LOVE WHAT YOU HEAR

CLEVELAND | LANCASTER | MINNEAPOLIS | BEIJING | DÜSSELDORF | RIGA

Telos Alliance

Telos

Omnia

Axia

LINEAR ACOUSTIC

25 seven

MINNETONKA AUDIO

jünger

LOVE WHAT YOU HEAR

At Telos Alliance®, we know that you want to be the audio expert for your media company. In order to do that, you need a way to simplify and optimize audio. The problem is, audio can be complicated, uncooperative, and doesn't always sound great. We believe you shouldn't have to struggle to deliver the best possible content to your audience, which is why we've helped hundreds of thousands of media pros like you solve their unique audio problems over the last three decades with award-winning products and custom solutions that translate to better audio and a bigger audience. You, and your audience, are going to Love What You Hear.

YOUR TRUSTED PARTNER

On your audio journey, Telos Alliance has been with you every step of the way. From analog to digital to AoIP to your entire networked ecosystem. We're not just a manufacturer and software developer, we're your partner.

We deliver innovative, intuitive audio solutions that meet the challenges you face head on, so that you can spend less time solving problems and more time creating the most exciting and engaging audio experiences imaginable for your audience.

ANY AUDIO CHALLENGE

Together, no audio challenge is too big, no technology is beyond reach. And no solution—large, small, or custom—is unobtainable. Partner with Telos Alliance to hear your future.

SCHEDULE A CONSULTATION

Schedule a consultation so you can stop losing audience to the competition and instead watch your operational costs go down, your audience grow, and your ratings skyrocket.

FIND A DEALER	CALL US	EMAIL US
TelosAlliance.com/Dealers	+1 (216) 241-7225	inquiry@telosalliance.com

ICON LEGEND

TV Application

RADIO Application

AV Application

PODCAST Application

Audio Delays

It's about time. 25-Seven Systems® provides the broadcast audio industry with the most cutting-edge audio delays on the planet.

Consoles & Audio Mixing

Telos Alliance® invented AoIP for broadcast in 2003. Since then, its Axia® AoIP mixing consoles and software have found a home in more than 9,700 facilities worldwide for various applications, including radio, TV, government, education, and more.

Distributed & Decentralized Routing (AoIP Ecosystem)

A distributed Livewire+™ AES67 'matrix' that routes audio using your existing IT infrastructure. Compact, cost-efficient Telos Alliance xNodes™ form the backbone of a distributed and decentralized AoIP routing system, available in SDI, AES, analog, mixed signal, microphone, and GPIO options.

File-Based Asset Management & Content Production Solutions

Minnetonka Audio® offers enterprise-ready audio solutions for file-based automated environments and industry-standard workstation plug-ins for Dolby® encoding and decoding.

IP Intercom & Communications

Telos Infinity® IP Intercom delivers a quantum leap in scalability, ease of integration, efficiency, and total cost of ownership, completely reimagining broadcast communications and doing away with centralized matrix technology.

Measurement & Monitoring

Linear Acoustic® and Jünger Audio meet the challenge of maintaining a consistent and high-quality viewer listening experience by ensuring that the audio emitted and transmitted from your broadcast plant meets increasingly strict compliance criteria.

ICON LEGEND

Multiline Call Handling

Telos Broadcast Telephone Systems embrace both the traditional and the new, delivering both VoIP and analog multiline call-handling and talkshow solutions products that create more compelling content–all with the legendary audio quality for which Telos is known.

On-Air TV Audio Processing

Linear Acoustic & Jünger Audio provide audio solutions for loudness compliance and regulatory demands, while delivering the best possible quality, from ingest to playout.

Radio Processing

For 20-plus years, Omnia Audio® has been obsessive about broadcast audio processors, and continues to innovate with what's next, helping broadcasters all over the world define and maintain their signature sound while remaining competitive on the dial.

Routing Control Solutions

Routing Control is about getting audio around your studio as efficiently as possible. Axia Patfhinder Broadcast Controller/VM does it all over IP.

Site-to-Site Connectivity

Telos Alliance network audio codecs are used around the world to provide ultra-reliable, best quality audio connectivity between remote locations, regardless of how challenging the network connection might be.

Stream Encoding & Processing

Dial-in your streaming sound with industry-leading stream encoders from Telos Alliance, many of which include legendary Omnia Audio processing.

Watermarking Monitoring & Enhancement

From accurately measuring your radio audience to creating personalized interactive TV broadcast content, Telos Alliance's groundbreaking watermarking solutions get the job done.

Next Generation Audio for TV

Telos Alliance® delivers on the promise of Next Generation Audio. Throughout this notebook, you'll find the following designations, defined here for better understanding of all that this new audio tech has to offer.

