels, and quality requirements. An operator who knows the personality and sound you want to present. One who's subtle enough to control watermark enhancement while avoiding abrupt or annoying changes. One who can pay perfect attention 24 hours a day, 7 days a week...

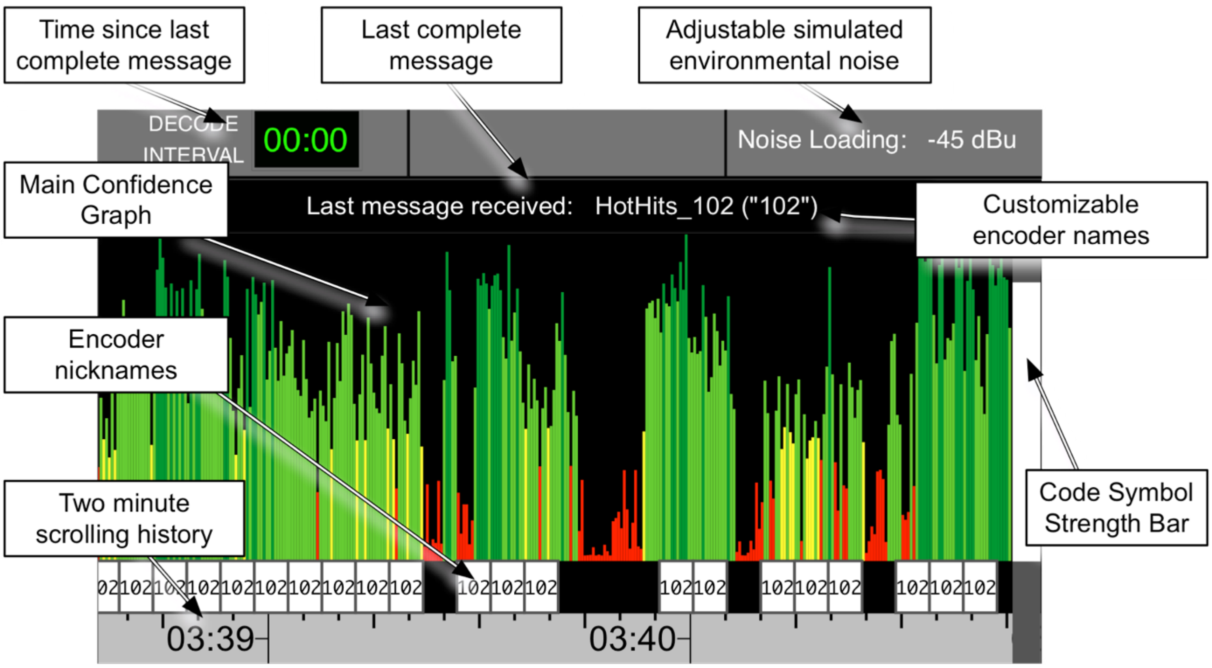
TVC-15's Intelligent Adaptive Enhancement can be that operator.

TVC-15, together with Voltair, closes the feedback loop around your watermarking ecosystem. It acts as a "smart AGC" for Voltair enhancement, monitoring actual encoding, and adjusting the amount of enhancement as quickly as twice per second. But like a good transmitter processor, you can fine-tune its behavior to preserve your station's unique sound, setting minimum desirable confidence levels, as well as maximum enhancement to annoying artifacts, how quickly enhancement can be changed, and more.

Finally: complete, full-time control over ratings enhancement levels!

FRONT PANEL

TVC-15 gives you a live, highly detailed display of actual watermark symbols, evaluated every 400 ms. for confidence, completeness and reliability.



Our proprietary algorithms constantly analyze the input signal, looking for valid code symbols that the system combines to build meaningful station identifications. The input can be any real-world source: your off-air signal (or a competitor's), a test file from a production studio, an Internet stream, or even a live mic listening to a sample radio or a public space[7]. If there are symbols hidden in the audio, TVC-15 will report their details.

The front panel LCD is arranged for maximum usability:

Time since last complete message

This reports minutes and seconds since the last successfully decoded message. It flashes green and restarts from 00:00 whenever a complete and coherent message is received.

- Continuously short timings are good: They mean the program includes a lot of reliable messages. The display will be green.
- Longer times between restarts mean the programming isn't supporting codes well. The display changes to yellow if ten seconds have gone by with no messages, to red if thirty seconds have gone by. While there may be exceptions, chances are a station won't be identified during those times.
- The Interval Display is constantly updating, and gives you a quick go/no-go indication of the current signal.

Last Complete Message Received

This is based on the actual Encoder ID that accompanied the last valid message, along with an optional display of the time stamp that accompanied it. Encoder IDs are arbitrary and set by the ratings agency, and don't include a station's call letters or frequency. So TVC-15 identifies them simply as **Encoder A**, **Encoder B**, and so on. You can rename them easily (to show call letters, frequency, HD stream, or any other useful tag), and TVC will use that name every subsequent time it sees that encoder.

The end of this line includes a short nickname in quotes. This nickname is used for flags at the bottom of the Main Confidence Graph.

Simulated Environmental "Noise Loading"

If everyone listened to broadcasts using headphones, the signal would go straight from the receiver into human ears. If they also used an adapter cable, it could go straight into a panelist's portable meter as well. But most listening is done with speakers, and in a variety of acoustic environments. Whether a panelist is driving their car, attending a sports event, or in a bar that has radio or TV for background, ambient noise is a factor that can affect how portable meters receive your code. So, to help gauge the impact of different noisy environments, we let you apply various levels of simulated noise.

25-Seven's Voltair is designed for real-world, real-time watermark evaluations while a station is broadcasting. It lets you simulate different listening situations with built-in recordings of actual random-noise environments (traffic with car honks, households with baby cries, dishes clattering in restaurant) and apply them to your measurements.

TVC's, however, can also be used for accurate offline comparisons of different program streams. The randomness of real-world noise can affect these comparisons, depending on each programs' timing. So TVC can generate a signal to simulate real-world noise in a repeatable way. It acts as a constant "load" on the watermark energy. It lets you compare different programs with the confidence that environmental noise will have a similar influence on each. You can also use this Noise Loading to scale TVC's measurements, for more convenient analysis and graphing.

If you want, you can substitute your own noise source instead. This can be recorded environmental noise that you mix with the test signal before feeding to TVC. Or it can be a live mic in a real-world space, picking up both your program and the location's actual noise.

This shows the confidence level for complete messages during the past two minutes. A complete message consists of twelve individual code symbols in a valid pattern, so TVC-15 draws a new Confidence line every 400ms. The line height displays zero to 100% confidence, and its color provides a quick visual reference:

- Dark green lines indicate 80% confidence or better. This can be the result of programming choices, Voltair enhancement, or a combination of both.
- Light green lines show at least 40% confidence. Many of your watermarks will probably get through, unless there's a lot of environmental noise.
- Orange lines show at least 30% confidence. Watermarks may be getting lost. Red lines show less than 30% confidence. There's a good chance panelists' meters won't register your station at all, even if they're actively listening.
- No line at all is rare, but can occur during prolonged silences.

Code Symbol Strength Bar

This white line constantly changes height to show the strength of potential code signals in watermark channels. This bar reacts instantly, to provide visual feedback that encoding *could* be taking place. Actual code symbols require 400 ms to broadcast, and they're measured and displayed in the Main Confidence Graph.

Encoder Nickname Tags

These are abbreviated from the encoder name, and mark each time a complete message comes through. The tag will be either the last word of the name, if it's short enough to fit on the graph; or the first three letters of the last word.

Two Minute History

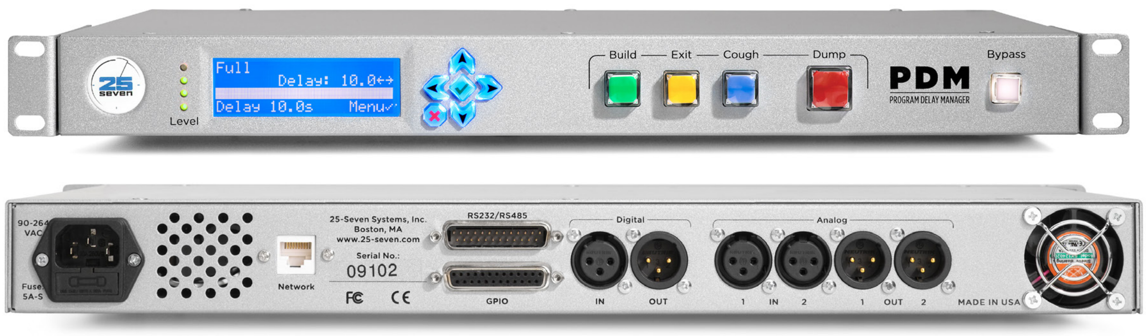
The time display on the bottom of the Confidence Graph is calibrated in minutes: seconds, based on TVC's real-time clock[10], to help you correlate confidence readings with moment-by-moment changes in your programming. This is *not* the time-stamp encoded on watermark messages.

Other User Controls

TVC-15 includes complete, flexible control over its operation. Clock and system settings, network access, how TVC controls a connected Voltair, and remote passwords can be set from the front panel. Most of these settings, along with maintenance and customization functions, are also available remotely using any web browser.

25-Seven® Program Delay Manager

Profanity Delay Reinvented



OVERVIEW

- PD Alert™ instantly emails time-stamped audio files whenever Dump is pressed
- Files capture what took place both on- and off-air
- Seamlessly builds and exits delay
- Configurable delay time, build and dump options
- Delayed IP data, serial streams and GPIO sync'd to audio

FEATURES

It's About Time

Leave it to 25-Seven® Systems to re-invent the profanity delay. Program Delay Manager (PDM) brings the possibilities of the Internet age to a "stand-alone box" technology that hasn't advanced much since the 1980's. Ease of use, transparent audio quality and program director friendly features converge in PDM to take an old process to a new level.

The Air Check is in the Email

Program Directors have more on their plates today than ever before. There's no way anyone can monitor every broadcast hour of every day, but PDs need to be the first to know what happened when that "dump" button got pressed.

With Program Delay Manager's patented PD-Alert™ feature, two time-stamped audio files capturing what took place both on air and off air get internally archived and emailed to the PD (or GM, or CE, or the legal team) every time questionable material is "dumped".

For stations serious about protecting their license, PDM provides an instant log record establishing your station's action and intent to keep the airwaves clean.

99 Seconds Of Delay Your Way

PDM comes standard with 99 seconds of stereo audio delay, and a dump button that can be set to remove any number of seconds you choose.

Build a delay through pre-rolling, time expansion or audio file play-out capabilities built right into PDM. Exit a delay through time compression or use the Cough button to simply wait and exit.

Dump audio through the standard "cut and rebuild" method, or use PDM's Overkill™ feature to play a "fill" file. Overkill allows you to select a show specific file from a list and play it over the dump buffer instead of collapsing the delay.

How PDM Does It

Superior Audio Algorithm Quality

25-Seven has a well-deserved reputation for offering the industry's most transparent time compression and expansion algorithms. Your listeners probably won't appreciate our superior, artifact-free audio because they won't perceive it's in use!

Flawless Expansion/Compression

25-Seven Systems' imperceptible audio time compression algorithms serve up smooth, crisp, stutter-free audio in PDM, even on stereo music. Unlike other products, we never splice at level thresholds or alter pitch. Clean audio is what we do best… now you can be sure the content is "clean" as well! Better algorithms mean delays can be rebuilt faster, so you can safely get back to callers. Build or Exit rates can be adjusted in real time, so you can be more or less aggressive, depending on audio content.

Audio, RDS, Data Streams and GPI/O Stay Synced

PAD or "now playing" data streams are delayed in precise synchronization with the audio as it grows, shrinks or whenever the dump button is pressed. PDM's data-follow-audio capabilities allow flexible synchronization from any data input to any data output. For example, serial data entering the RS-232 input can be routed to an IP output while remaining synchronized to the audio. 2 independent data delays are supported, and GPI/O closures stay in sync, too.

Future-Proof Audio Quality

Superior balanced analog I/O, with AES digital standard. 85dB s/n, response 25Hz-18kHz (+0/-0.2dB) and 0.02% THD+N… even during compression/expansion. Audio is always linear, so no lossy data reduction enters your signal path.

AES Digital, Balanced Analog or Livewire® AoIP

The first program delay to provide Audio over IP (AoIP) and control over Axia® Livewire audio networks, PDM comes in two models: one with balanced analog and AES digital I/O and the other with AoIP for Livewire. Whether you already have a Livewire network or you want to keep your plant AoIP-capable, PDM has you covered with Ethernet connectivity.

Superior Control

Choices, choices! PDM presents you with easy-to-use front panel controls, designed for the rigors of radio. Contact closure commands can be synced to the audio delay by the smart, programmable 8x8 GP I/O. Full bi-directional serial control over both RS232 and IP include advanced real-time status monitoring of parameters such as current delay depth and audio levels. A comprehensive web interface allows your PDM to be managed from nearly anywhere. Our Multi-View web feature permits networks and big facilities to monitor and manage up to 20 PDM's from a single browser screen.

Web Configurable

Say goodbye to hieroglyphs. Navigating through "set and forget" parameters is a breeze with our built-in web server. Change your settings, upload audio files and manage PDM's dump archives using simple browser screens, so you don't waste time trying to enter data though an ill suited LCD interface. Talk to PDM over your LAN or WAN. What could be easier?

IN DEPTH

Web Interface

In addition to controlling Program Delay Manager using GPIO (contract closures), PDM comes with a built-in, password-protected web server, allowing you to remote control your unit across a local or wide area network.

The server gives you five separate pages for complete and convenient control over your PDM.

Front Panel

An Adobe Flash-based application replicates PDM's front panel on your web browser, so every button and display is present and functions just like the real front panel. Through careful client-server communications management, round-trip latency is almost imperceptible, creating a seamless user experience. You can even control PDM from multiple computers. Just open a web browser interface on each, and anything you do on one computer will be reflected on the others, as well as on PDM's physical front panel.

Configuration

Tired of learning hieroglyphics just to configure a profanity delay? Navigating though "set and forget" parameters is a breeze with the PDM's Configuration page. You'll find obvious control with all your settings on one simple screen, so you don't waste time entering data though an ill-suited LCD interface.

PD Alerts™

A dedicated page lists all of the PD Alert emails the PDM has sent to your chosen staff.

Dump Archive

A Dump Archive shows all your saved dumped audio files. Easily review just what's been cut out of your air stream.

Insert Files

Easy management of the files you can use for quickly building your buffer at the beginning of the show. No more flash drives and cryptic file names!

SPECIFICATIONS

Power Input

PDM comes with a standard IEC C14 power connector.

Network

PDM connects to a standard 100BASE-T network connection. This port is used for Axia Livewire, synchronization to a network time server, and secure remote control via a web browser. If connected to the Internet, it should be behind a hardware firewall.

Analog Inputs and Outputs

Stereo inputs are electronically balanced XLR females, pin 2 hot, with a load of 20kΩ: this makes it compatible with all modern electronically-balanced outputs. If fed from a transformer-balanced output, we recommend bridging a 680Ω resistor between pins 2 and 3. Outputs are electronically balanced XLR males, pin 2 hot, designed to feed a load of 600Ω or greater. Input and output sensitivity default levels can be set from the front panel, and can range between +20dBu and -10dBu.

Digital Inputs and Outputs

When set to AES/EBU via the configuration menu, this input conforms to IEC 958 Professional (5v p-p, 110Ω balanced) on XLR connectors. When set to s/pdif, the voltage and impedance switches to IEC 958 Consumer (.5v p-p, 75Ω unbalanced): connect signal to pin 2 and shield to pins 1 and 3. Digital output (selectable AES or s/pdif) is always active, regardless of whether you are using analog or digital inputs. PDM will lock to any valid 32 kHz, 44.1 kHz, or 48 kHz signal at the digital input connector, even if you have selected analog for the input. In that case, the digital input controls PDM's internal sample rate. If PDM is not connected to a digital input, it uses its own high-reliability 44.1 kHz sample clock

Axia Livewire Version Inputs and Outputs

On the Livewire version of PDM, audio connections are exclusively via the network. PDM-Axia also supports Livewire-based GPIO.

GPIO

Eight parallel control inputs and eight parallel control outputs appear on a DB-25 connector. Input and output functions are assigned through a configuration menu on the front panel. Inputs and outputs are opto-isolated for easy interface to other equipment. A +5v supply and ground are also brought out to the DB-25 for simple remote controls using pushbuttons and LED status readouts. The +5v supply can carry 200 mA, more than adequate for 8 LEDs and 8 logic inputs. It is protected by an internal, self-resetting thermal circuit breaker.

Detailed Specifications

Audio

- S/N ≥ 84 dBA with 10 dB headroom (≥94 dB dynamic range); THD @1 kHz < .01%; IMD (IHF) < .01%; Frequency response ± 0.5 dB, 20 Hz – 20 kHz, measured analog input to analog output.

Dimensions

- 1RU (rack unit); 19" W (with rack ears) x 12" D x 1.75" H (483 x 305 x 44mm)

Power

- 100-240 VAC, 50/60 Hz; typical consumption 32 VA.

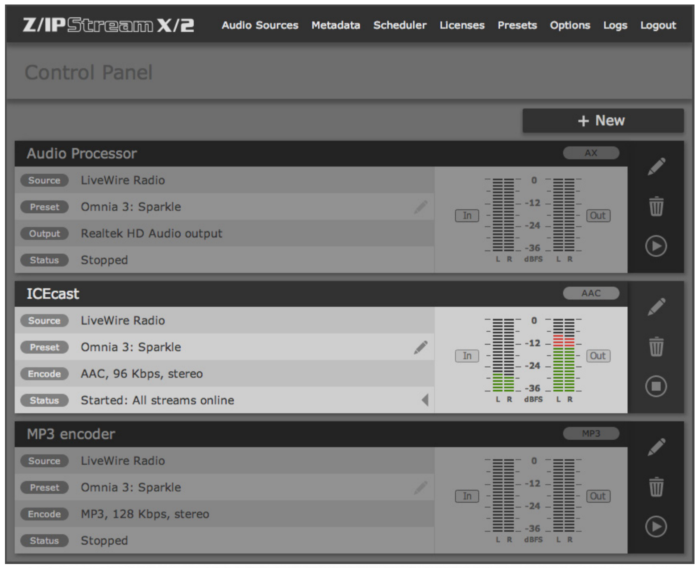
Regulatory

North America: FCC and CE tested and compliant, power supply is UL approved.

Europe: Complies with the European Union Directive 2002/95/EC on the restriction of the use of certain hazardous substances in electrical and electronic equipment (RoHS), as amended by Commission Decisions 2005/618/EC, 2005/717/ EC, 2005/747/EC (RoHS Directive), and WEEE.

Telos Alliance® Z/IPStream® X/2

The Future is Streaming. The Future is Here.



OVERVIEW

WiFi and Internet connections are available everywhere these days — and so is streaming audio. Already, many are using their smartphone as a 21st-century transistor radio. Soon, the connected car will make it easier than ever to listen to high-quality streamed audio on the move. These are big changes in listening habits, but don't worry: Z/IPStream X/2 is here to help.

Z/IPStream X/2 is the third-generation streaming software from the Telos Alliance®; a new combined audio processing/streaming platform designed for broadcasters who understand that streaming audio quality and reliability are just as important as terrestrial transmission. Z/IPStream X/2 gives you the power to fine-tune your streams for clear, clean, audio output — no matter the bitrate, codec, or delivery platform.

Z/IPStream X/2 stands above the rest with Adaptive Streaming technology. With Adaptive Streaming, the connection between streaming server and listener is automatically managed, dynamically adjusting bitrate and audio quality to maintain a solid connection with the best possible audio — regardless of Wi-Fi limitations or Internet behavior. Z/IPStream X/2 supports generating multiple streams to a server simultaneously using different codecs and bitrates to support these adaptive streaming applications.