ATSC 3.0

ATSC 3.0 is a new standard for OTA (over the air) broadcast television in North America and South Korea, though its adoption may be more widespread in the future. From an audio standpoint, ATSC 3.0 provides the viewer with personalized audio options including the ability to choose alternate languages and commentary. It also allows for a more advanced and enhanced immersive audio experience.

Dolby® AC-4 | MPEG-H AUDIO

The ATSC 3.0 standard specifies the use of two audio codecs to deliver the audio bit-stream to viewers: MPEG-H, developed in part by Fraunhofer, and Dolby AC-4.
Both offer many of the same core capabilities, though each also provides some unique features. Regions and countries introducing ATSC 3.0 can choose to standardize on either codec. Currently, MPEG-H has been chosen by South Korea, while Dolby AC-4 has been selected in North America.

NEXTGEN TV

Technically speaking, NEXTGEN TV is the same as ATSC 3.0, but where ATSC 3.0 will be more familiar to broadcasters, it's the NEXTGEN TV logo that consumers will seek out when they walk into their local electronics retailer looking for a television that offers the most current technology and features.

DOLBY ATMOS®

Although there are several ways to create immersive audio, one of the best known is Dolby Atmos®. Atmos is an immersive experience that can be incorporated into OTA, OTT, streaming platforms, and Blu-ray for enjoyment in the home. Today, Atmos is delivered to viewers as a bitstream using Dolby Digital Plus® with JOC. For ATSC 3.0 and future formats, it will be delivered using Dolby AC-4.

AES67 Livewire+™

Tap Into the Power of the Livewire AoIP Ecosytem

Livewire+™ AES67 is an AoIP protocol that not only enables high-reliability, low-latency uncompressed digital audio over Ethernet, but crucially, it incorporates logic, control, and program associated data (PAD) abilities.

Since no signal routing solution is useful unless you can control and manage the system effectively, the Telos Alliance® Pathfinder Broadcast Controller/VM is the key to unlocking the true potential of a Livewire+ AES67 ecosystem.

By discovering and identifying all compatible devices connected to the system, Pathfinder identifies each available AoIP source and destination. Acting as a middle-ware layer using its onboard protocol translator to connect with many commonly used third-party controllers, it effectively serves as an AoIP router, an ideal core infrastructure component for a host of diverse radio and TV audio applications.

Pathfinder can schedule and trigger events, detect audio silence, issue alarms and instruct failover routing. It even allows the user to create custom screen-based control and monitoring panels.

Route & Monitor with Pathfinder

Control with Pathfinder

Throughout this notebook, you'll find the designations "Control with Pathfinder" and "Route & Manage with Pathfinder," indicating those products are part of the Livewire+ AES67 ecosystem and can interface easily and efficiently with Pathfinder.

Telos Infinity® IP Intercom

Next-Generation Intercom

The IP Intercom Solution that has everyone talking.

- Telos Infinity is a unique IP-based Intercom system providing fully featured communications in a format that does away with outdated matrix-based technology
- Livewire+™ AES67 and SMPTE ST2110-30 compliant audio ensures seamless interoperability
- Hardware includes the brand-new BP-4 Quad Channel Wired Beltpack , BP-2 Dual Channel version, MP-16 1RU Master Panel, MXP-20 Expansion Panel, DS-16 Master Desktop Station, SmartBoom Headsets and CrewCom License-Free Wireless Comms
- Infinity Dashboard Advanced: The latest software release adds VOX Detect, ISO Keys, External AES67 I/O, Scene Snapshots, Remote Device Control and Network Failover Protection
- Infinity Link connects Infinity systems in remote locations over WAN (including the Internet) and VPN. It uses the OPUS codec optimized for voice communication, and is available as a license for Beltpacks, Panels, and through Standalone Infinity Link Gateway hardware

TelosAlliance.com/Telos/Telos-Infinity

Telos VX® Prime+

Broadcast VoIP Phone System

Route & Monitor with Pathfinder

Large facility performance at small facility price.

- 8 fixed hybrid/faders (not expandable)
- Plug-and-play connectivity to Axia® Livewire® or other AES67 network
- Save money monthly on costly analog lines
- Eliminate racks full of dual couplers
- Native support of G.722 'HD Voice' codec
- Smart AGC ensures consistent caller audio levels
- Digital Dynamic EQ (DDEQ) by Omnia® for call-to-call consistency
- Connects with VoIP PBXs supplying SIP endpoints (extensions)
- Compatible with many cloud-based VoIP providers
- Connects to traditional POTS and ISDN telephone lines via Asterisk servers
- Dial-up IFBs
- Remote engineering co-ords
- On-air and straight-to-tape contribution

TelosAlliance.com/Telos/VX-Prime-Plus

Telos VX® Enterprise
Broadcast VoIP Phone System

Route & Monitor with Pathfinder

The only VoIP Phone System you'll ever need.