FEATURES

- Genuine, high-quality audio codecs from Fraunhofer IIS (the inventors of MP3), including MP3, AAC-LC, HE-AAC v1, HE-AAC v2, and xHE-AAC
- Simultaneous MP3/AAC/aacPlus encoding, compatible with Shoutcast, Icecast, Wowza, and RTMP servers
- xHE-AAC works well at low bitrates and therefore has more encoding power. Whereas, other codecs like AAC and MP3 sound much better for music than they do for speech, xHE-AAC sounds great for both speech and music, even at the lowest bitrates
- Processes and encodes streaming audio for multiple platforms and bitrates simultaneously
- Includes 3-band processing from Omnia Audio
- Need even more processing power? Upgrade to Z/IPStream 9X/2, with up to seven bands of multiband AGC and limiting plus Undo technology that can restore poorly mastered audio to clarity and brilliance
- Sophisticated software routines enable you to handle streaming complications such as programming blackouts, metadata insertion, variable listener environments, and more
- Flexible audio routing accepts input from sound cards, RTP and Livewire AoIP connections
- Unprecedented level of control: Use the built-in HTML5 web interface, or fine-tune even further using the REST API
- Cloud-Ready: Z/IPStream X/2 may be hosted and run using your cloud-based server
- Built-in SNMP and email notification of system events
- Supports Kantar Watermarking

IN DEPTH

Adaptive Streaming

Adaptive Streaming is a stream-delivery method that allows media players to switch bitrates when network conditions change. Z/IPStream X/2 supports Microsoft's Smooth Streaming and Apple HLS adaptive streaming technologies, encoding the same stream at multiple bitrates and keeping audio packets sample-aligned. Adaptive Streaming ensures that your listeners are automatically receiving optimal quality and consistency based on the bandwidth of their connection.

Audio Replacement/Blanking

It is not uncommon for certain programming to be blacked out or contractually restricted from streaming online. Z/IPStream X/2 makes quick work of programming blackouts by enabling you to replace restricted material with content from a separate audio source, or audio from files. You have full control over the switch points and the duration, and the switch points are sample-accurate when using timestamped RTP audio for input.

Stream Synchronization

Stream synchronization is essential when implementing resilient streaming. Using Stream Synchronization, separate encoder instances (running on different PCs and even at different locations) are able to synchronize so that bitstreams generated by all instances are identical. This enables resilient streaming deployment through redundancy. If one encoder goes down (or is taken down for maintenance), the other encoder(s) continue to generate the appropriate stream, with no interruptions to service. Timestamped RTP input and Smooth Streaming for output are required to use Stream Synchronization.

Direct Livewire and RTP Audio Input

Z/IPStream X/2 works seamlessly with native Livewire audio sources, and can also accept RTP unicast sources.

SNMP Alarms

Z/IPStream X/2 can be monitored via SNMP, a feature particularly important for large-scale deployments. SNMP monitoring gives you peace of mind that your stream is fully functional, and if anything does go wrong, SNMP alarms will detect and immediately inform you of any problems.

REST API

In addition to an HTML5 web interface, Z/IPStream X/2 provides full programmatic control over its functions. Customers can use the REST API for configuration, monitoring, or dynamic control. REST is a web standard that can be used with the majority of scripting or programming languages from JavaScript to Python, Ruby, and more. Z/IPStream X/2 gives you complete control of your stream through a variety of industry-standard interfaces.

Cloud-Ready

Z/IPStream X/2 is a software-only application that's cloud-ready. It is designed to run in the background as a Windows service, and its HTML5 web interface makes remote configuration a breeze from PCs, Macs, tablets, or even smart phones. The REST API is ready to handle any additional custom control or monitoring requirements. Whether off-site or on, Z/IPStream X/2 gives you the flexibility to set up your stream however it best suits your needs.

Z/IPStream 9X/2: Full Omnia.9 Audio Processing

Z/IPStream X/2 can be upgraded to Z/IPStream 9X/2 at any time. Z/IPStream 9X/2 takes the already rock-solid 3-Band Omnia processing in the X/2 and elevates it with full Omnia.9 audio processing by Leif Claesson, which includes exclusive "Undo" Technology, the full Omnia.9 toolbox, and much more.

SPECIFICATIONS

- Windows 8 or later OS, 32-bit or 64-bit version
- 1 gigahertz (GHz) or faster 32-bit (x86) or 64-bit (x64) processor
- 1 gigabyte (GB) RAM (32-bit) or 2 GB RAM (64-bit)
- 200 MB free disk space required for installation
- Additional disk space is used for logging
- Internet access
- Administrative privileges required during installation
- Web browser required for configuration and management
- When AES67 is used as input, only stereo mode is supported
- Supports multiple simultaneous Wave audio interfaces

Telos Alliance® Z/IPStream® 9X/2

The ultimate, high-quality processing/encoding software with proprietary audio correction and sonic management

OVERVIEW

Z/IPStream 9X/2 includes all the functionality of its little brother the X/2, but adds technology found in the popular Omnia.9 audio processor. The 9X/2 is not simply a streaming processor-encoder, but a complete audio management system that will actually improve the flaws found in most recorded source material – both music and voice – as well as address the specific technical challenges of Internet distribution.

FEATURES

- Exclusive "Undo" technology with de-clipper prevents listener fatigue by removing distortion and selectively, undoing the over-compression so common in mastering today
- Optimizes sound quality of low bitrates by removing distortion components so that they do not waste bits during encoding.
- 6-band Parametric EQ for your signature sound
- Downward Expansion (source noise reduction)
- Multiband stereo enhancer
- Supports Kantar Watermarking

Additional Features

- Software only, no special cards required
- Includes Axia Livewire® driver
- Runs as a Windows service in the background, no need to log in
- Configure and monitor the application from any PC, tablet, or smartphone using an HTML5 web browser
- Manage the Omnia.9 audio processing from anywhere with NfRemote, locally or across the Internet
- Run multiple, fully independent stereo processors in one instance. Pay only for what you need. Upgrades available
- Flexible remote control application with touch screen support, comprehensive instrumentation, and remote audio streaming of any patch-point, also includes full speaker controller
- Separately adjustable sample rate (high-quality conversion) and gain control per encoded stream
- Extremely high audio quality, efficient CPU usage and low memory footprint

IN DEPTH

Z/IPStream 9X/2 comes with both a GUI application and a service that contain the exact same processing. During initial set-up (sound card configuration, etc.), use the GUI application. Once initial configuration is done and tested, switch over to using the Service, which you can then control with NfRemote from any computer.

Everything can be controlled with NfRemote except for which sound cards to use. Z/IPStream 9X/2 and NfRemote are standard Windows 32-bit native applications and do not use Microsoft.NET or similar.

Z/IPStream 9X/2 is primarily designed for streaming and only has one local sound card output. However, NfRemote has built-in dedicated PCM audio streaming for monitoring, so that you can monitor with low delay from any computer, for example, while adjusting the processing.

9X/2 can encode audio to MP3, AAC, and HE-AAC v1/v2 h (aacPlus). Low complexity AAC (AAC-LC), high-efficiency AAC (HE-AAC), and extended HE-AAC (xHE-AAC) are all supported. AAC has been standardized under both MPEG-2 and MPEG-4. The format most commonly used is MPEG-4 AAC-LC. Often this is called just 'AAC'. HE-AAC adds Spectral Band Replication to AAC and it is sometimes called AAC+ (sometimes seen as 'aacPlus' or 'AACplus'). There is also an HE-AAC v2 format which adds parametric stereo optimizations to HE-AAC. Sometimes this is called AAC+ v2 or Enhanced AAC+. Finally, xHE-AAC, the latest Fraunhofer codec, works well at low bitrates and therefore has more encoding power. Whereas, other codecs like AAC and MP3 sound much better for music than they do for speech, xHE-AAC sounds great for both speech and music, even at the lowest bitrates. 9X/2 can also use Windows Media codecs installed on the system, 48kbps or higher.

9X/2 can directly feed SHOUTcast-style servers (SHOUTcast, Icecast, Steamcast, etc.). The Wowza server is also supported for streaming to Flash clients. Windows Media streams can be sent to Windows Media server.

A few words about Undo

Undo is two stages:

First, the de-clipper removes distortion by detecting clipped edges of the waveform and resynthesizing the missing part. Unlike simpler algorithms, no distortion is ever created as the resynthesizing is performed entirely in frequency domain.

Second, the amount of short-term dynamics is detected for each of 5 frequency bands, and automatically controls the threshold and expansion ratio of 5 upwards expanders, to undo excessive compression and peak limiting.

Both techniques together result in an incredible "is that really the same recording" level of improvement. Audio quality of low bitrate codecs is also vastly improved, as a less distorted waveform is less complicated for the codec to encode (thus using fewer bits) and more dynamic, punchy sound gives the codec a place to hide the bitrate reduction artifacts.

SPECIFICATIONS

System requirements

- 9X/2 will run on Windows XP or newer. Minimum requirements are Core 2 Duo, 512 MB RAM

General

- A Core i7 2600 and 4 GB RAM comfortably runs 16 stereo processors with several encoders each
- Supports multiple simultaneous Wave audio interfaces
- Simultaneous MP3/AAC/aacPlus encoding, compatible with Shoutcast, Icecast, Wowza, and RTMP servers
- Only stereo AES67 input is supported

Telos Alliance® Z/IPStream® R/1

The Professional, One-Box Streaming Appliance

OVERVIEW

Z/IPStream® combines audio processing with MP3 and AAC encoding in one convenient, single-rack unit. The AAC encoder supports AAC-LC, HE-AAC and HE-AAC v2 formats, and is fully managed and configured remotely with any standard Web browser. Z/IPStream features a wideband AGC, 3-band compressor/limiter, EQ, low-pass filter and a precision look-ahead final limiter; processed audio can then be encoded directly to MP3 or AAC streams to feed a remote replication server at your ISP. Streams can be "tagged" with "now-playing" information received from automation systems. A full complement of I/O, including Livewire, analog, and AES/EBU, is standard.

FEATURES

- Audio pre-processing, stream encoding and delivery to remote replication server, all in a professional 1RU appliance.
- Pro-grade 24-bit A/D converter for studio-reference quality audio.
- Choice of MP3 or AAC-LC, HE-AAC, HE-AAC v2 stream coding, with output bit rates from 16 kbps to 320 kbps (dependent upon active codec).
- Omnia audio processing includes wideband AGC, 3-band compressor/limiter, EQ, low-pass filter and precision look-ahead final limiter.
- Metadata support for all popular playout platforms allows streams to be dynamically tagged with "now-playing" information from automation systems.
- Studio-grade analog and Livewire IP-Audio I/O, with separate LAN & WAN Ethernet ports.
- Directly supports ICEcast, SHOUTcast, SHOUTcast v2, Adobe Flash Media Server as well as Adobe RTMP, RTP streams (including RTP multicast), as well as LimeLight, Akamai and other popular streaming servers.
- Dual encoder support can be used to provide high and low bitrate streams, or MP3 and AAC at the same time.
- Can accept metadata over TCP/IP, UDP, or RS-232 (using USB to RS232 adapter).
- HTML5 web user interface for remote configuration and management from computers and mobile devices.

IN DEPTH

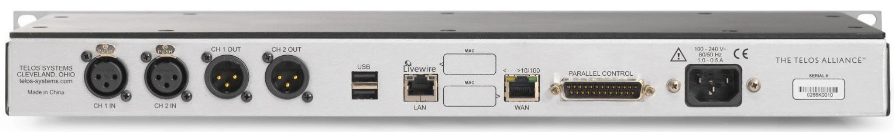
Plug. Play. Stream.

For years, the way to stream audio to Internet listeners included unbalanced mini-jacks, poor-quality sound cards, one or more PCs to maintain, and a collection of software that didn't always play nicely together. Broadcasters asked for a professional, PC-free Web streaming solution — and Telos delivers.

Z/IPStream R/1 takes the hassle out of streaming. There's no PC needed; Z/IPStream R/1 takes just 1RU of rack space. Slide it in and it's ready to gostreaming. Just send audio to Z/IPStream R/1, make a few setup selections and, within minutes, you'll be streaming your programming to your favorite stream server or streaming service for worldwide distribution.

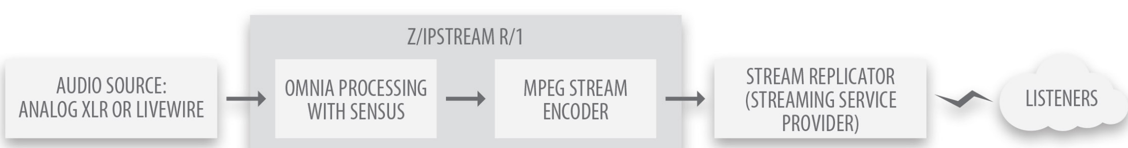
Broadcasters know that Telos is the codec expert, and Omnia is the processing authority. Z/IPStream R/1 puts all of our expertise into one integrated streaming appliance. First, incoming audio gets treated to pre-processing from Omnia, using algorithms that work hand-in-glove with Z/IPStream R/1's codecs to shape and optimize audio prior to encoding. Then, genuine MPEG encoding algorithms from FhG, the inventors of MP3, ensure the most artifact-free sound quality at whatever bit rate you choose. Encode directly to an MP3 or MPEG-AAC stream, then send it to a Shoutcast, Wowza, Icecast, LimeLight, Akamai, Adobe Flash Media server, or other popular streaming server for distribution to your waiting listeners.

Setup is a breeze. Log in with a laptop and Web browser for easy setup or remote control, or tweak the front-panel controls. This web interface is based on HTML5 so it's accessible from most modern browsers, including smart phones, tablets, and other devices. There's also a convenient built-in headphone amp with 1/4" jack and volume control for last minute in-the-rack fine tuning.



Z/IPStream R/1 comes with studio-grade analog inputs and outputs, plus Livewire Audio over IP. On the output side, Z/IPStream R/1 delivers fully processed, unencoded audio as well as encoded audio, providing your studio with another source for processed sound. Full network connectivity is provided via two Ethernet jacks, one for the LAN (including Livewire) and the other for the WAN and streaming.

The Professional Choice for Streaming Audio.



Optimizing sound quality is as essential on the web as it is on traditional formats. Z/IPStream R/1 has a built-in processing section that works together with the streaming encoder, optimizing your audio for stunning sound — even after bit-reduction. This isn't just some cheap leveler – it's real processing by Omnia, complete with wideband AGC, a 3-band combined compressor/limiter, high-frequency EQ, an adjustable-bandwidth low-pass filter, and Omnia's famous anti-aliasing final Look-Ahead limiter. There are even a selection of presets, tailored to specific formats and bit rates, to help you get up and running quickly.

Of course, the foundation for high fidelity audio distribution rests on professional encoding technology. The quality of the encoder directly affects the quality of the output. Telos has a long history of partnership with Germany's Fraunhofer Gesellschaft Laboratory (FhG), the world leader in audio compression research and the inventors of MP3; Z/IPStream R/1 uses genuine MP3 and MPEG-AAC encoding algorithms to ensure the most artifact-free sound quality at any bit rate you choose, from 16 kbps all the way to 320 kbps. No other encoder has this pedigree, or achieves this level of quality and performance. Generic "mp3" encoders can't come close.

Z/IPStream R/1 gives you a wide choice of genuine Fraunhofer encoding algorithms, which include MP3, the Standard for digital audio. It's the safest codec choice for compatibility with the widest variety of listening devices. Or choose AAC-LC, a high performance codec for excellent audio quality at lower bitrates. AAC-LC is in widespread use, most notably in Apple's iTunes. And then there's High Efficiency Advanced Audio Coding, or HE-AAC, a newer AAC codec which incorporates Spectral Band Replication (SBR) bandwidth expansion to improve audio at very low bitrates. HE-AAC v2 applies a Parametric Stereo feature to HE-AAC codec allowing for even further reduction in bandwidth.

When you're done processing and encoding, select your metadata source and feed your stream to any SHOUTcast or SHOUTcast v2-compatible media server, or a Wowza server for streaming to Flash clients. Z/IPStream R/1 works with ICECast and Adobe Flash Media and Adobe RTMP servers too, as well as popular streaming services from LimeLight, Akamai, and other popular streaming service providers. You can feed directly to a streaming server on your LAN, to an Internet streaming relay service via the WAN port, or take processed audio from the rear-panel XLR outputs. No matter what your audio source or how you stream, Z/IPStream R/1 delivers flawlessly optimized audio that sounds terrific.

SPECIFICATIONS

Audio Coding

Codec Choices:

- MP3: 16 to 320 kbps
- AAC-LC: 24 to 320 kbps
- HE-AAC: 24-96 kbps
- HE-AAC v2 (aacPlus): 24-96 kbps

AAC Transport Modes:

- ADTS
- ADTS-CRC
- ADIF
- RAW

Metadata Formats:

- Character Parser Sample
- Line Parser Sample
- Nexgen Audio Sense
- Simian Template 1
- XML Parser Sample
- XML-Jazler
- XML-Jazler2
- XML-MediaTouch
- XML-MediaTouch2
- XML-Sample2
- XML-Zetta
- User-definable

Input

- Analog: Balanced XLR, +4 dBu
- Input Impedance: 6K Ohm differential
- Analog to Digital Converter: 24bits
- Digital: Livewire AoIP, via LAN port

Output

- Analog: Balanced XLR
- Output Clipping: + 22dBu
- Output Impedance: 50 Ohm differential
- Digital to Analog Converter: 24bits
- Digital: Livewire AoIP, via LAN or WAN port

Audio Performance

- THD+N: < 0.03% @ +12dBu, 1 kHz Sine
- Frequency Response: +/- 1dB 25– 20 kHz
- Headroom: 18dB
- Dynamic Range: > 87dB Unweighted > 90 dB "A" Weighted
- Crosstalk: > 80 db

Remote Control

- HTML5 web interface via built-in webserver

Power

- Internal supply, 85–250 VAC auto-switching, 50–60 Hz
- Power consumption: 14.2 Watts

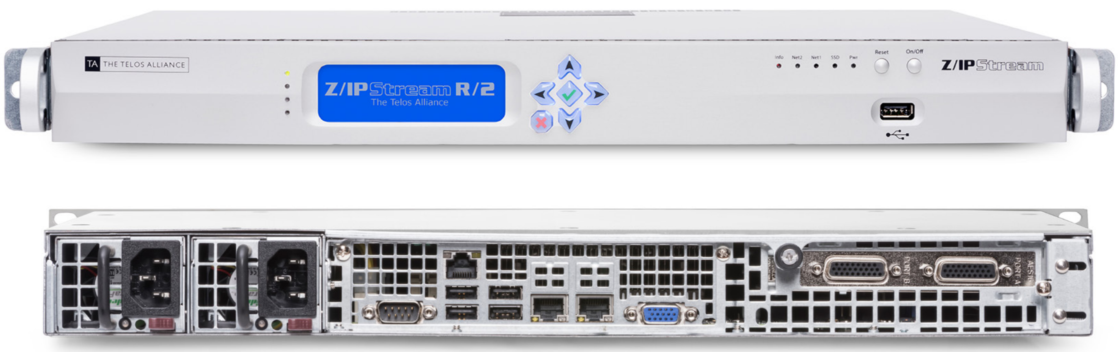
Regulatory

North America: FCC and CE tested and compliant, power supply is UL approved.

Europe: Complies with the European Union Directive 2002/95/EC on the restriction of the use of certain hazardous substances in electrical and electronic equipment (RoHS), as amended by Commission Decisions 2005/618/EC, 2005/717/ EC, 2005/747/EC (RoHS Directive), and WEEE.

Telos Alliance® Z/IPStream® R/2 Stream Encoder/Processor

The Best-Sounding Streams...from the Smallest Box.



OVERVIEW

Processing and Encoding of Multiple Programs

Z/IPStream® R/2 produces the best possible streams by providing a multitude of streaming options for the broadcaster and maximizing audio quality for the listener. This second-generation Z/IPStream processor and encoder is essentially the hardware appliance version of the successful X/2 and 9X/2 software, allowing flexible, multi-format stream-encoding for up to eight audio programs in a single 1RU chassis.

Ideal for high-density processing and encoding applications, R/2 offers the simplicity and reliability of a single 1RU dedicated hardware appliance. R/2 is available with 3-band Omnia processing or full Omnia.9 processing, both featuring high-quality Telos® encoding.

FEATURES

- Processing and stream encoding of up to eight audio programs in 1RU
- Available in models with 3-band or full Omnia.9 processing
- AES/EBU and Livewire® audio I/O
- Encode a program at multiple bitrates for adaptive streaming applications. Apple HLS and Microsoft Smooth Streaming formats are supported
- AAC-LC, HE-AAC, HE-AAC v2, xHE-AAC and MP3 stream encoding from 16 kbps to 320 kbps depending on codec used
- xHE-AAC for low-bitrate streaming
- Dual power supplies and dual gigabit Ethernet ports for reliable, 24/7 operation
- Processing or encoding can be used independently if desired
- Process and encode the same audio program in multiple formats. Simultaneously send the encoded streams to multiple destinations
- Supported server platforms include ICEcast, SHOUTcast, SHOUTcast v2, Adobe Flash Media Server, Wowza, as well as Triton Digital, LimeLight, Akamai, and other popular streaming services
- Includes support for RTP and RTP multicast streams
- Built-in HTTP server can directly serve HLS streams
- HTML5 web-based remote control for administration
- SNMP support allows direct monitoring from your SNMP management system, or you can receive alerts via email
- Integrates into your workflow: REST-ful API allows full control from your application to start/stop streams, switch audio sources or insert audio content from files; or monitor multiple devices simultaneously
- Dedicated IP remote control software with test instrumentation (RTA, FFT, oscilloscopes, loudness metering) for audio-processing adjustments when using Omnia.9 processing
- New flexible Metadata allows R/2 to accept metadata from multiple play-out systems and lets broadcasters tweak the fields they want to present to listeners
- Supports Kantar Watermarking

IN DEPTH

Z/IPStream R/2 is the latest generation of streaming audio processing and encoding hardware in the Z/IPStream family, handling processing and encoding of multiple audio programs in a compact 1 RU chassis. Processing and encoding of up to eight audio programs is supported, with Livewire and AES/EBU I/O audio input.

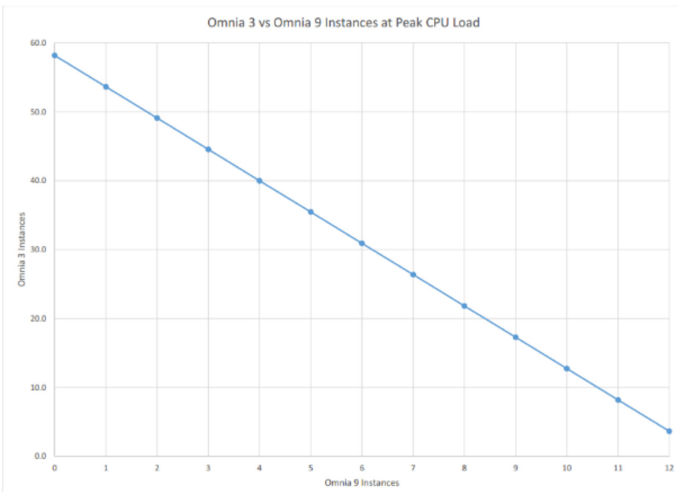
The base unit includes processing and encoding of two audio programs using the standard 3-band Omnia audio processing. Encoding formats including MP3, AAC-LC, HE-AAC, HE-AAC v2, and xHE-AAC. Multiple codecs and bitrates are supported simultaneously on each audio program. A special multirate AAC encoder is included for adaptive bitrate streaming applications. Supported streaming platforms include ICEcast, SHOUTcast, SHOUTcast v2, Adobe Flash Media Server, Adobe RTMP, Triton Digital, LimeLight, Akamai, and Wowza. Additional audio program inputs and Omnia.9 processing are available as options.

The Z/IPStream R/2 with Omnia.9 processing models include exclusive 'Undo' de-clipping, 6-band parametric EQ, downward expansion (source noise reduction), multiband stereo enhancer, up to three-stage AGC with adjustable sidechain filter, 2-7 bands of processing, and final two-band look-ahead limiter. Full IP remote control of processing parameters is available via NFRemote, along with the complete Omnia.9 suite of test instrumentation (loudness metering, FFT, RTA, oscilloscope, and remote client audio streaming).

SPECIFICATIONS

Processing

- Includes standard 3-band Omnia processing. Optionally use full Omnia.9 processing with up to 7 bands of processing. The number of audio processing instances that may be used simultaneously depends on overall system configuration and resource usage. As expected, Omnia.9 is more resource-intensive than the 3-band Omnia processor. The chart below illustrates the number of instances that can be run under typical usage scenarios. It is provided as a guide, the actual number may be different for your specific application.



Stream Encoding

- Includes AAC-LC, HE-AAC, HE-AAC v2, xHE-AAC, and MP3 encoding at bitrates from 16 kbps up to 320 kbps (depending on codec). A program may be encoded using multiple codec formats and bitrates simultaneously. A special multirate encoder supports encoding for adaptive streaming applications. The multirate encoder properly generates the required Stream Access Points for adaptive streaming.

Ethernet Remote Control

- Gigabit Ethernet supports HTML web interface for administration, REST API for remote control, and SNMP monitoring. Also used with dedicated remote control application for Omnia.9 processing. Various metadata update methods via Ethernet supported as well.

Front Panel Controls and Indicators

- Directional navigation cluster
- Graphical LCD display
- Power/Reset controls
- Diagnostic LED indicators (power, network, drive activity)
- USB port

Audio I/O

- Livewire and AES/EBU audio I/O
- Supports AES/EBU input at up to 24 bits, 192 kHz
- Supports direct input from RTP streams

Power Requirements

Dual power supplies, each rated at 100-264 VAC, 50/60Hz, auto-sensing, 100W max total

Dimensions and Weight

- One rack unit— 1.75"H x 19"W x 15.5"D (44 x 483 x 394 mm)
- Net weight: 9 lbs (4 kg); shipping: 12 lbs (5.4 kg) approximate

Environmental

- Fan cooled
- Operating: 0 to 50 degrees C
- Non-operating: –20 to 70 degrees C

Regulatory

North America: FCC and CE tested and compliant. Power supply is UL approved.

Europe: Complies with the European Union Directive 2002/95/EC on the restriction of the use of certain hazardous substances in electrical and electronic equipment (RoHS), as amended by Commission Decisions 2005/618/EC, 2005/717/ EC, 2005/747/EC (RoHS Directive), and WEEE.

Warranty

For the latest Telos Alliance warranty, visit: telosalliance.com/warranty

Linear Acoustic® ARC (Automatic Realtime Control)

Next Generation Television Loudness Processor

OVERVIEW

The Linear Acoustic ARC is a budget-friendly, easy-to-use, 2-channel television processor specifically designed for regions and applications that do not require support for multi-channel, coded audio, but where no-compromise audio quality is valued.

FEATURES

- Audio processing using the Linear Acoustic APTO® loudness adaptation algorithm
- Compliance for any loudness recommendation or regulation including EBU R 128 and ATSC A/85 (CALM)
- Processing for two PCM stereo or mono program streams
- Ability to specify target loudness and True Peak values
- 3Gb/s HD/SD-SDI and AES-3 I/O
- AES67 I/O supports SMPTE ST 2110-30 workflows
- Rolling 6-hour loudness logging for each program
- Separate loudness event log to easily identify loudness issues plus system event log
- System event log
- SNMP
- Dual internal redundant auto-ranging power supplies
- Browser-based remote control

IN DEPTH

The Right Features at the Right Price

Until now, broadcasters looking for a straightforward, stereo, television processor had to make some difficult compromises.

Less costly solutions were budget-friendly, but the savings came at the expense of audio quality. Products that delivered excellent audio performance also included features such as support for multi-channel, coded audio and audience measurement watermarking that simply aren't required in many regions – and with a price tag that put them out of reach to smaller broadcasters.

Linear Acoustic ARC eliminates that compromise by offering a 1RU DTV audio processor for two independent PCM stereo or mono program sources that delivers the viewer-pleasing audio upon which Linear Acoustic has built its reputation.

APTO Processing

ARC features the Linear Acoustic APTO loudness adaptation algorithm which carefully controls levels in a way that preserves transients, sonic image, and the artistic intent of the original audio source while still ensuring full compliance with any loudness recommendation or regulation including EBU R 128 and ATSC A/85 (CALM).

Simple, Straightforward Setup

Setting up ARC couldn't be simpler: Select a suitable adaptation profile for your programming from the ample list of factory presets, adjust a single control to determine the amount of overall processing desired, set the desired loudness target, and walk away with confidence.

A front panel navigation cluster is used for initial setup, while a web-based, browser- and OS-agnostic remote user interface makes more detailed setup and monitoring easy and convenient on any computer or mobile device. A front panel color LCD display clearly shows audio levels and loudness information.

Ready for Today, Ready for the Future

I/O includes AES-3, 3Gb/s HD/SD-SDI, and AES67, making ARC suitable for use in today's typical broadcast facilities, but also compliant with SMPTE ST 2110-30.

SPECIFICATIONS

Processing

- Linear Acoustic APTO® Loudness Adaptation algorithm; processing for two stereo or mono programs

Logging

- Rolling 6-hour loudness logging for each program
- Separate loudness event log to easily identify loudness issues
- System event log

AES-3 I/O

- One 2-channel input and one 2-channel output via 75 Ohm BNC female connectors, internally terminated; signal levels per SMPTE 276M/AES-3ID-2001

SDI I/O

- One auto-sensing 3Gb/s HD/SD-SDI (SMPTE ST425/292M/259M) input and one output via 75 Ohm BNC female connectors, internally terminated; video formats up to 1080p/60/59.94/50Hz

AES67 I/O

- 16-channels of bi-directional AES67 I/O in support of SMPTE ST 2110-30 workflows

Reference

- 48kHz reference via SDI, PTP, AES-3, or internal clock

Sample Rate/Resolution/Frequency Response

- 48kHz, 24-bit, 20Hz – 20kHz

Ethernet

- Two Gigabit RJ-45 connections – one for AES67, one for networked remote control

Parallel GPI/O Control Port

- 15-pin female D connector, 0-5V TTL levels, 5 GPI/O inputs, 5 GPI/O output

SNMP

- Traps include loudness above/below target, loudness within target window, change in reference, and power supply status

Front Panel Controls and Indicators

- 5-key navigation cluster; graphical color LCD display; LED status indicators for each power supply, system status, and reference

Power

- Dual internal redundant auto-ranging power supplies, each rated at 100-264VAC, 50/60Hz, 40 Watts maximum

Dimensions and Weight

- 19" W x 9" D x 1.75" H (approximately 48.2 x 22.9 x4.5 cm)
- Net weight: Approximately 9.0 lbs (4.08 kg)
- Shipping weight: Approximately 12.0 lbs (5.44 kg)

Environmental

- Operating: 0 to 50 degrees C
- Non-Operating: -20 to 70 degrees C

Intended Location

- Telecommunications center or dedicated computer/machine room

Regulatory

- North America – FCC and CE tested and compliant with UL-approved power supplies
- Europe – Complies with European Union Directive 2002/95/EC on the restriction of use of certain hazardous substances in electrical and electronic equipment (RoHS), as amended by Commission Decisions 2005/618/EC, 2005/717/EC, 2005/747/EC (RoHS directive), and WEEE

Warranty

- Standard Telos Alliance 2-year limited parts and labor

Linear Acoustic® AERO.10

DTV Audio Processor

OVERVIEW

Cost-effective PCM loudness control without compromising quality.

AERO.10 hosts a single AEROMAX® processing instance in AMX5.1 (5.1+2+2), AMX2.0 (2+2+2), or AMX 5x2 (2+2+2+2+2) configurations (user-selectable) plus upmixing/downmixing via our UPMAX®-II algorithm.

I/O includes de-embedding and re-embedding of eight pairs of HD/SD-SDI audio, four pairs of AES audio, and balanced analogue stereo.

Support for SAP/DVS, EAS, local emergency audio, local voiceover, and optionally, Audio Description (warble tone) functionality is included.

ITU-R BS.1770-3 and EBU R-128 metering and logging (including True Peak) is provided for all program outputs. NfRemote software is included for remote configuration, control and metering over an Ethernet connection while a built-in HTTP server enables control of I/O, presets, and individual processing parameters using simple IP commands. Compensating video delay and dual redundant internal power supplies are standard.

FEATURES

- Linear Acoustic AEROMAX loudness and dynamics control
- UPMAX-II automatic upmixing and downmixing
- Single processing instance in AMX5.1 (5.1+2+2), AMX2.0 (2+2+2), or AMX 5x2 (2+2+2+2+2) configuration (user-selectable)
- 8 audio pairs via HD/SD-SDI I/O with included video delay
- 4 audio pairs of AES I/O with reference input
- Balanced +4dBu stereo analog inputs and outputs
- Dual auto-ranging power supplies
- Relay bypass of all I/O
- Front panel GUI plus extensive TCP/IP and HTTP control
- Logging of loudness and True Peak data

IN DEPTH

The highest quality industry standard television loudness control has never been more affordable. AERO.10 is a full-featured audio processor specifically for PCM (non-coded) audio that uses the same AEROMAX processing engine as our AERO.100 and AERO.2000 products. Add upmixing, downmixing, and ITU- and EBU-compliant loudness metering and logging and it's easy to see why AERO.10 is such a powerful solution for nearly any application despite its low cost.

AERO.10 offers a simple LCD front panel with a headphone output designed to provide plenty of level even for difficult loads or quiet sources - useful for checking audio or adjusting processing right in the rack.

Comprehensive TCP/IP remote provides control over all system settings and processing parameters plus extensive loudness metering. It also offers reporting of physical I/O details, power supply status, and environmental health. The remote application also delivers remote audio, up to 5.1 channels, so the user can audition signal quality anywhere link bandwidth permits. An HTTP server is included for simple get/set control of all parameters and retrieval of status and logging information.

Constantly active logging captures 24 hour, 48 hour, and 7.5 day rolling weekly reports as well as specific time slots controlled by start/stop. Loudness measurements with multiple integration times as well as True Peak measurements are captured and available for download.
Failover bypass relays on AES and SDI I/O maintain signal continuity and dual auto-ranging power supplies enable redundancy and worldwide compatibility.

Designed and built in the USA, the lightweight and rugged single rack-unit AERO.10 is a solid investment in performance and flexibility.

SPECIFICATIONS

Processing

- AEROMAX processing for PCM audio in your choice of AMX5.1 (5.1+2+2), AMX2.0 (2+2+2), or AMX 5x2 (2+2+2+2+2) configurations (user-selectable)
- Dual UPMAX-II two-channel to 5.1 channel upmixers plus main channel downmixing and automatic bypass of discrete content

Sample Rate/Resolution/Frequency Response

- 48kHz, 24-bit, 20Hz to 20kHz below threshold

AES I/O

- Four audio input pairs plus reference via 75-Ohm BNC female connectors, internally terminated; Four audio output pairs; Signal levels per SMPTE 276M/AES-3ID-2001

SDI I/O

- Eight I/O audio pairs with auto-sensing HD/SD-SDI (SMPTE 292M/259M) inputs, up to 1080i/60/59.94/50Hz, access to audio and VANC metadata

Analog I/O (stereo)

- 9-pin female D connector; 10K Ohm balanced stereo inputs; Balanced stereo outputs, +4dBu nominal, +24dBu maximum into 600 Ohms.

Parallel GPI/O Control Port

- 25-pin female D connector, 0-5V TTL levels for 8 inputs and 8 outputs; controls simple preset recalls, voiceover/EAS insertion, or customized scripts

Ethernet Remote Control

- Gigabit Ethernet supports included TCP/IP remote control application and HTTP server access

Front Panel Controls and Indicators

- Rotary navigation cluster, Graphical LCD display, headphone volume control
- 6.3mm front panel headphone connector, +12dBu Max into 600 Ohms

Serial Metadata

- 9-pin female D connector; 115.2 kbps; pinout per SMPTE 207M (RS-485); Designed to directly interface with Dolby serial metadata (SMPTE RDD6)

Power Requirements

- Dual power supplies, each rated at 100-264 VAC, 50/60Hz, auto-sensing, 100W max. total

Dimensions and Weight

- One rack unit - 1.75"H x 19"W x 15.5"D (44 x 483 x 394 mm) Net weight: 9 lbs (4 kg); shipping: 12 lbs (5.4 kg) approximate.

Environmental

- Fan cooled. Operating: 0 to 50 degrees C, non-operating –20 to 70 degrees C.

Regulatory

North America: FCC and CE tested and compliant, power supply is UL approved.

Europe: Complies with the European Union Directive 2002/95/EC on the restriction of the use of certain hazardous substances in electrical and electronic equipment (RoHS), as amended by Commission Decisions 2005/618/EC, 2005/717/ EC, 2005/747/EC (RoHS Directive), and WEEE.

Warranty

For the latest Telos Alliance warranty, visit: telosalliance.com/warranty

Linear Acoustic® AERO.100™

Loudness Management Platform for DTV

OVERVIEW

High quality, compliant audio in a compact 1RU design.

AERO.100 hosts one or two AEROMAX® processing instances in your choice of AMX5.1 (5.1+2+2), AMX2.0 (2+2+2), or AMX 5x2 (2+2+2+2+2) configurations (a minimum of one is required) plus upmixing/downmixing via our UPMAX®-II algorithm.

Optional Dolby® Digital Plus transcoding provides for decoding of Dolby Digital/Dolby Digital Plus content to PCM audio for loudness processing and encoding to Dolby Digital/Dolby Digital Plus for transmission.

I/O includes de-embedding and re-embedding of eight pairs of HD/SD-SDI audio and four pairs of AES audio.

Support for SAP/DVS, EAS, local emergency audio, local voiceover, and optionally, Audio Description (warble tone) functionality is included. CrowdControl™ is standard for increased dialogue intelligibility.

ITU-R BS.1770-3 and EBU R-128 metering and logging (including True Peak) is provided for all program outputs. NfRemote software is included for remote configuration, control and metering over an Ethernet connection while a built-in HTTP server enables control of I/O, presets, and individual processing parameters using simple IP commands. Compensating video delay and dual redundant internal power supplies are standard.

FEATURES

- Linear Acoustic AEROMAX loudness and dynamics control
- UPMAX-II automatic upmixing and downmixing
- One or two processing instances in AMX5.1 (5.1+2+2), AMX2.0 (2+2+2), or AMX 5x2 (2+2+2+2+2) configuration
- CrowdControl for increased dialog intelligibility
- Support for SAP/DVS
- Available Dolby® Digital Plus transcoding, including Dolby Digital/Dolby Digital Plus decoding to PCM and encoding to Dolby Digital/Dolby Digital Plus for transmission
- Available Nielsen watermark encoding
- 8 audio pairs via HD/SD-SDI I/O with included video delay
- 4 audio pairs of AES I/O with reference input
- Dual auto-ranging power supplies
- Relay bypass of all I/O
- Extensive TCP/IP remote control and HTTP control
- Logging of loudness and True Peak data

IN DEPTH

AERO.100 brings our industry standard, no-compromise AEROMAX television loudness control and UPMAX-II upmixing/downmixing algorithm to a space-saving 1RU design. Add ITU- and EBU-compliant loudness metering and logging to the package along with the ability to host two processing instances and it's clear that preserving valuable rack space doesn't require giving up features or performance.

Ideal for situations where a full front panel display and controls are not required, AERO.100 includes a comprehensive TCP/IP remote to provide control over all system settings and processing parameters plus extensive loudness metering. It also offers reporting of physical I/O details, power supply status, and environmental health. The remote application also delivers remote audio, up to 5.1 channels, so the user can audition signal quality anywhere link bandwidth permits. An HTTP server is included for simple get/set control of all parameters and retrieval of status and logging information.

Constantly active logging captures 24 hour, 48 hour, and 7.5 day rolling weekly reports as well as specific time slots controlled by start/stop. Loudness measurements with multiple integration times as well as True Peak measurements are captured and available for download.

Failover bypass relays on all I/O maintain signal continuity and dual auto-ranging power supplies enable redundancy and worldwide compatibility.