- Powerful and scalable for entire plants with 120 hybrids
- Plug-and-play connectivity to Axia® Livewire® or other AES67 network
- Save money monthly on costly analog lines
- Eliminate racks full of dual couplers
- Native support of G.722 'HD Voice' codec
- Smart AGC ensures consistent caller audio levels
- Digital Dynamic EQ (DDEQ) by Omnia® for call-to-call consistency
- Connects with VoIP PBXs supplying SIP endpoints (extensions)
- Compatible with many cloud-based VoIP providers
- Connects to traditional POTS and ISDN telephone lines via Asterisk servers
- Dial-up IFBs
- Remote engineering co-ords
- On-air and straight-to-tape contribution

TelosAlliance.com/Telos/VX-Enterprise

Telos® Hx6

6-Line POTS Broadcast Phone System

Route & Monitor with Pathfinder

Give your phones an instant upgrade.

- Advanced six-line talkshow system with two integrated high-performance digital hybrids manages up to six callers
- Includes Telos' famous Digital Dynamic EQ (DDEQ), noise gate, caller ducking, and acoustic echo cancellation
- Single-cable Ethernet hookup to Axia® consoles via Livewire®

TelosAlliance.com/Telos/Hx6

Telos® iQ6

6-Line POTS Broadcast Phone System

Route & Monitor with Pathfinder

Your portal to better-sounding on-air calls.

- Six-line digital phone system for use with Axia® Livewire® networked mixing consoles
- Supplies caller audio, mix-minus, Program-On-Hold audio, and switching control for six lines
- Features two high-performance digital hybrids
- Includes Telos' famous Digital Dynamic EQ (DDEQ), noise gate, caller ducking, and acoustic echo cancellation

TelosAlliance.com/Telos/iQ6

Telos® Hx1 / Hx2

Digital Hybrid Telephone Interfaces

Make your POTS phones sound incredible.

- Advanced one- and two-line POTS telephone hybrids
- Third-generation hybrids for superior audio quality
- Auto-Answer and Disconnect-Signal detection
- Features Telos' famous Digital Dynamic EQ (DDEQ) and adjustable smart leveler
- Symmetrical wide-range AGC and noise-gating by Omnia®

TelosAlliance.com/Telos/Hx

Telos® Z/IP ONE

IP Codec

Control with Pathfinder

Drops jaws. Not audio.

- IP-audio codec for remote broadcasting, STL Links, remote studio connections, and more
- 1RU perfect for studios, TOCs, and remote kits
- Livewire® port for connection to Axia® AoIP networks
- Telos Agile Connection Technology (ACT) dynamically adjusts streams as connection bandwidth or quality varies
- Includes 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
- Comes standard with Livewire AoIP, analog, and AES I/O connections

Telos® Zephyr® iPort Plus
Multi-Codec Gateway

Control with Pathfinder

The workhorse of codecs.

- Livewire® to MPEG gateway that transports multiple channels of stereo audio across IP networks
- Perfect for large-scale distribution of audio to single or multiple locations
- Connects to Axia® networks with single CAT-6 cable or using Axia xNode™ audio interfaces for use as standalone multiple-stream codec
- 16 stereo codecs—choose from either eight bi-directional codecs or 16 uni-directional
- Dual-path IP connections
- Optional aptX® codec available
- Optional content delay package available
- Up to 20 end-to-end supported unidirectional GPIO contact closures per codec are available in several modes to allow considerable flexibility of control

TelosAlliance.com/Telos/iPort

Omnia.11

FM & FM+HD Audio Processor

Flagship FM processing for the most competitive broadcasters on the planet.

- Pepino Mode 2 (new in v3.6) clipper provides more brightness at aggressive clipping levels
- Transient Detail Enhancer in the dynamics section
- Unified FM/HD Bass Clipper improves audio consistency
- New Clipper "Sparkle," Bass Clarity, and Bass Sensitivity controls
- Industry-leading G-Force Dynamics Engine
- Highly refined density detection scheme
- Program adaptive attack, release, and ratio values for smooth auto acceleration/deceleration
- AGC sections synchronize with program material
- Multiband Limiters adapt to the multiband AGC activity
- Optional Perfect Declipper replaces clipped areas from audio, restores dynamics
- Exclusive "One Louder" (embedded pilot) feature

TelosAlliance.com/Omnia/Omnia11

Omnia.9
Multi-Band AM/FM Audio Processor

AES67 Livewire+™ | Route & Monitor with Pathfinder

Unrivaled flexibility, audio integrity, and sonic impact.