Designed and built in the USA, the lightweight and rugged single rack-unit AERO.100 is a solid investment in performance and flexibility today and in the future.

SPECIFICATIONS

Processing

- Can host one or two instances of AEROMAX processing in your choice of AMX5.1 (5.1+2+2), AMX2.0 (2+2+2), or AMX 5x2 (2+2+2+2+2) configurations
- Dual UPMAX-II two-channel to 5.1 channel upmixers per instance plus main channel downmixing and automatic bypass of discrete content

Audio Encoding/Decoding

- Available Dolby® Digital Plus transcoding, including Dolby Digital/Dolby Digital Plus decoding to PCM and encoding to Dolby Digital/Dolby Digital Plus for transmission
- Nielsen watermark encoding

Reference

- 48kHz via AES DARS (or any AES signal applied to the Ref In connector), AES In 1, SDI, or from the internal 48kHz clock (standalone use only)

Sample Rate/Resolution/Frequency Response

- 48kHz, 24-bit, 20Hz to 20kHz below threshold

AES I/O

- Four audio input pairs plus reference via 75-Ohm BNC female connectors. Four audio output pairs outputs plus encoder output. All digital inputs are 75 Ohm internally terminated, unbalanced. Signal levels per SMPTE 276M/ AES-3ID-2001

SDI I/O

- Auto-sensing HD/SD-SDI (SMPTE 292M/259M) inputs, up to 1080i/60/59.94/50Hz. De-embed up to eight audio pairs from applied SDI signal, process and/or encode, and re-embed up to to eight audio pairs. Supports SMPTE 2020 A and B VANC metadata

Parallel GPI/O Control Port

- 25-pin female D connector, 0-5V TTL levels for 8 inputs and 8 outputs; controls simple preset recalls, voiceover/EAS insertion, or customized scripts

Serial Metadata Input

- 9-pin female D connector; 115.2 kbps; pinout per SMPTE 207M (RS-485); Designed to directly interface with Dolby serial metadata (SMPTE RDD6)

Ethernet

- Gigabit Ethernet via RJ45 supports included TCP/IP remote control application; HTTP server included for get/set control of all parameters

Front Panel Controls and Indicators

- Graphical OLED display

Power Requirements

- Dual power supplies, each rated at 100-264 VAC, 50/60Hz, auto-sensing, 150W max. total

Dimensions and Weight

- 1RU - 1.75"H x 19"W x 15.5"D (44 x 483 x 394 mm)
- Net weight: 9 lbs. (4 kg), approximate.

Shipping Dimensions and Weight

- 22"W x 20"D x 7"H (559 x 508 x 178 mm)
- Net weight: 15 lbs. (6.80 kg), approximate.

Environmental

- Fan cooled. Operating: 0 to 50 degrees C, non-operating -20 to 70 degrees C

Regulatory

North America: FCC and CE tested and compliant, power supply is UL approved.

Europe: Complies with the European Union Directive 2002/95/EC on the restriction of the use of certain hazardous substances in electrical and electronic equipment (RoHS), as amended by Commission Decisions 2005/618/EC, 2005/717/ EC, 2005/747/EC (RoHS Directive), and WEEE.

Warranty

For the latest Telos Alliance warranty, visit: telosalliance.com/warranty

Linear Acoustic® AERO.2000™

Loudness Management Platform for DTV

OVERVIEW

High quality, compliant audio with a front panel GUI.

Like AERO.100, AERO.2000 hosts one or two AEROMAX® processing instances in your choice of AMX5.1 (5.1+2+2), AMX2.0 (2+2+2), or AMX 5x2 (2+2+2+2+2) configurations (a minimum of one is required) plus upmixing/downmixing via our UPMAX®-II algorithm - but adds the flexibility and convenience of a full color display, menu navigation controls, and a headphone output to the front panel.

Optional Dolby® Digital Plus transcoding provides for decoding of Dolby Digital/Dolby Digital Plus content to PCM audio for loudness processing and encoding to Dolby Digital/Dolby Digital Plus for transmission.

I/O includes de-embedding and re-embedding of eight pairs of HD/SD-SDI audio and eight pairs of AES audio.

Support for SAP/DVS, EAS, local emergency audio, local voiceover, and optionally, Audio Description (warble tone) functionality is included. CrowdControl™ is standard for increased dialogue intelligibility.

ITU-R BS.1770-3 and EBU R-128 metering and logging (including True Peak) is provided for all program outputs. NfRemote software is included for remote configuration, control and metering over an Ethernet connection while a built-in HTTP server enables control of I/O, presets, and individual processing parameters using simple IP commands. Compensating video delay and dual redundant internal power supplies are standard.

FEATURES

- Linear Acoustic AEROMAX loudness and dynamics control
- UPMAX- II automatic upmixing and downmixing
- One or two processing instances in AMX5.1 (5.1+2+2), AMX2.0 (2+2+2), or AMX 5x2 (2+2+2+2+2) configuration
- CrowdControl for increased dialog intelligibility
- Support for SAP/DVS
- Optional Dolby® Digital Plus transcoding provides for decoding of Dolby Digital/Dolby Digital Plus content to PCM audio for loudness processing and encoding to Dolby Digital/Dolby Digital Plus for transmission
- Available Nielsen watermark encoding
- 8 audio pairs via HD/SD-SDI I/O with included video delay
- 8 audio pairs of AES I/O with reference input
- Dual auto-ranging power supplies
- Relay bypass of all I/O
- Extensive TCP/IP remote control and HTTP control
- Front panel color display, menu navigation controls, and headphone output
- Logging of loudness and True Peak data

IN DEPTH

AERO.2000 offers our industry standard, no-compromise AEROMAX television loudness control and UPMAX-II upmixing/downmixing algorithm in a 2RU package.

The extra 1RU of height compared to AERO.100 makes room for an additional four audio pairs of AES I/O. It also allows us to include a full color front panel display, menu navigation controls, and a headphone output to adjust or monitor processing locally, though our NfRemote TCP/IP remote control software can also be used to provide control over all system settings and processing parameters remotely.

AERO.2000 can host two processing instances, both with ITU- and EBU-compliant loudness metering and logging.

It also offers reporting of physical I/O details, power supply status, and environmental health. The remote application can delivers remote audio, up to 5.1 channels, so the user can audition signal quality anywhere link bandwidth permits. An HTTP server is also included for simple get/set control of all parameters and retrieval of status and logging information.

Constantly active logging captures 24 hour, 48 hour, and 7.5 day rolling weekly reports as well as specific time slots controlled by start/stop. Loudness measurements with multiple integration times as well as True Peak measurements are captured and available for download.

Failover bypass relays on all I/O maintain signal continuity and dual auto-ranging power supplies enable redundancy and worldwide compatibility.

Designed and built in the USA, AERO.2000 is a solid investment in performance and flexibility today and in the future.

SPECIFICATIONS

Processing

- Can host one or two instances of AEROMAX processing in your choice of AMX5.1 (5.1+2+2), AMX2.0 (2+2+2), or AMX 5x2 (2+2+2+2+2) configurations
- Dual UPMAX-II two-channel to 5.1 channel upmixers per instance plus main channel downmixing and automatic bypass of discrete content

Audio Encoding/Decoding

- Optional Dolby® Digital Plus transcoding provides for decoding of Dolby Digital/Dolby Digital Plus content to PCM audio for loudness processing and encoding to Dolby Digital/Dolby Digital Plus for transmission
- Nielsen watermark encoding

Reference

- 48kHz via AES DARS (or any AES signal applied to the Ref In connector), AES In 1, SDI, or from the internal 48kHz clock (standalone use only).

Sample Rate/Resolution/Frequency Response

- 48kHz, 24-bit, 20Hz to 20kHz below threshold

AES I/O

- Eight main inputs plus reference via 75-Ohm BNC female connectors. Eight main outputs plus encoder output. Eight additional channels of auxiliary digital I/O on DB-25 female connector. All digital inputs are 75 Ohm internally terminated, unbalanced. Signal levels per SMPTE 276M/ AES-3ID-2001.

HD/SD-SDI I/O

- Auto sensing HD/SD-SDI input up to 1080p/60/59.94/50Hz supported. De-embed up to eight audio pairs from applied SDI signal, process and/or encode, re-embed up to eight audio pairs. Supports SMPTE 2020A and B VANC metadata.

Headphone Output

- 1/4'' (6.35mm) front panel connector with volume control.

GPI/O

- Parallel GPI/O Parallel Control Port
- 25-pin female D connector, 0-5V TTL levels for 8 inputs and 8 outputs; controls simple preset recalls plus voiceover/EAS insertion

Serial Metadata Input

- 9-pin female D connector; 115.2 kbps; pinout per SMPTE 207M (RS-485); Designed to directly interface with Dolby serial metadata (SMPTE RDD6)

Ethernet

- Gigabit Ethernet via RJ45 supports included TCP/IP remote control application; HTTP server included for get/set control of all parameters.

Remote Control

- Windows®-compatible TCP/IP remote control Included application for full setup and control, ITU-R BS.1770 metering for all programs, encoder statistics, and return audio for remote monitoring (network speed permitting). HTTP server allows get/set control from PC Front Panel Controls and Indicators.

Front Panel Controls

- Rotary encoder and control keys plus color display and headphone output.

Power Requirements

- Dual redundant power supplies, each rated at 100-264 VAC, auto-sensing, 50/60 Hz, 175W each maximum

Dimensions and Weight

- 2RU: 3.50"H x 19"W x 17"D (89mm X 483mm X 432mm)
- Net weight: 13 lbs. (5.9 kg), approximate.

Shipping Dimensions and Weight

- 22"W x 20"D x 9"H (559 x 508 x229 mm)
- Net weight: 18 lbs. (8.2 kg), approximate.

Environmental

- Fan cooled. Operating: 0 to 50 degrees C, non-operating -20 to 70 degrees C

Regulatory

North America: FCC and CE tested and compliant, power supply is UL approved.

Europe: Complies with the European Union Directive 2002/95/EC on the restriction of the use of certain hazardous substances in electrical and electronic equipment (RoHS), as amended by Commission Decisions 2005/618/EC, 2005/717/ EC, 2005/747/EC (RoHS Directive), and WEEE.

Warranty

Standard 2-year limited parts and labor

Linear Acoustic® AERO.8000™

Loudness Management Platform for DTV

OVERVIEW

The future of Audio over IP loudness control - today.

AERO.8000 is a 1RU high-density loudness management platform featuring Livewire+ AES67 AoIP I/O for enterprise-wide audio access.

It can host up to eight AEROMAX® processing instances (a minimum of one is required) in your choice of AMX5.1 (5.1+2+2), AMX2.0 (2+2+2), or AMX 5x2 (2+2+2+2+2) configurations.

Upmixing is provided by the Hollywood-approved UPMAX®-II algorithm which delivers engaging 5.1-channel audio from two-channel sources. AutoMAX™ detection and switching between 2.0 and 5.1 surround sources makes full time 5.1 surround output easy to achieve with viewer-pleasing results.

All processing instances support SAP/DVS, LoRo/LtRt downmix, local audio insertion, and optional Audio Description (warble tone) functionality. ITU-R BS.1770-3 or EBU R128 meters are present on each audio program output. Linear Acoustic Crowd Control™ is standard and eliminates viewer complaints about "missing" or "hard to hear" dialogue.

Dolby® Digital Plus transcoding is optionally available for each AMX instance and provides for decoding of Dolby Digital/Dolby Digital Plus content to PCM audio for loudness processing and encoding to Dolby Digital/Dolby Digital Plus for transmission. Nielsen® watermark encoding is also available as an option for each AMX5.1 and AMX2.0 instance.

An included comprehensive TCP/IP remote control application provides control over all system settings, processing and coding parameters, per-channel signal presence, and loudness metering from any Windows PC. Multiple logs of LKFS loudness values, True Peak, and Dolby dialnorm and acmod values are provided for each program output. The remote control application also delivers remote audio monitoring so the user can audition signal quality anywhere link bandwidth permits. An http server is included for log retrieval and provides control of all parameters and retrieval of status via network commands.

Audio over IP (AoIP) connectivity via Livewire+™ AES67 enables enterprise-wide audio access. Audio from any video source can be placed on the AoIP network using a wide variety of Livewire+™ AES67 compliant devices such as Telos Alliance xNodes.

FEATURES

- Ideal for high density enterprise applications
- Up to eight processing instances
- I/O independence via Livewire+™ AES67 Audio over IP
- Linear Acoustic AEROMAX loudness/dynamics control
- UPMAX-II automatic upmixing and downmixing with AutoMAX 5.1 surround/2.0 detection and switching
- ITU-R BS.1770-3- and EBU R128-compliant metering
- Extensive remote GUI over IP control enables metering, control, and local audio monitoring (up to 5.1 channels)
- Internal logging of all program outputs in 24 hour, 48 hour, 7.5 day logs of 3s, 10s, 30s integration, TP and DN, and 1 hour/100ms logs
- Optional Dolby® Digital Plus transcoding provides for decoding of Dolby Digital/Dolby Digital Plus content to PCM audio for loudness processing and encoding to Dolby Digital/Dolby Digital Plus for transmission
- Nielsen watermark encoder option

IN DEPTH

Livewire®

Since 2003, broadcast facilities have been using Axia Livewire AoIP (audio over IP) technology to route hundreds of audio channels using standard Ethernet switches, cables, and other components. With thousands of devices on the air every day, reliability is proven.

Livewire is lightning fast, simple to implement, easy to manage, and now as Livewire+™ AES67, fully compliant with the AES67 standard and ready to incorporate new standards as they are developed. It significantly reduces the number of cables and wires needed to transport and share audio throughout the broadcast plant.

Nearly any audio source – analog, AES, or SDI, plus a wide variety of equipment from phone systems to codecs to satellite receivers – can be a part of the AoIP network.

SPECIFICATIONS

Processing

- Can host up to eight instances of AEROMAX processing in your choice of AMX5.1 (5.1+2+2), AMX2.0 (2+2+2), or AMX 5x2 (2+2+2+2+2) configurations
- Dual UPMAX-II two-channel to 5.1 channel upmixers per instance plus main channel downmixing and automatic bypass of discrete content.
- Includes extensive support for SAP/DVS, voiceover, downmix auto replacement of SAP/DVS and optional Audio Description (warble tone) functionality

Remote Control

- Dedicated TCP/IP remote control application provides extensive metering, control, system management and remote monitoring of one or many instances; http server included for log access and also for control of all parameters using commands over the network

Audio Encoding/Decoding

- Optional Dolby® Digital Plus transcoding provides for decoding of Dolby Digital/Dolby Digital Plus content to PCM audio for loudness processing and encoding to Dolby Digital/Dolby Digital Plus for transmission
- Nielsen watermark encoding

Linear Acoustic® AMS

Authoring and Monitoring System

OVERVIEW

Leading the way to ATSC 3.0 audio.

The Linear Acoustic AMS Authoring and Monitoring System is a comprehensive solution for real-time authoring, rendering, and monitoring of advanced audio programs for the ATSC 3.0 Digital Television System while simultaneously supporting 5.1- and 2-channel audio for ATSC 1.0 broadcasts.

FEATURES

- Real-time authoring, rendering, and monitoring of ATSC 3.0 advanced audio programs
- Utilizes Next Generation Audio (NGA) technologies including MPEG-H
- Legacy support for 5.1- and 2-channel ATSC 1.0 broadcasts
- Features Linear Acoustic APTO loudness processing
- Full-featured web interface for configuration, status, I/O routing, loudness processing, audio mixing, object panning, authoring configuration, and monitoring

IN DEPTH

The audio system in ATSC 3.0 provides listeners with a personalized, immersive audio experience but requires unique and specialized solutions to deliver on its potential. Using Next Generation Audio (NGA) technologies including MPEG-H, Linear Acoustic AMS simultaneously provides advanced audio for ATSC 3.0 broadcasts as well as 5.1- and 2-channel audio for ATSC 1.0 broadcasts.

AMS features Linear Acoustic APTO, the state-of-the-art loudness adaptation technology designed to carefully control audio levels in a way that preserves transients, sonic image, and the artistic intent of the source audio while ensuring loudness consistency and compliance for any target and regulatory standard.

Configuration, signal routing, loudness processing, mixing, object panning and authoring configuration are managed by an intuitive web interface. Individual audio elements can be combined with user-specified metadata to create immersive, personalized audio programs for the viewer which can all be monitored directly through the interface.

Customized loudness logs can be created, saved and retrieved for analysis and compliance purposes. Because engineers must be able to deal with a multitude of I/O configurations in the field, the Telos Alliance xNode family of Audio over IP devices allow Linear Acoustic AMS to handle analog, AES/EBU, and SDI I/O effortlessly with full Livewire+ AES67 AoIP support.

The Telos Alliance SDI xNode can de-embed 3G/HD/SD-SDI inputs and extract up to 16 channels of audio to the Livewire+ AES 67 port. This audio can be re-embedded into the SDI output stream with full video delay compensation for each SDI input to ensure A/V synchronization.

Up to 36 outputs can be configured including 16 channels of authored audio plus control channels, up to 12 channels for monitoring, and two additional sets of rendered broadcast outputs.

SPECIFICATIONS

Processing

- Linear Acoustic APTO loudness processing

Ethernet Remote Control

- Gigabit Ethernet supports included TCP/IP remote control application and HTTP server access

Power Requirements

- Dual redundant internal 95-240 VAC 50/60 Hz power supplies

Dimensions and Weight

- One rack unit - 1.75"H x 19"W x 18.5"D (44 x 483 x 470 mm) Net weight: 12 lbs (4 kg);

Environmental

- Fan cooled. Operating: 0 to 50 degrees C, non-operating –20 to 70 degrees C.

Regulatory

North America: FCC and CE tested and compliant, power supply is UL approved.

Europe: Complies with the European Union Directive 2002/95/EC on the restriction of the use of certain hazardous substances in electrical and electronic equipment (RoHS), as amended by Commission Decisions 2005/618/EC, 2005/717/ EC, 2005/747/EC (RoHS Directive), and WEEE.

Warranty

Standard 2-year limited parts and labor

Linear Acoustic® LA-5291

Audio Encoding, Transcoding, and Decoding for Live Dolby Atmos® Workflows

OVERVIEW

The Linear Acoustic LA-5291 Professional Audio Encoder allows you to create an exciting and immersive Dolby Atmos® experience via a Dolby Digital Plus JOC bitstream for production workflows, platforms, and delivery streams that do not require Dolby AC-4. The LA-5291 offers decoding, encoding, and transcoding to and from PCM and select Dolby® coded formats for up to 16 audio channels.

FEATURES

- Decoding from Dolby ED2 and Dolby E to PCM
- Transcoding from Dolby ED2 and Dolby E to Dolby Digital Plus and Dolby Digital Plus JOC
- Encoding PCM to Dolby Digital Plus and Dolby Digital Plus JOC
- 3Gb/s HD/SD-SDI and AES-3 I/O
- AES67 I/O in support of SMPTE ST 2110-30 and -31 workflows
- Optional Quad-Link 3Gb/s SDI I/O for supporting 4K video workflows, or optional MADI I/O (mutually exclusive options)
- Dual internal redundant auto-ranging power supplies
- Browser-based remote control

IN DEPTH

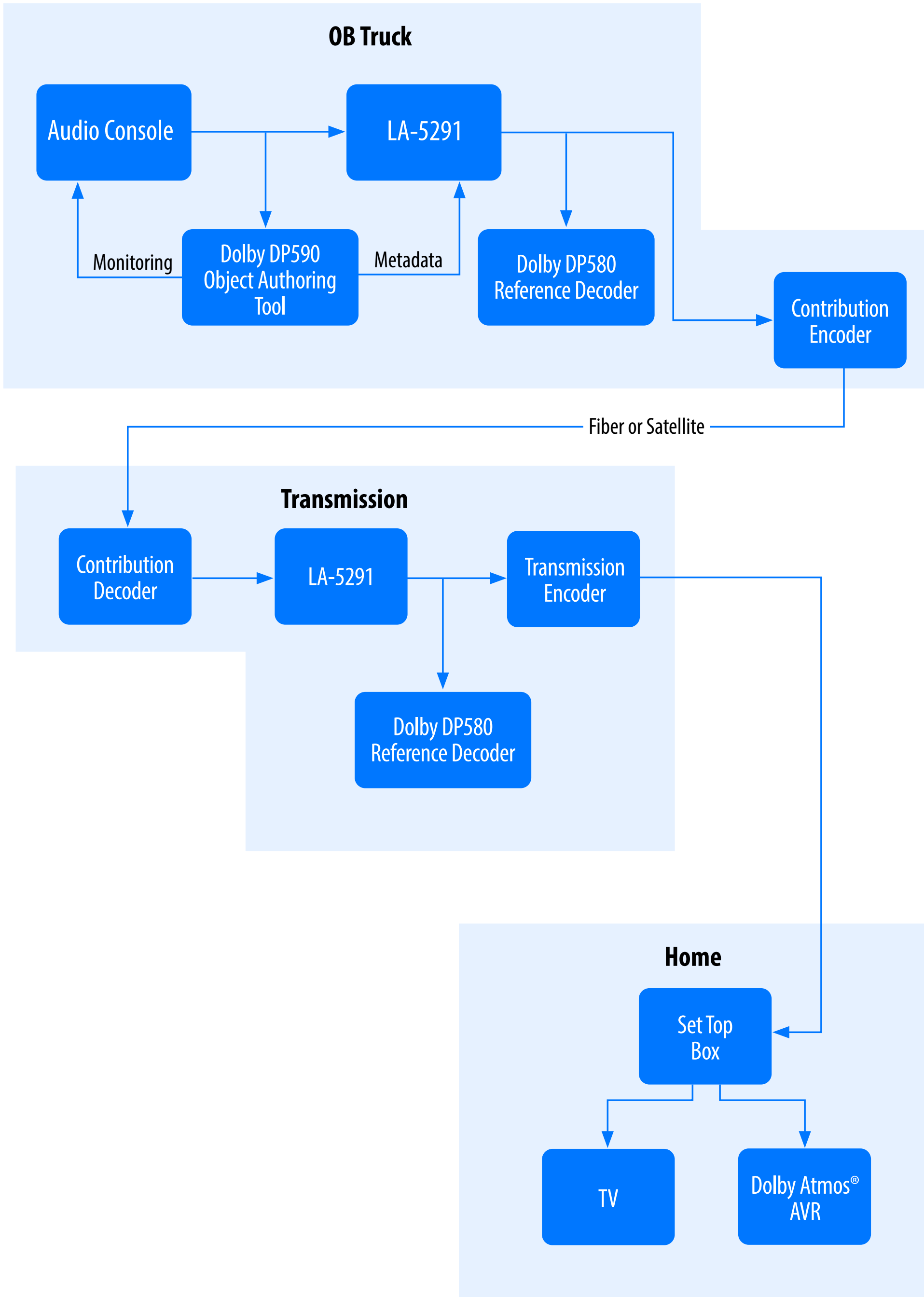
Supporting Today's Dolby Atmos Workflows

While emerging standards such as ATSC 3.0 continue to advance and provide improved immersive audio for OTA broadcast, the amount of cinematic and live content produced in Dolby Atmos continues to grow as well reaching consumers via Blu-ray Disc, satellite, cable, and streaming services. The workflow required to create, transport, and deliver that content to viewers at home can be complex, requiring the ability to work with PCM, Dolby ED2, Dolby E, Dolby Digital Plus, and Dolby Digital Plus JOC formats - tasks for which the LA-5291 was purpose-built.

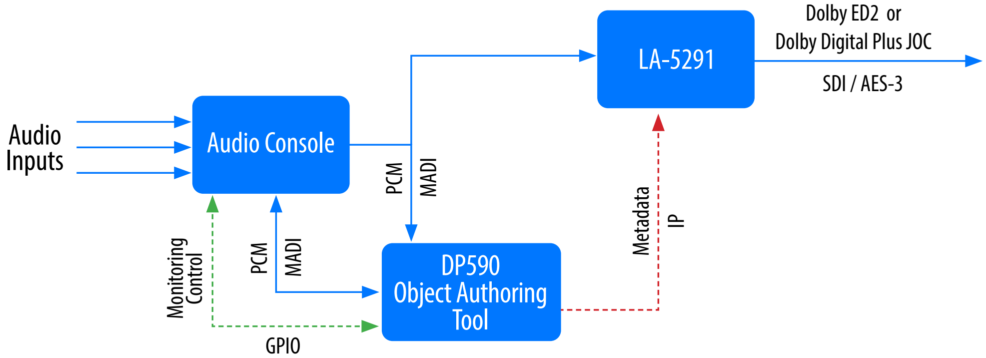
Equally at Home for Contribution and Distribution

The LA-5291 integrates well into remote and OB trucks for encoding multi-channel PCM audio into Dolby Digital Plus or Dolby Digital Plus JOC for distribution. Further downstream, it can transcode Dolby ED2 and Dolby E directly to Dolby Digital Plus and Dolby Digital Plus JOC, and can encode PCM to Dolby Digital Plus and Dolby Digital Plus JOC for final distribution.

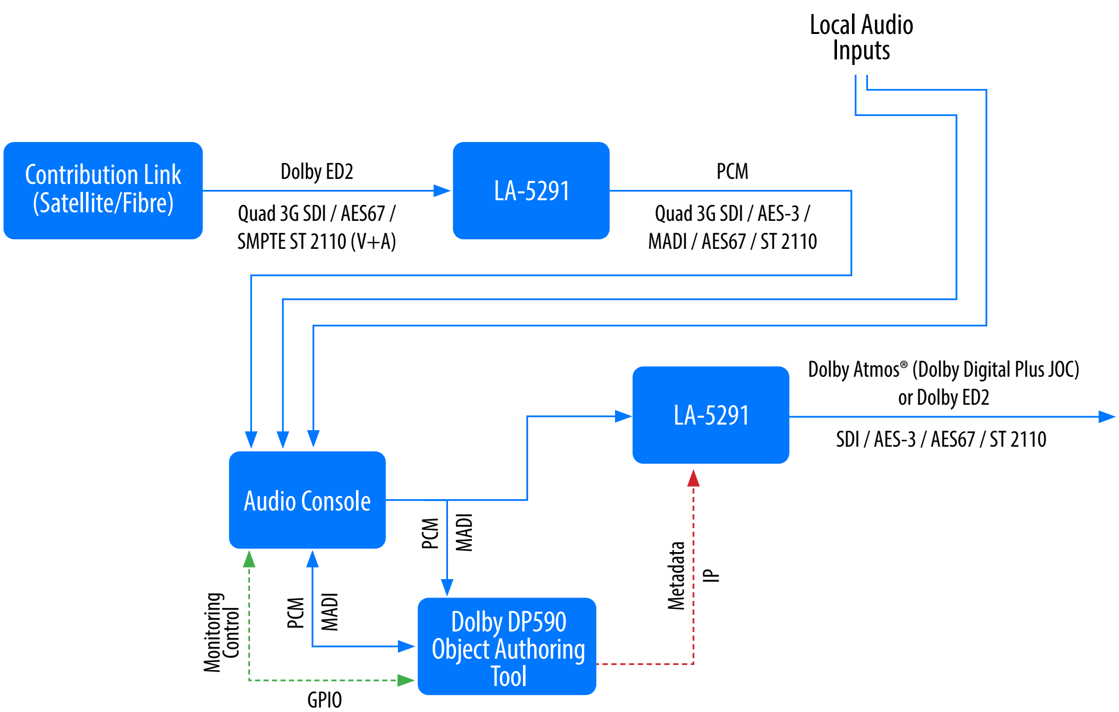
End-to-End Workflow



OB Truck/Production Workflow



Broadcast Center Decode / Insert / Encode



SPECIFICATIONS

Decoding

- Decodes Dolby ED2 and Dolby E to PCM

Transcoding

- Transcodes Dolby ED2 and Dolby E to Dolby Digital Plus and Dolby Digital Plus JOC

Encoding

- Encodes PCM to Dolby Digital Plus and Dolby Atmos (via Dolby Digital Plus JOC)

3G HD/SD-SDI I/O

- Two independent auto-sensing 3Gb/s HD/SD-SDI inputs (SMPTE ST 425-1, 292M, and 259M), up to 1080i/60/59.94/50Hz
- Optional Quad-Link 3Gb/s SDI for 4K workflows (mutually exclusive with MADI option)

AES-3 I/O

- 5 stereo pairs via 75 Ohm BNC unbalanced female connectors, internally terminated; signal levels per SMPTE 276M/AES-3ID-2001

AES67 I/O

- 16 channels of bi-directional AES67 I/O in support of SMPTE ST 2110-30 and -31 workflows

MADI I/O

- Optional MADI I/O supports up to 16 channels for processing
- Passthrough and shuffling for up to 64 channels
- I/O via coax or optical SFP socket (SFP sold separately)
- MADI option mutually exclusive with Quad-Link SDI option

Reference

- 48kHz reference via SDI, PTP, AES-3, internal clock (standalone use only), or MADI (when MADI option is installed)
- Vref

Sample Rate/Resolution/Frequency Response

- 48kHz, 24-bit, 20Hz – 20kHz

Ethernet

- Two Gigabit RJ-45 connections – one for AES67, one for networked remote control

Parallel GPI/O Control Port

- 15-pin female D connector, 0-5V TTL levels, 5 GPI/O inputs, 5 GPI/O outputs

Power

- Dual internal redundant auto-ranging power supplies
- 95-240 VAC, 50/60 Hz, 100W maximum total

Dimensions and Weight

- 19" W x 15.5" D x 1.75" H (approximately 48.2 x 39.4 x4.5 cm)
- Net weight: Approximately 9.0 lbs (4.08 kg)
- Shipping weight: Approximately 12.0 lbs (5.44 kg)

Regulatory

- North America – FCC and CE tested and compliant with UL-approved power supplies
- Europe – Complies with European Union Directive 2002/95/EC on the restriction of use of certain hazardous substances in electrical and electronic equipment (RoHS), as amended by Commission Decisions 2005/618/EC, 2005/717/EC, 2005/747/EC (RoHS directive), and WEEE

Warranty

- Standard Telos Alliance 2-year limited parts and labor

Linear Acoustic® UPMAX® ISC

Immersive Soundfield Controller

OVERVIEW

UPMAX ISC builds upon the Hollywood-approved and industry-trusted line of upmixing products from Linear Acoustic, and carries it forward to meet the additional demands introduced by the immersive audio capabilities of Next Generation Audio systems.

Employing a completely new upmixing algorithm, UPMAX ISC can upmix 2-channel, 3-channel, 5.1-channel, and 7.1-channel audio to immersive audio formats including 5.1.4 and 7.1.4. Upmixing to legacy formats including 5.1-channel and 7.1-channel audio is also supported, while content presented in native immersive formats is automatically recognized and passed through without processing. Compensating video delay is included to ensure perfect A/V sync regardless of input format or operating mode.

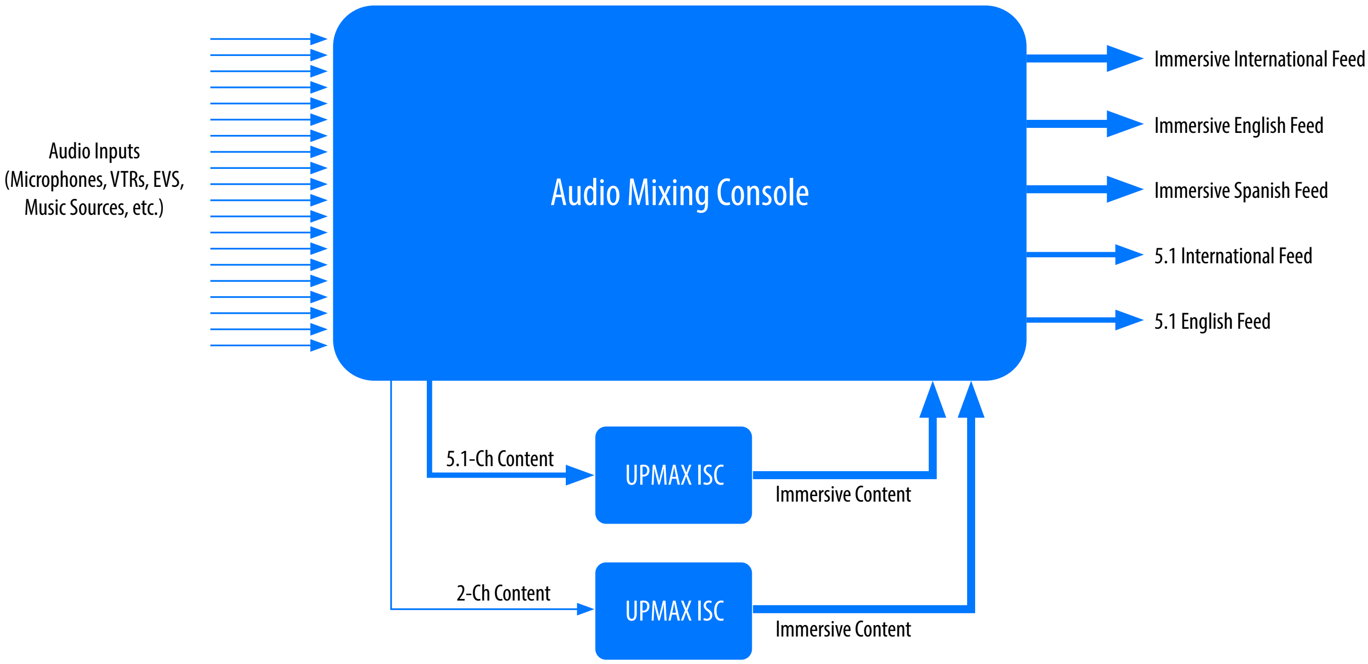
For live events and broadcasts, UPMAX ISC can be inserted on a mix bus or placed at the output of the mixing console in the OB truck at the venue for upmixing sources including music, effects, and legacy content not natively presented in an immersive format.

UPMAX ISC is equally at home in the airchain at the Network Operations Center to ensure all programming - including commercial and interstitial content - is delivered to the viewer as a consistent immersive experience.

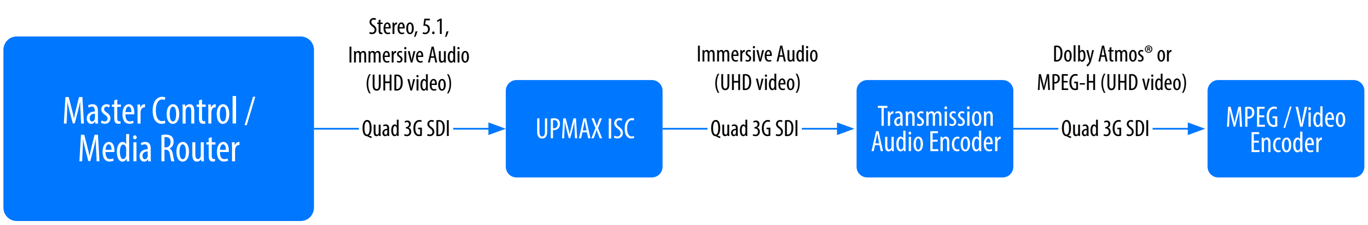
I/O includes auto-sensing 3Gb/s HD/SD-SDI, AES-3, and AES67 to natively support SMPTE ST 2110-30 workflows.

UPMAX ISC includes a 5-button front panel navigation cluster and color LCD display for basic setup and control, plus the convenience of a web-based GUI for full control and monitoring capabilities. Dual auto-ranging internal power supplies provide redundancy and worldwide compatibility.

ISC Workflow 1



ISC Workflow 2



FEATURES

- Latest Linear Acoustic UPMAX algorithm
- Upmixing from 2-channel, 3-channel, 5.1-channel, and 7.1-channel to immersive 5.1.4 and 7.1.4 formats
- Upmixing to legacy formats including 5.1 and 7.1
- 3b/s HD/SD-SDI, AES-3, and AES67 I/O to support SMPTE ST 2110-30 workflows
- Optional Quad-Link SDI I/O for 4K video workflows or MADI I/O (mutually exclusive options)
- Dual internal redundant auto-ranging power supplies
- Browser-based remote control

SPECIFICATIONS

Processing

- Next-generation Linear Acoustic UPMAX algorithm; upmixing from 2-channel, 3-channel, 5.1-channel, and 7.1-channel to 5.1.4 and 7.1.4

Reference

- 48kHz reference via SDI, PTP/AES67, AES-3, internal clock, or MADI (with optional MADI I/O)

Sample Rate/Resolution/Frequency Response

- 48kHz, 24-bit, 20Hz – 20kHz (passthrough); PCM audio only

AES-3 I/O

- Five two-channel inputs/outputs via 75 Ohm BNC unbalanced female connectors, internally terminated; signal levels per SMPTE 276M/AES-3ID-2001

SDI I/O

- Two auto-sensing 3Gb/s HD/SD-SDI inputs (SMPTE ST425/292M/259M), up to 1080i/60/59.94/50Hz, with de-embedding for 8 audio pairs; re-embedding for 8 audio pairs via one SDI output
- Optional Quad-Link SDI for 4K workflows (mutually exclusive with MADI option)

AES67 I/O

- 16 channels of bi-directional AES67 I/O in support of SMPTE ST 2110-30

Parallel GPI/O Control Port

- 15-pin female D connector, 0-5V TTL levels, 5 GPI/O inputs, 5 GPI/O outputs

MADI I/O

- Optional MADI I/O supports up to 16 channels for processing
- Passthrough and shuffling for up to 64 channels
- I/O via coax or optical SFP socket (SFP sold separately)
- MADI option mutually exclusive with Quad-Link SDI option

Ethernet Connection

- Two independent Gigabit Ethernet ports allows web-based remote access/control and AES67 I/O

Front Panel Display and Controls

- 5-button navigation cluster, color LCD display, LEDs for power supply, status, and sync

Power

- Dual internal power supplies, each rated at 100-264VAC, 50/60Hz, auto-sensing, 100W maximum total

Dimensions and Weight

- 1RU – 1.75"H x 19"W x 15.5"D (44 x 483 x 394 mm)
- Approximate net weight 9 lbs (4 kg)
- Approximate shipping weight 12 lbs (5.4 kg)

Environmental

- Fan-cooled; operating temperature 32 to 122 degrees F (0 – 50 degrees C); non-operating temperature -4 to 158 degrees F (-20 – 70 degrees C)

Regulatory

- North America – FCC and CE tested and compliant; power supplies are UL-approved

- Europe – Complies with European Union Directive 2002/95/ EC on the restriction of the use of certain hazardous substances in electrical and electronic equipment (RoHS), as amended by Commission Decisions 2005/618/EC, 2005/717/ EC, 2005/747/EC (RoHS Directive), and WEEE

Warranty

- Standard 2-year limited parts and warranty

Linear Acoustic® LA-5300

The Complete Audio Processor for ATSC 3.0

OVERVIEW

The Linear Acoustic LA-5300 Broadcast Audio Processor provides everything broadcasters need to be ready for NEXTGEN TV/ATSC 3.0 audio in a single, compact, integrated package, including loudness control, upmixing, encoding, transcoding, audience measurement watermarking, bitstream analysis for up to four simultaneous real-time program streams, and a confidence monitor output.

FEATURES

- Dolby® AC-4 encoding from PCM
- Transcoding from Dolby Digital and Dolby Digital Plus to AC-4
- AC-4 decoding for watermarking and bitstream analysis and monitoring
- Linear Acoustic UPMAX® ISC upmixing
- Optional Verance or Nielsen watermarking
- Dolby Real-Time Loudness Leveler when encoding to AC-4
- Dual 3Gb/s HD/SD-SDI, 5 pairs of AES-3 I/O, and AES67 I/O in support of SMPTE ST 2110-30 and -31 workflows
- Optional Quad-Link 3Gb/s SDI I/O for supporting 4K workflows or MADI I/O (mutually exclusive)
- Dual 1000BaseT Ethernet connections (AES67 and control)
- SNMP alarm and status reporting
- Web-based user interface provides comprehensive setup, configuration, routing, control, and metering

IN DEPTH

A Complete ATSC 3.0/NEXTGEN TV Audio Solution

The roll-out of ATSC 3.0 - better known to consumers as NEXTGEN TV - brings with it a unique set of requirements, challenges, and opportunities for television broadcasters. The Linear Acoustic LA-5300 provides a single solution for loudness control, upmixing, decoding AC-4 for audience measurement watermark insertion, bitstream analysis, and monitoring, transcoding of Dolby Digital and Dolby Digital Plus to AC-4, and Dolby AC-4 encoding. The ability to handle up to 4 programs means one LA-5300 can provide unique encoded streams for the main program audio, SAP, and video descriptive services—all in a single 1RU solution. A decode and monitor-only version is also available which omits upmixing, watermarking, and AC-4 encoding.

I/O for Any Facility

The LA-5300 comes standard with 5 pairs of AES-3 I/O, dual 3G SDI I/O, and AES67 I/O to support SMPTE ST 2110-30 and -31 workflows. Options include Quad-Link 3G SDI for facilities utilizing 4K workflows, or MADI I/O. Two Gigabit Ethernet ports are provided, one for AES67, and one for remote control via the web-based GUI.

Ready for Today, Ready for Tomorrow

The LA-5300 meets the immediate requirements for NEXTGEN TV/ATSC 3.0 audio right out of the box, with channel-based AC-4 encoding, loudness control, and upmixing for multiple programs. As ATSC 3.0 adoption grows, support for its additional features and benefits such as immersive and object-based audio, interactive consumer control, personalized audio, and multiple presentations within a single stream will be incorporated into the LA-5300 via software updates.

SPECIFICATIONS

Processing

- Processing for up to 4 independent program streams
- Linear Acoustic UPMAX ISC upmixing
- Dolby Real-Time Loudness Leveler (RTLL) when encoding to AC-4
- Optional Nielsen and Verance audience measurement watermarking

Decoding

- Decodes Dolby AC-4 at the input for watermarking and monitoring with AC-4 passthrough ability

Transcoding

- Transcodes Dolby Digital and Dolby Digital Plus to Dolby AC-4

Encoding

- Encodes to Dolby AC-4

Watermarking

- Optional Nielsen and Verance audience measurement watermarking

Bitstream Analysis and Monitoring

- Bitstream analysis of all streams simultaneously plus a confidence monitoring output for one stream at a time

HD/SD-SDI I/O

- Two independent auto-sensing 3Gb/s HD/SD-SDI inputs (SMPTE ST 425-1, 292M, and 259M) up to 1080p/60/59.94/50Hz, each with de-embedding/re-embedding for up to 8 audio pairs
- Optional Quad-Link 3Gb/s SDI for 4k workflows (mutually exclusive with MADI option)

AES-3 I/O

- 5 inputs/outputs via 75 Ohm BNC unbalanced female connectors, internally terminated; signal levels per SMPTE 276M/AES-3ID-2001

AES67 I/O

- 16 channels of bi-directional AES67 I/O in support of SMPTE ST 2110-30 and -31 workflows

MADI I/O

- Optional MADI I/O supports up to 32 channels for processing (mutually exclusive with Quad-Link SDI option)
- Passthrough and shuffling for up to 64 channels
- I/O via coax or optical SFP socket (SFP sold separately)

Reference

- 48kHz reference via SDI, PTP, AES-3, internal clock (standalone use only), or MADI (when MADI option is installed)
- Vref

Sample Rate/Resolution/Frequency Response

- 48kHz, 24-bit, 20Hz – 20kHz

Ethernet

- Two Gigabit RJ-45 connections – one for AES67, one for networked remote control

Parallel GPI/O Control Port

- 15-pin female D connector, 0-5V TTL levels, 5 GPI/O inputs, 5 GPI/O outputs.

Power

- Dual internal redundant auto-ranging power supplies
- 95-240 VAC, 50/60 Hz, 100W maximum total

Dimensions and Weight

- 19" W x 15.5" D x 1.75" H (approximately 48.2 x 39.4 x4.5 cm)
- Net weight: Approximately 9.0 lbs (4.08 kg)
- Shipping weight: Approximately 12.0 lbs (5.44 kg)

Regulatory

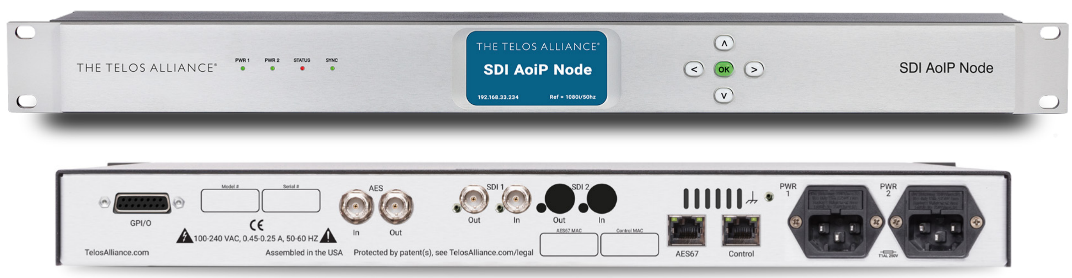
- North America – FCC and CE tested and compliant with UL-approved power supplies
- Europe – Complies with European Union Directive 2011/65/EU of the European Parliament and of the Council of 8 June 2011 on the restriction of the use of certain hazardous substances in electrical and electronic equipment (RoHS)

Warranty

- Standard Telos Alliance 2-year limited parts and labor

Telos Alliance® SDI AoIP Node

Bridging the Present and Future of TV Technology



OVERVIEW

As the successor to the SDI xNode, the Telos Alliance SDI AoIP Node continues our commitment to bring the power and flexibility of Audio over IP to broadcast television by de-embedding and converting up to 8 pairs of audio from two SDI inputs to AES67. Audio can then be shared on the network, processed for loudness compliance, and ultimately re-embedded into two SDI output streams.

FEATURES

- Provides de-embedding, routing, and re-embedding of up to 8 audio pairs through two SDI inputs and outputs
- 3Gb/s HD/SD-SDI supports UHD applications
- AES67 supports SMPTE ST 2110-30 workflows
- Compensating video delay maintains proper A/V sync
- Provides SRCs for all SDI output channels
- Full-width 1RU hardware with dual internal redundant auto-ranging power supplies

IN DEPTH

Bringing the Power of IP to Television

Radio discovered the myriad benefits of AoIP nearly two decades ago, and hasn't looked back since. Television broadcasters are now realizing the benefits of networked audio and interoperability between manufacturers and products.

Flexible Routing for Two SDI Signals

The SDI AoIP Node offers two independent 3Gb/s HD/SD-SDI inputs and outputs. Up to 8 audio pairs from either or both SDI inputs can be de-embedded and converted to AES67 and become available anywhere on the network for monitoring, distribution, or loudness control via a Linear Acoustic® AERO.8000 Processing Engine. Up to 8 audio pairs can then be re-embedded (and pair shuffled if desired) to two independent SDI outputs.

Ready Today, Ready for the Future

The SDI AoIP Node supports 3G video standards to seamlessly integrate into UHD facilities. Installations built using SMPTE ST 2110 workflows will appreciate its AES67 I/O which provides native support for SMPTE ST 2110-30.

User-Friendly Setup and Configuration

Despite its powerful features and signal routing capabilities, SDI AoIP Node is easy to configure. Its user-friendly web-based UI is device, OS, and browser-agnostic, and can be used on any desktop or laptop computer, tablet, or smartphone.

SPECIFICATIONS

HD/SD-SDI I/O

- Two independent auto-sensing 3Gb/s HD/SD-SDI inputs (SMPTE 292M, 259M, and 424M) with de-embedding for up to 8 audio pairs
- De-embedded audio is converted to AES67 and can be routed anywhere on the network
- Up to 8 audio pairs can be re-embedded to two independent HD/SD-SDI outputs

AES67

- Fully AES67 compliant and supports SMPTE ST 2110-30 workflows
- Supports 2-ch streams with a 1 ms packet time (ptime)

Reference

- User selectable reference clock
- Reference clock options include PTP (AES67), SDI, Internal 48 KHz

Ethernet

- Two Gigabit RJ-45 connections – one for AES67, one for networked remote control

Power

- Dual internal redundant auto-ranging power supplies
- 95-240 VAC, 50/60 Hz, 30W maximum

Dimensions and Weight

- 19" W x 9" D x 1.75" H (approximately 48.2 x 22.8 x4.5 cm)
- Net weight: Approximately 6.0 lbs (2.72 kg)

Regulatory

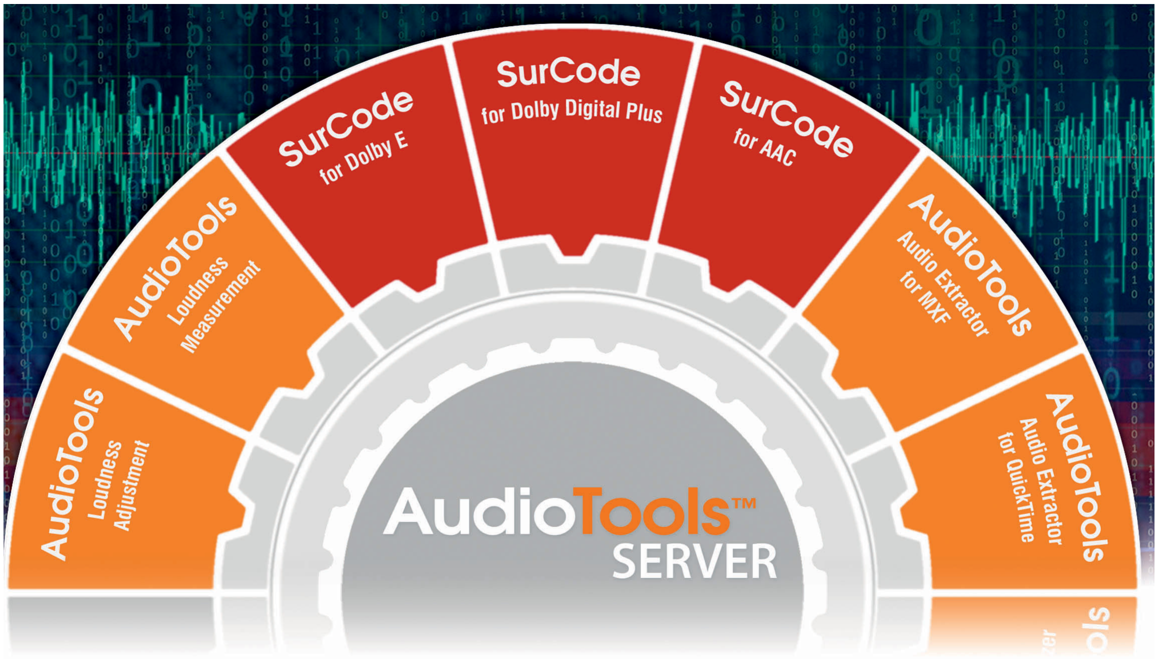
- North America – FCC and CE tested and compliant with UL-approved power supplies
- Europe – Complies with European Union Directive 2002/95/EC on the restriction of use of certain hazardous substances in electrical and electronic equipment (RoHS), as amended by Commission Decisions 2005/618/EC, 2005/717/EC, 2005/747/EC (RoHS directive), and WEEE

Warranty

- Standard Telos Alliance 2-year limited parts and labor

Minnetonka Audio® AudioTools™ Server

Quality-First Audio Automation



OVERVIEW

AudioTools Server is a collection of enterprise-ready audio solutions designed to add file-based expertise to automated environments. The AudioTools Server family represents decades of expertise encapsulated in flexible, focused packages of audio specialization.

AudioTools Server automates the most sophisticated audio tasks and offers a wide variety of processing, specifically created for use in cable, satellite, terrestrial and IPTV, radio, and post production facilities. As a pure software platform running on commodity hardware, including VM or cloud deployment, AudioTools Server is flexible and customizable, allowing for new workflows as requirements inevitably change.

- State-of-the-art audio processing
- Customized, efficient file-based workflows
- Unrivaled loudness tools
- Compliance to broadcast standards
- Interoperability with major environments
- Modular scalable platform

AudioTools Server - Leader in Loudness

Loudness Control is a significant use case that employs Linear Acoustic APTO™ loudness processing. Our loudness normalization processes are designed to fully preserve the existing audio and only apply a gain change combined with optional peak limiting. There are other use cases that require changing the dynamic range and more complex parameters of the audio content, such as dialog intelligibility. Advanced Loudness Adaptation is a collection of intelligent loudness profiles designed to adapt a theatrical audio mix for broadcast to create the best possible audio experience for all modern platforms such as OTT/web, mobile/handheld and VOD/SVOD.

FEATURES

Use Cases

Loudness Measurement & Adjustment

- State-of-the-art loudness control based on international standards & practices. Advanced Loudness Adaptation profiles for improved and compliant dialog intelligibility in high-dynamic range content. Netflix Compliance Profiles are available for both Loudness Adjustment and Advanced Loudness Adaptation.

Audio Adaptation

- Automated adaptation of audio content to specific output specifications, including: upmix, downmix, channel management, and frame rate conversion.

Dolby Automation

- Automated Dolby encoding and decoding including metadata handling. Dolby E quality control, with optional correction.

Quality Control

- Audio specific quality control of audio files or container formats, including channel assignment detection and correlation check.

IN-DEPTH

The Server Architecture

The AudioTools Server communicates directly with modules to configure the overall system for the required functionality. AudioTools Server, as a system, can then interface directly with your existing DAM, MAM, CMS or DBM workflow manager.

Modular Design

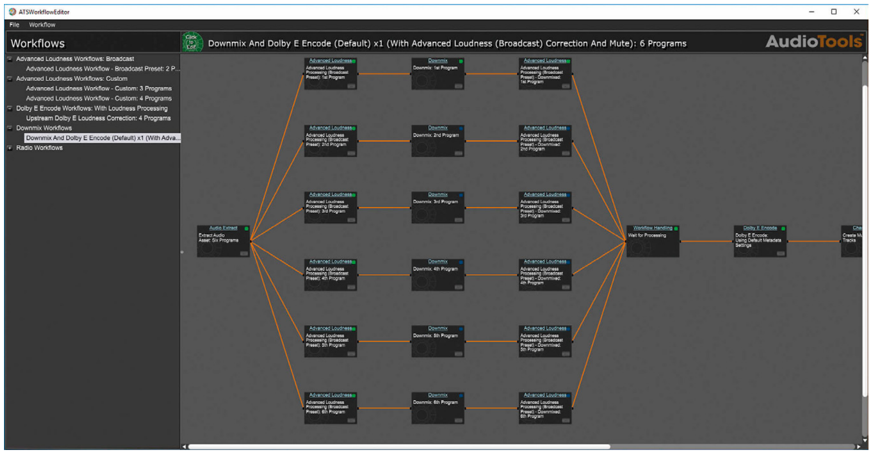
While a modular architecture allows AudioTools Server to always be "state-of-the-art", the real power of this approach is using workflows to combine module functions in a sequential or highly conditional profile. AudioTools Server is also a VST plug-in host, allowing for 3rd party plug-ins to be used as part of an overall AudioTools Server workflow.

Enhanced Workflows with AudioTools Server

AudioTools Workflow Control is the command and control option for AudioTools Server that enables standalone operation along with support for threaded multiple concurrent processes, load balancing and dynamic reconfiguration of workflows on the fly. AudioTools Server can deploy floating licenses through a license server, offering a scalable system for small businesses or enterprise class facilities.

AudioTools Server is internally driven from tailored XML profiles. For an operator, preset workflows can easily be called up and edited through the AudioTools Operator app or built on the fly and submitted with AudioTools WorkflowEditor, while Queue Control provides an overview with detailed access to all running processes.

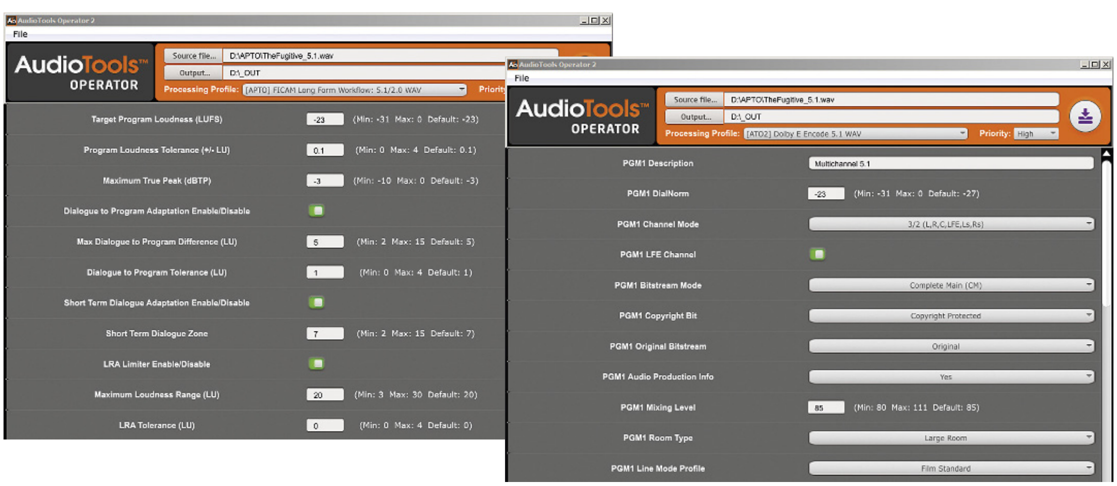
AudioTools WorkflowEditor



AudioTools WorkflowEditor is a flexible tool for setting up and configuring AudioTools Server workflows. The software comes pre-installed with a comprehensive selection of over 1000 workflow templates, covering most common workflow types and featuring all aspects of AudioTools Server processing. Each template is displayed and can be configured through the software's user interface, making it easy and fast for operators to conform workflows to meet their requirements. Preset management tools allow you to save and recall your conformed workflows in a user defined library for quick and simple access. Jobs and watchfolders can be submitted to the server for processing directly from the WorkflowEditor, or workflows can be exported for submission via third party tools or web services integration.

AudioTools OPERATOR

Operators get assistance and access on parameters through a simple, custom user interface.
Example implementations:
Advanced Loudness Adaptation (left) and Dolby E encoder (right)



AudioTools WebClient

The WebClient allows you to monitor AudioTools Server processes via the network. It is possible to check logs, results, and progress from any machine or mobile device with access to the network of AudioTools Server.

Integration

AudioTools Server supports manual job submission, hot folders, and a full web services API, guaranteeing that it fits into any environment, on any integration level.

Integration Partners

Arvato, Aspera, Aveco, AVID, Cinnafilm, Dalet, Dolby, Evertz, Geminisoft, Harmonic, IBM, Kantar Media/Civolution, Root 6, Sony, Tedial, Telestream, Vector 3, VIZRT, and others

SPECIFICATIONS

Content & Formats

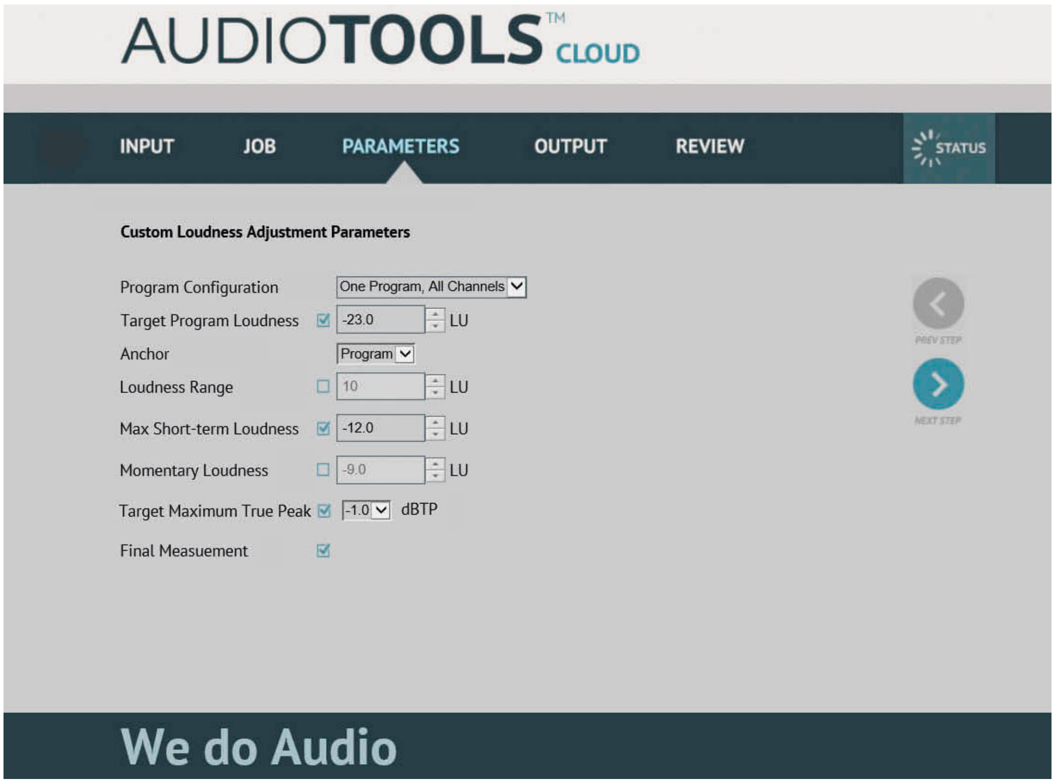
AudioTools Server is a complete solution for managing and processing audio formats. With audio at the center of a complex audio ecosystem, a payload may contain "containerized" video assets, transport streams, Dolby–encoded content, and metadata, requiring that each layer of the asset must be carefully handled.

Formats:

Linear PCM Audio: WAV, BWF, AIFF, RF64
Codecs: Dolby E, Dolby Digital, Dolby Digital Plus, Dolby Pro Logic II, mp2, mp3, MPEG-4, HE-AAC, AAC
Container: MXF, QuickTime™, LXF, GXF, selected Transport Streams

Minnetonka Audio® AudioTools™ Cloud

Quality-First On-Demand Audio Processing



OVERVIEW

AudioTools Cloud™ is an advanced audio processing solution for audio, video, and broadcast professionals that provides loudness control, encoding, decoding, channel management, frame rate conversion, quality control, and container management from an easy-to-use user interface designed for the Amazon AWS Marketplace.

AudioTools Cloud is based on Minnetonka's AudioTools™ Server – a platform that delivers interoperable, scalable, file-based audio automation. AudioTools Server has become the number one enterprise level platform for automated and unattended file-based audio processing and has helped broadcasters add Loudness Management processes to their existing video-centric file-based environments.

From the occasional job, to expanding throughput for higher volume workloads, AudioTools Cloud brings proven processes to the cloud, allowing on-demand, case-by-case use (OPEX) vs larger CAPEX sized projects. Businesses only pay for the infrastructure they need, when they need it.

FEATURES

- Professional file-based audio processing
- Extensive selection of job types
- Intuitive user interface enables complete control over every job parameter
- Support for global loudness compliance standards
- SurCode encoding and decoding technologies included
- 3 flexible configurations

Submitting Jobs to AudioTools Cloud

AudioTools Cloud ON-DEMAND is available as different Amazon EC2 instances. Each instance will launch the AudioTools Cloud Web Client Interface. Users can choose to upload content to Amazon S3 (Simple Storage Service). After the upload is complete, select your input buckets, output buckets, and audio processing parameters.

Users pay for the EC2 instance, storage, and AudioTools Cloud software.

IN-DEPTH

AudioTools Cloud ON-DEMAND

AudioTools Cloud ON-DEMAND is designed for the Amazon AWS Market Place and offers ready-to-use audio processing profiles. A simple click-through browser based “configurator” is used to assign input and output locations and file types, or adjust loudness target levels for different specifications. These configurations can be downloaded and stored locally as templates for future use.

Self-service: Users choose what they want and when they want it.
Scalable: Users can choose how much capacity they want to ramp up if necessary.

AudioTools Cloud BYOL - Bring Your Own License

AudioTools Server users can benefit from a cloud based deployment by adding AudioTools Server instances to any public or private cloud environment. An AudioTools LicenseServer will allow for floating licenses across all AudioTools Server instances in the cloud, on premise or any VM or datacenter deployment.

The AudioTools Server system in the cloud will ping the LicenseServer for available licenses, benefiting from cloud based flexible scalability within the existing license pool. AudioTools Cloud BYOL is perfect for users that want the flexible scalability of cloud based processing, combined with defined processing profiles and licenses. This is also the ideal strategy for smoothly and gradually moving work from an on premise installation to cloud based processes.

AudioTools Cloud Node

AudioTools Server v4 will offer an on-demand Amazon Cloud-based AudioTools Server instance in addition to an existing AudioTools Server installation. The cloud-based service is being added as a processing node for AudioTools Server. AudioTools Workflow Control can then use the Cloud Node to add more processing resources to a local system. In a load-balanced environment, if a local system is not licensed for a specific ATS module, Workflow Control can assign those tasks to the Cloud Node, which by default includes all possible modules and licenses for AudioTools Server. **AudioTools Server Cloud Node** is the perfect add-on for flexible scalability and additional licenses on a project-by-project basis.

SPECIFICATIONS

AUDIO PROCESSING OPTIONS

Frame Rate Conversion

- Film - NTSC - PAL
- Pitch Shift
- Time Stretch
- Sample Rate Conversion

Loudness Control

- EBU R 128 Loudness Adjustment
- CALM A/85 Loudness Adjustment
- AS-11 UK DPP
- Loudness Measurement
- Linear Acoustic APTO audio processing
- Advanced Loudness Adaptation

Quality Control

- Channel Detection
- Program Correlation Check
- Silence Detection
- Data Corruption Detection

Audio Codecs

- MP2/MP3 Encoding and Decoding
- AAC Encoding and Decoding

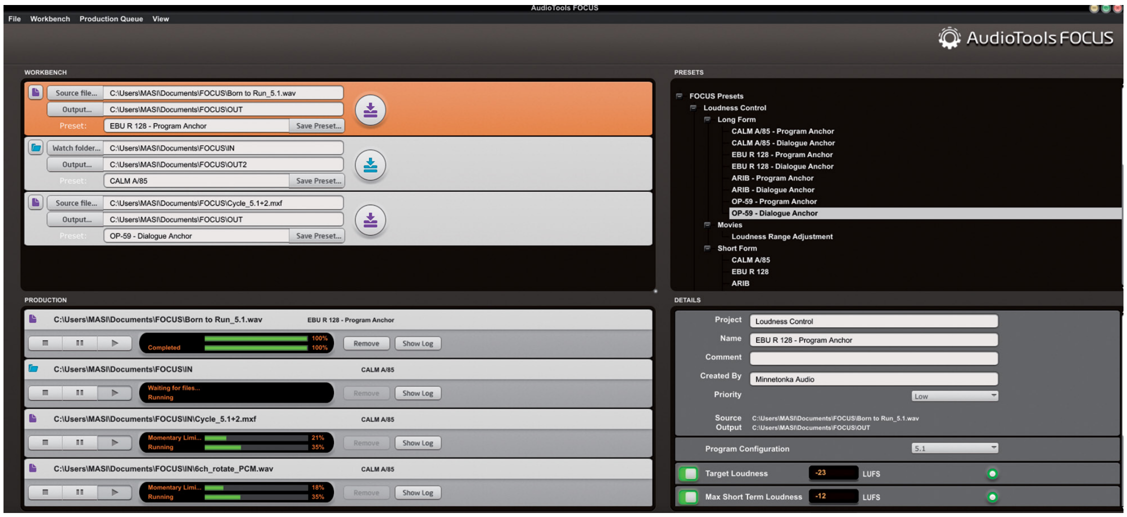
Container Management

- Extract audio from MXF, LXF, GXF, QuickTime™, and Transport Streams
- Re-wrap audio to MXF, LXF, GXF, QuickTime™, and Transport Streams

Channel Management

- Upmix with Linear Acoustic UPMAX™
- Downmix
- Channel Swapping
- Channel Replacement
- Channel Mixing

Minnetonka Audio® AudioTools™ FOCUS for Loudness Control



OVERVIEW

Loudness in a Video World

AudioTools FOCUS for Loudness Control is an easy to use, standalone Windows application that takes the complexity out of the audio processing equation. Proven AudioTools presets conforming to every loudness standard are built in and ready to use or modify. With all the standards, file formats and deliverables needed today, loudness management adds one more layer of intricacy to an already complex ecosystem. In a video–centric world, we focus on handling the complexity and dependencies of automating loudness control...AudioTools FOCUS brings audio control into focus.

FEATURES

AudioTools FOCUS for Loudness Control is intended for:

- Sole proprietors needing painless yet high quality logging & adjustment
- Post houses & networks needing container handling
- Any facility that doesn't have deep automation in place

Hit All Targets

AudioTools FOCUS for Loudness Control makes quality loudness management painless, taking the complexity of files, containers and processing, and making it all available in an easy to use, preset–driven application. A unique AudioTools feature is its ability to hit all loudness targets, not just one or two. With AudioTools FOCUS' logic–driven approach, you can simultaneously comply with Program Loudness, Loudness Range, and True–Peak, without exceeding your Maximum Momentary or Maximum Short Term loudness thresholds..

Intricacy Unraveled

Simply type in your target parameter changes to the factory presets, and let FOCUS handle the processing complexity:

- Start with a preset, & type in your target requirements for each delivery spec
- Define your program configuration
- Set your file I/O – Watched Folders or Files
- Submit!

AudioTools FOCUS provides:

- Loudness managed while preserving artistic intent
- Complete compliance with North American (CALM) & international loudness control standards including:
  EBU R 128, ARIB TR-B32 & OP-59
- Linear Acoustic APTO™ for Loudness Range processing
- File-based Omnia.9 processing for podcasts or on-demand (FOCUS Server for Radio)
- Hot Folder or File-driven processing
- MXF & QuickTime container handling – extract/process/rewrap
- Loudness adjustment or pure logging
- File type auto-detection
- Clearly labeled, proven presets right out of the box

IN-DEPTH

AudioTools FOCUS – An Affordable, Quality-First Solution For Loudness Control

Not only are there many variations of video file formats, but a quality loudness management process itself is complex. AudioTools FOCUS for Loudness Control provides:

- Program Loudness adjustments with combined True–Peak limiting
- Program Loudness combined with Maximum Short Term Loudness control for short for & interstitials
- Active & passive" Max. Short Term Loudness combined with Max Momentary limiting
- Pure gain adjustment versus dynamic processing

File-based In Mind

The world is moving toward IT-dominated, file–based workflows, and audio is only a part of the overall video picture. AudioTools FOCUS extracts the audio, processes that essence, preserves any audio metadata and rewraps the audio back into the container without modifying the video. This streamlines your workflow while ensuring downstream compatibility. The ITU-R BS.1770-3 Program Loudness standard was defined as file-based. AudioTools FOCUS directly answers that need with a standalone, software solution for loudness control automation.

Simplicity...It's In The Details

Unlike hardware investments that cannot keep up with changing business strategy, delivery/platform requirements and government mandates, AudioTools FOCUS for Loudness Control is a pure software solution.

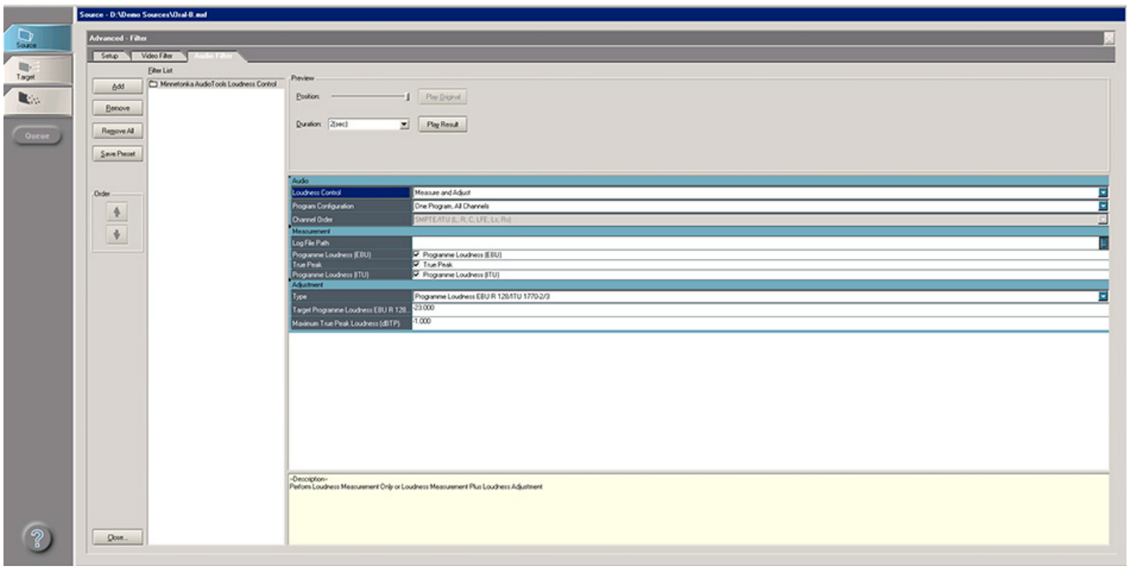
SPECIFICATIONS

The AudioTools FOCUS and Server Family

Features		FOCUS	FOCUS Server	FOCUS Server Plus	FOCUS Server for Radio	AudioTools Server
Operation	Automation through watch folders	✓	✓	✓	✓	✓
	Automation through web services API		✓	✓	✓	✓
	Number of concurrent jobs	up to 2	up to 4	up to 4	up to 4	flexible licensing
	Presets via graphical UI	✓	✓	✓	✓	✓
	Configurable graphical UI					✓
	Email Notification		✓	✓	✓	✓
	XML scriptable		✓	✓	✓	✓
	Custom Workflows					✓
Format Support	LPCM Audio: WAV, BWF, AIFF	✓	✓	✓	✓	✓
	Codecs: MP2, MP3, AAC, Dolby Digital Plus				✓	✓
	Dolby E Support			✓		✓
	Container: MXF, QuickTime	✓	✓	✓		✓
	Container: GXF, LXF, Transport Stream					✓
	Audio channels	up to 8	up to 16	up to 16	up to 16	up to 128+
	Audio program configurations	Presets	Presets	Presets	Presets	Scriptable
Loudness Processing	ITU 1770 & EBU R 128 Measurement	✓	✓	✓	✓	✓
	Program Loudness Adjustment	✓	✓	✓	✓	✓
	Momentary & Short-Term Loudness Adjustment	✓	✓	✓	✓	✓

AudioTools FOCUS and FOCUS Server can be upgraded to a full AudioTools Server configuration.

Minnetonka Audio® AudioTools™ - Loudness Control for Harmonic ProMedia™ Carbon



OVERVIEW

The AudioTools Loudness Control for Harmonic Pro-Media™ Carbon plug-in (a.k.a. ALCR) adds versatile and convenient loudness control to Harmonic's file-based ProMedia™ Carbon transcoding software. The plug-in provides comprehensive loudness measurement and flexible loudness adjustment using the same technology that powers Minnetonka Audio's AudioTools Server - the worldwide benchmark for file–based loudness control.

FEATURES

All Standards Supported

ALCR is compliant with all current standards and delivers your choice of loudness adjustment actions to cover all essential metrics, quantifying Program Loudness along with Maximum True Peak Level. EBU and CALM–specific profiles are provided to address all international regulations. Measurement results are formatted as XML, making them available to ProMedia™ Carbon for logging and subsequent reporting. The XML results are also compatible with the visualization and reporting application from Videomenthe and with other XML tools.

More than Stereo

AudioTools Loudness Control for Harmonic ProMedia™ Carbon handles mono, stereo and multichannel configurations up to 24 channels, including 32 presets and 4 custom program configuration profiles. By processing all programs in one pass, users save time prepping material, whether it's a simple stereo or multi-program configuration.

IN-DEPTH

Recommended Practice

EBU and ATSC recommended practices discourage the use of preset compression and peak limiting for loudness control in file–based environments. The AudioTools Loudness Control for Harmonic ProMedia™ Carbon plug-in intelligently applies loudness adjustment to the incoming essence to reach the desired target Program Loudness value. Loudness adjustment can be performed in accordance with EBU R 128 and, as with the measurement module, mono, stereo and multichannel PCM configurations can be processed with target values set for Program Loudness according to ITU-R BS.1770-3. If the loudness level deviates from the Target Loudness Value, adjustment is applied, and Maximum True Peak Level can be limited to previously specified target levels.

Harmonic users now have a comprehensive solution that amplifies the capabilities of ProMedia™ Carbon — an adaptable AudioTools plug-in to measure and maintain loudness on your already familiar Harmonic ProMedia™ Carbon platform. AudioTools Loudness Control for Harmonic ProMedia™ Carbon can be added to an existing video transcoding process, providing loudness control functionality to a large variety of video and container formats on the fly.

AudioTools Loudness Control for Harmonic ProMedia™ Carbon is also supported in Harmonic WFS environments allowing users to create machine groups from a collection of authorized nodes.

SPECIFICATIONS

Measurement

- Supports ITU-R BS.1770-3, EBU R 128 and other international standards & practices
- Is in accordance with the Commercial Advertisement Loudness Mitigation (CALM) Act and ATSC's Recommended Practice A/85: Techniques for Establishing and Maintaining Audio Loudness for Digital Television
- Measures:
    - Program Loudness
    - Maximum Momentary Loudness
    - Maximum Short Term Loudness
    - Loudness Range
    - Maximum True Peak Level
    - Sample Peak Level

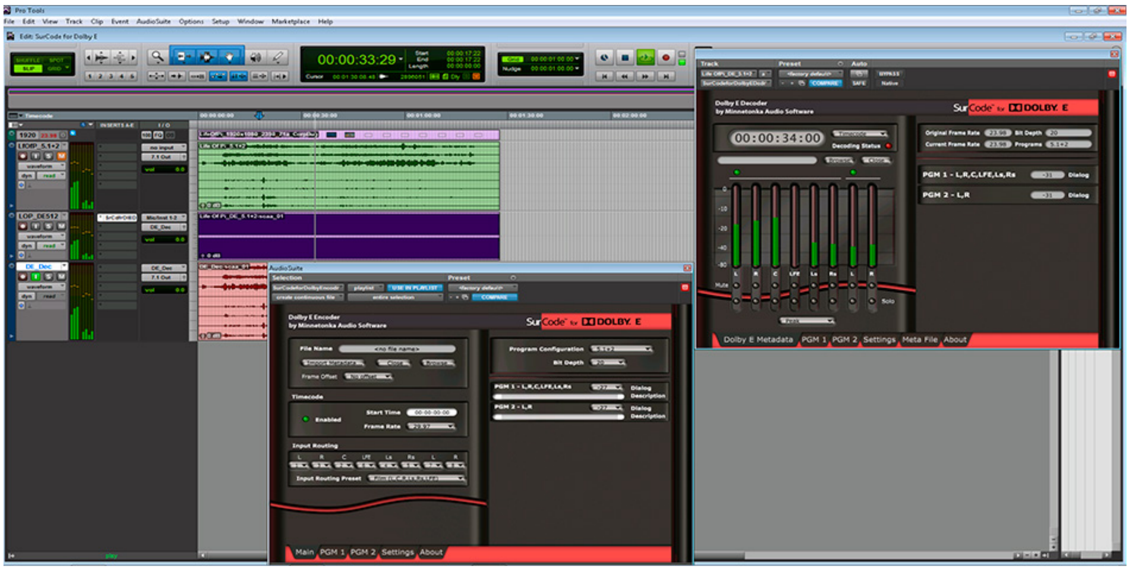
Measurement results are formatted as XML in a log file, & for use in 3rd party applications, such as the Videomethe plotter. In addition to overall loudness measurement results, maximum overall values are reported as well as measured loudness parameters throughout the file.

Adjustment

- AudioTools Loudness Control for Harmonic ProMedia™ Carbon applies loudness adjustment to the incoming essence to reach the desired target program Loudness and control the Maximum True Peak
- Loudness adjustment is performed in accordance with ITU BS.1770, 1770-2/3, EBU R 128, ARIB TR-B32, DPP & OP-59
- Mono, stereo and multichannel PCM configurations can be processed - up to 24 channels
- All programs within an asset are processed in a single pass
- Target values can be set for all adjustable parameters

AudioTools Loudness Control for Harmonic ProMedia™ Carbon can be activated on a single machine using a license file. A user-supplied USB dongle provides portable licensing. On a single host or node, AudioTools Loudness Control for Harmonic ProMedia™ Carbon can be used as multiple instances for parallel processes.

Minnetonka Audio® SurCode for Dolby® E



OVERVIEW

The Industry Standard for Dolby E On Your Workstation

Dolby E is the industry standard method of transporting multichannel digital audio across an asset's entire post–production life cycle. As the first and most complete tool kit for Dolby E, SurCode for Dolby E is a certified Dolby E encoder and decoder suite that allows fast and easy processing and management of file-based Dolby E assets and metadata.

- Structured workflow
- Increased productivity
- Flexible license handling
- Encode, decode & monitor
- Broad platform support – Avid, Adobe, Final Cut & VST
- Interoperable with Dolby hardware
- Lowest cost Dolby monitoring solution in the industry

Easy & Flexible Workflow

SurCode for Dolby E provides a simple and flexible workflow tailored specifically to your platform and workstation. All platforms, including Pro Tools, Media Composer Family products, Adobe Premiere Pro CC, VST and Final Cut are included on a single license.

- Settings are conveniently embedded in your session for instant recall or rework
- Supports all Dolby metadata
- Choose the best way to integrate into your own workflow

Complete Metadata Management

SurCode for Dolby E generates and preserves all Dolby metadata. This gives broadcast professionals the ability to control channel mode, downmix parameters, profiles, dialog level and other ancillary metadata. Browse or drag and drop DBMD to read existing metadata since SurCode for Dolby E imports and exports metadata as:

- DBMD - add your Dolby metadata chunk into WAV headers
- XML - universal format for AudioTools Server or any XML–aware application
- Text - human–readable metadata for manual QC

SurCode for Dolby E Plugin

SurCode for Dolby E includes encoder and decoder plug–ins for Pro Tools, Final Cut, Media Composer Family products, and VST versions for qualified surround–capable workstations such as Adobe Premiere Pro CC, Nuendo, Pyramix, and Sequoia..

FEATURES

Dolby E Encoder

SurCode for Dolby E Encoder provides in–depth Dolby E encoding, supporting all Dolby E encoder program configurations. Dolby E and program metadata are displayed and can be updated, all via a simple, tabbed user interface. The SurCode for Dolby E Encoder plug–in seamlessly integrates with your workstation using the current session information.

- Pro Tools and Media Composer AudioSuite; cross–platform VST & Final Cut Pro Export
- Up to 8 input channels, with flexible input routing
- Full metadata management
- Includes plug–in templates & presets
- DBMD import/export
- DP600 emulation option for guard band & other file settings
- Optionally enable or disable timecode in encoded files
- Frame offset control for tape–based or Dolby hardware layback
- Supports both dual mono or single interleaved WAV files
- Stream metadata from legacy hardware via optional USB-to-RS-422 serial cable

SurCode for Dolby E Decoder

SurCode for Dolby E Decoder decodes Dolby E files or streams, and provides output and routing of audio streams. Playback control is at your fingertips, with convenient program selection that optionally routes stereo audio to your default 1/2 bus pair, eliminating the need to repatch. The user interface displays program configuration, output metering, metadata, decoding status and a decoding error indicator for both the Dolby E file and individual programs. SurCode for Dolby E Decoder also enables real-time testing and playback for tight and consistent quality control.

- Supports all Dolby E encoder program configurations including 5.1 + 2
- Dolby E Data Bit Depth of 20 or 16 bit
- Up to 8 output channels per file or stream
- Export DBMD chunk, XML or text metadata
- Provides both peak & RMS metering (AAX, VST, RTAS & AudioSuite)
- Display & update Dolby E + per-program metadata
- Save user–defined presets

SurCode for Dolby E Stream Player

The SurCode for Dolby E Stream Player plug–in delivers real-time decoding capability for all your post–production stakeholders, even when they are only equipped for stereo routing playback. As a complement to a complete Dolby E workflow or, a low cost quality control solution for any of your rooms, SurCode for Dolby E Stream Player easily integrates the Dolby E format into existing workflows across platforms and applications, providing more choice for video post production professionals. It's a problem solver for broadcasters, contractors, second units, location crews and anyone who needs to incorporate Dolby E into their workflow without the bulk and expense of full surround. SurCode for Dolby E Stream Player enhances Media Composer family, Pro Tools, Final Cut Pro 6 or 7, Nuendo, Pyramix, or qualified VST hosts.

- Real-time downmixing of 5.1 to stereo
- Downmix parameters taken directly from Dolby E metadata for a true emulation
- Real-time program selection during playback
- Adjustable downmix headroom control
- Comprehensive Dolby E metadata display
- Export DBMD chunk, XML or text metadata

IN-DEPTH

SurCode for Dolby E – for AVID Media Composer

For faster–than–real-time processing, the SurCode for Dolby E plug–in appears as an AudioSuite encoder and decoder, right in the AudioSuite menu of your Avid workstation. For real-time audition from the timeline, the SurCode for Dolby E Decoder AAX and RTAS plug–ins are also included. The time saving Mixdown feature for Media Composer Family products allows you to apply the AAX and RTAS decoders, in a faster–than–real-time capacity, to a multichannel bus. It not only decodes the audio but automatically creates a multichannel track, and prints the decoded PCM audio to that track.

SurCode for Dolby E – for Pro Tools

For faster–than–real-time processing, the SurCode for Dolby E plug–in appears as an AudioSuite encoder and decoder, right in the AudioSuite menu of your Pro Tools workstation. For real-time audition from a file, the timeline, or a live input, the SurCode for Dolby E Decoder RTAS and AAX plug–ins are also included. When only QC or confidence monitoring is required, the low cost SurCode for Dolby E Stream Player saves money and frees up more costly resources

SurCode for Dolby E - for Adobe Premiere Pro CC

For faster–than–real-time processing, the SurCode for Dolby E Encoder is available as an export plug-in in the Adobe Media Encoder export engine. SurCode for Dolby E VST Decoder decodes directly from the Premiere Pro timeline. Simply drop a Dolby E encoded file onto the timeline and instantiate SurCode for Dolby E Decoder in the audio mixer to either audition or decode to PCM.

SurCode for Dolby E - for Nuendo

For faster–than–real-time processing, the SurCode for Dolby E VST Encoder encodes by selecting the Audio Mixdown option in the File Export menu. The SurCode for Dolby E Decoder VST plug–in allows real-time audition from a file, the timeline, or a live input as an Insert plug–in on your multichannel track. SurCode for Dolby E Stream Player saves you time and resources, decoding in Nuendo in stereo.

SurCode for Dolby E - for Pyramix

For faster–than–real-time processing, the SurCode for Dolby E VST Encoder encodes by selecting the Project menu, where you select Mix Down. The SurCode for Dolby E Decoder VST plug–in allows real-time audition from a file, the timeline, or a live input as an Insert plug–in on your multichannel track. For your B room, SurCode for Dolby E Stream Player saves time and money, with stereo decoding in Pyramix.

SurCode for Dolby E - for Final Cut

For faster–than–real-time processing, the SurCode for Dolby E Encoder and Decoder appear as options in the File Export menu. For confidence monitoring and QC in a stereo environment, SurCode for Dolby E Stream Player is the lowest cost solution available.

SurCode for Dolby E - for Sequoia

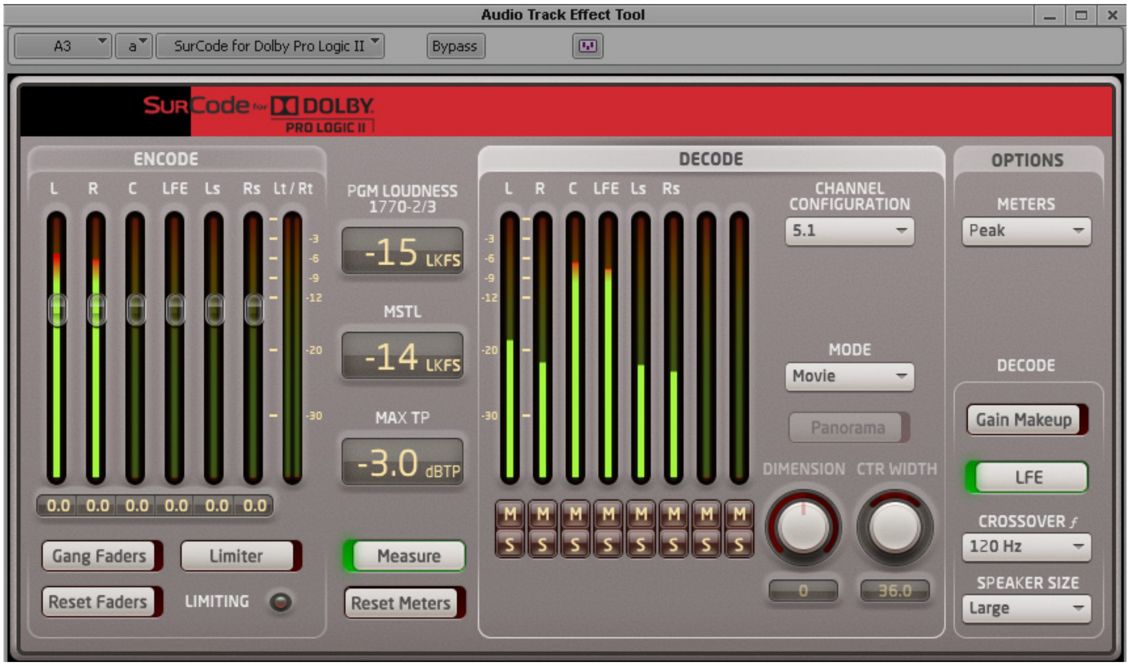
For real-time processing, the SurCode for Dolby E VST Encoder encodes via the Plug-ins menu, where you select VST FX. The SurCode for Dolby E Decoder VST plug–in gives you real-time auditioning from a file or the timeline, and SurCode for Dolby E Stream Player lets you monitor Dolby E in stereo right in Sequoia.

SPECIFICATIONS

System Requirements

For specific system requirements and product bundles, visit our website at www.minnetonkaaudio.com.

Minnetonka Audio® SurCode for Dolby® Pro Logic II



OVERVIEW

SurCode for Dolby Pro Logic II – LtRt plug-in for Workstations

SurCode for Dolby® Pro Logic II is a certified Dolby Pro Logic II encoder and decoder that allows fast and easy processing and management of file-based and real-time Dolby Pro Logic assets.

SurCode for Dolby Pro Logic II offers complete Dolby Pro Logic II encoding and Pro Logic II, Pro Logic IIx and Pro Logic IIz decoding of up to 8 channels of audio. The product enables auditioning, encoding and decoding of audio, making it easy to produce surround–ready mixes in real-time and, being file–based, at faster than real-time. SurCode for Dolby Pro Logic II also decodes back to multichannel audio, making it the perfect complement to post–production LtRt needs, and multichannel field surround microphones.

FEATURES

- Complete encoding, decoding, measuring, monitoring and QC for LtRt and LoRo in one unified environment
- Encode & decode simultaneously in real-time to easily optimize your input
- Perfect complement to SurCode for Dolby E Encoder
- v2.5 includes an RTAS plug-in running on Pro Tools 8, 9 HD, 10 or PT 8 LE with Complete Production Toolkit; 32 bit VST, and cross-platform Standalone support
- V3.0 includes a 32bit AAX plug-in running on Pro Tools 10.3.6 or higher, a 64 bit AAX plug-in running on Pro Tools 11, 12, and 2018.x and Media Composer 8.1 or higher; 64 bit VST for qualified 2.4 surround capable hosts.
- Built–in loudness measurement for simplified workflows & EBU R 128 & CALM–compliant deliverables

IN-DEPTH

Built For Loudness Standards

Broadcasters the world over need to deliver loudness–compliant content for mandated delivery channels, and should correct all material, regardless of channel and platform. SurCode for Dolby Pro Logic II displays seven different standard loudness measurement parameters:

- ITU 1770 ungated Integrated Loudness in LKFS
- ITU 1770-2/3 gated Integrated Loudness in LKFS
- EBU R 128 Integrated Loudness in LUFS
- Dialog–anchored Integrated Loudness
- Maximum Momentary & Short Term Loudness
- Maximum True–Peak

These real-time loudness measurements allow you to adjust and verify objective loudness parameters of both your source files and encoded LtRt content.

Complete – End to End

SurCode for Dolby Pro Logic II allows an end–to–end, encode/decode cycle for direct control of your mix's entire life cycle. In addition to a reference grade LtRt encoder, SurCode for Dolby Pro Logic II also provides a facility for stereo or LoRo output as well. The encoded LtRt can be monitored as a non–decoded stereo stream or decoded as a surround stream.

SurCode for Dolby Pro Logic II enables engineers to use SurCode technology in your favorite Avid or VST DAW, making it a powerful, time-saving tool to efficiently mix directly in your DAW. Now the exact same phase and amplitude behavior of the Pro Logic II consumer decoding chain can be monitored and levels corrected if necessary, saving time and ensuring the highest quality output during the mixing process. SurCode for Dolby Pro Logic II version 2.5 also offers a cross–platform, standalone version for use in any workstation environment.

SurCode for Dolby Pro Logic II monitoring and loudness measurement tools and built-in encoding/decoding features enable mixes to be optimized for all playback situations, regardless of whether the LtRt is decoded into surround or remains in stereo mode as delivered.

Tools For Today ... and Tomorrow

SurCode for Dolby Pro Logic II continues the Minnetonka Audio tradition of supplying discriminating audio professionals with full featured Dolby codecs. In keeping with our parallel philosophy to streamline workflows, version 3 integrated loudness measurement saves time and money while reinforcing industry best practices.

SurCode is the world's most trusted family of professional codecs for Dolby formats, with pro–level service and support. As with all members of the SurCode family, SurCode for Dolby Pro Logic II is fully Dolby–certified, allowing the use of associated Dolby trademarks and branding.

For television, radio, corporate, industrial, games, OTT or Mobile/Hybrid; anywhere your client wants to deliver engaging, compelling, forward and backwards–compatible content, Dolby Surround is the format of choice, and SurCode for Dolby Pro Logic II delivers easy LtRt content with complete confidence.

SPECIFICATIONS

System Requirements

SurCode for Dolby Pro Logic II is built for use in Avid's Pro Tools, Media Composer, NewsCutter and Symphony, using the AAX, RTAS and AudioSuite formats. The installer provides cross–platform VST support as well, for Nuendo, Pyramix, Sequoia, and other surround VST 2 compliant hosts. Also installed are cross–platform, standalone applications for Windows and Mac OS.

For specific system requirements and product bundles, please visit our website at www.minnetonkaaudio.com.

Minnetonka Audio® SurCode for Dolby® Digital Plus

OVERVIEW

SurCode for Dolby Digital Plus

SurCode for Dolby Digital Plus Encoder encodes 5.1, 6.1 and 7.1 surround sound audio to Dolby Digital Plus (E-AC-3) or Dolby Digital (AC-3) formats. The product accepts up to 8 channels of 44.1 or 48 kHz PCM audio at word lengths of 16, 24 or 32 bits per sample. SurCode for Dolby Digital Plus Encoder outputs frames through your favorite DAW or to an .ec3/.ac3 or WAV file and supports all standard metadata and pre-processing options.

SurCode – The Standard

Dolby Digital is the industry standard method of delivering discrete surround audio to consumers. As the most complete toolkit for Dolby Digital and Dolby Digital Plus production, SurCode for Dolby Digital Plus is the certified encoder and decoder that allows fast and easy processing and management of file-based linear PCM and Dolby Digital assets. As a plug–in, it runs in all current and legacy workstation formats.

FEATURES

SurCode for Dolby Digital Plus offers complete Dolby Digital Plus and Dolby Digital support for up to 8 channels of audio. The product enables auditioning, encoding and decoding of audio, making it easy to produce discrete surround files in real-time and, being file–based, at faster–than–real-time speeds. SurCode for Dolby Digital Plus also decodes back to multichannel PCM audio, making it the perfect complement to your post–production needs for film, TV, mobile, gaming and VOD.

- Complete encoding, decoding, monitoring and QC in one unified environment
- Encode & decode simultaneously in real-time to easily optimize your metadata
- Encoder includes a confidence monitor/decoder
- Writes ac3, ec3 & WAV files
- Supports Dolby Digital as well as Dolby Digital Plus
- Perfect complement to SurCode for Dolby E Encoder
- Cross–platform AAX & AudioSuite, for Mac OS & Windows, all on iLok

IN-DEPTH

Why Plus?

Dolby Digital Plus was designed from the ground up to be backward compatible yet broadly adaptable as your content evolves. It's scalable enough to address a wide range of delivery methods, from streaming and download, broadcast and BD to gaming, with outstanding fidelity while still delivering the reliable performance we've all come to expect from AC-3.

For theatrical release and many home theater enthusiasts, 5.1 is not enough. Dolby Digital Plus adds up to eight main channels to support increasingly popular 7.1 playback. Those extra channels result in more precise placement and localization for your audience. Thanks to significant gains in coding efficiency, all this extra capacity is carried as fully discrete, individual channels with no matrixing, so imaging is solid and repeatable across a variety of speaker configurations and playback conditions. That extra efficiency also translates into either lower data rates with quality equivalent to Dolby Digital, or superior fidelity at the same rate or file size as your legacy content.

Being a Dolby–certified product, audio data encoded by SurCode for Dolby Digital Plus is fully-compatible with all Dolby Digital and Dolby Digital Plus decoders, either in software versions, or as hardware in CE electronics or standalone pro decoders. Your Dolby Digital Plus output can be relied on to work the way you expect in the consumer's home, with no guesswork or inconsistencies.

SurCode for Dolby Digital Plus allows you to deliver an enhanced stream everywhere, with no loss in quality should conversion to AC-3 be needed.

Complete – End to End

SurCode for Dolby Digital Plus allows for an end–to–end, encode/decode cycle for direct control of your mix's entire life cycle. Choose 5.1, 6.1 or 7.1 encoding that perfectly matches your delivery platform. SurCode for Dolby Digital Plus also lets you repurpose 5.1 stems, with 6.1 and 7.1 upmixing. In addition to a reference grade E-AC-3 encoder and legacy AC-3 encoder, SurCode for Dolby Digital Plus also provides a facility for downmixing to stereo as well for QC of mixes on consumer equipment.

SurCode for Dolby Digital Plus enables engineers to use SurCode technology in Avid Pro Tools and the Media Composer Family, making it a powerful, time-saving tool to efficiently mix directly in your DAW. Now the exact same dialnorm and downmix coefficients of a Dolby Digital or Dolby Digital Plus consumer decoding chain can be monitored, and metadata corrected if necessary, saving time and ensuring the highest quality output during playback.

Tools for Today ... and Tomorrow

SurCode for Dolby Digital Plus continues the Minnetonka Audio tradition of supplying discriminating audio professionals with full featured Dolby codecs. SurCode is the world's most trusted family of professional codecs for Dolby formats, with pro–level service and support. As with all members of the SurCode family, SurCode for Dolby Digital Plus is fully Dolby–certified, allowing the use of associated Dolby trademarks and branding.

For television, radio, corporate, industrial, games, OTT or Mobile/Hybrid; anywhere, audiences demand compelling high fidelity content, SurCode delivers.

SPECIFICATIONS

System Requirements

SurCode for Dolby Digital Plus is built for use in Avid's Pro Tools and Media Composer Family, using the AAX real-time AAX AudioSuite formats. For specific system requirements and product bundles, please visit our website at www.minnetonkaaudio.com.

Telos Alliance

TelosAlliance.com | inquiry@telosalliance.com | +1.216.241.7225