- Optional HD1, HD2, HD3, streaming, and RDS
- Dual Path (two stations) available
- "Undo" for audio source restoration
- Psychoacoustic composite embedder for extra loudness
- Built-in oscilloscopes, spectrum analyzers, and RTA instrumentation
- MPX composite baseband over AES (Omnia Direct)
- Very low-latency independent Studio output for talent-monitoring
- AutoPilot automatically drops stereo pilot on mono material
- Two to seven bands of multi-band AGC and limiting
- Exclusive "One Louder" (embedded pilot) feature
- Phase Correction with Mono Bass reduces multipath distortion
- Streaming engine supports Shoutcast 2 and lossless streaming
- Optional µMPX™ encoding and optional Kantar-Certified Watermarking Support available
- Livewire+™ AES67 available for even more flexible IO (requires MKII platform)

TelosAlliance.com/Omnia/Omnia9

Omnia.7AM
AM Audio Processor

AM never sounded this good.

- Purpose-built for AM radio
- Processing infrastructure, bandwidth, and output signals engineered for maximum efficiency and performance on AM
- Provides jaw-dropping AM audio—the holy grail of loud, clean, and clear
- Loudness for AM translates to increased signal coverage, especially on the fringes of the signal, where increased modulation is critical to counteract noise and interference
- Psychoacoustically Controlled Distortion Masking Clipper, Undo Declipper, Dry Voice Detector—all the features that make Omnia sound legendary, now available in a mid-priced unit specially engineered for AM
- Available w/optional HD and/or streaming

TelosAlliance.com/Omnia/Omnia7AM

Omnia.7FM

Multi-Band FM Audio Processor

The "just right" balance of features and price.

- Switchable FM or single HD + streaming standard, optional FM + single HD + streaming + RDS
- "Undo" for audio source restoration
- Dry Voice Detector for the clearest speech
- Psychoacoustic composite embedder for dramatic, extra loudness
- Two to five bands of multi-band AGC and limiting allows full control over spectral balance and consistency
- A built-in speaker-calibration feature simplifies creating a reference environment to ensure accurate processing adjustments
- Dual internal power supplies are standard
- Exclusive "One Louder" (embedded pilot) feature, which achieves a 10% modulation increase within the normal modulation level

TelosAlliance.com/Omnia/Omnia7FM

Omnia VOLT®

Multiband, Multipurpose, Audio Processor for FM, AM, HD/DAB or Studio Applications

Route & Monitor with
Pathfinder

The four-in-one processing powerhouse.

- Unprecedented processing power and sound in a one-rack-unit package
- Switch between FM, AM, HD/DAB or dedicated FM Stereo Generator applications with a simple (free) software download
- QuickTweak™ adjustment feature lets anyone tune processing like a pro
- Future-proof versatility
- Next-generation Frank Foti-designed clipper
- Six separate AGC sections, five separate time-aligned limiter sections
- Bass pre-clipper for strong, listener-pleasing bass without intermod distortion
- Stereo enhancement for FM analog, without adding multipath
- Variable high-pass and switchable phase rotator
- Automatic mono “dry voice” sensing
- MIB2 compliant SNMP support
- Dozens of modern presets give you an awesome sound straight out of the box

TelosAlliance.com/Omnia/Volt-Audio-Processor

Omnia® SST

Audio Processing Software

Transform your PC into an Omnia processor.

- Full-featured, pro software transforms a broadcast PC into a feature-rich Omnia audio processor
- First processor to support Omnia µMPX™, a specialized codec able to transport high-quality Multiplexed FM signals over a small 320kbps data pipe
- Cleans and repairs incoming audio, optimizing it before it hits compression, limiting, and final processing stages
- The Perfect Declipper detects/reconstructs clipping-damaged audio
- Pre-processing strategy results in a clean, loud, and open sound
- Using an I7 class processor, latency can be reduced to as low as 5ms
- Includes Declipper, Delossifer, and Dehummer

Omnia.9 PTN

High-Density Virtual Audio Processing

AES67 Livewire+

The Custom Solution That Puts Omnia Processing in the Cloud.

- **Undo** – Processes clipped audio via a declipper that reconstructs audio peaks and a multiband expander that adds dynamic range to highly compressed material
- **Phase Processing** – Adjustable phase rotator eliminates distortion in asymmetrical voices. Phase scrambler minimizes distortion on instruments with strong odd-order harmonics
- **Final Limiter** – A two-band "brick wall" look-ahead limiter to deliver precise peak control
- **Downward Expanders** – Fully adjustable multiband expanders minimize noise in recorded audio sources or via mic 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

Omnia.9 PTN

High-Density Virtual Audio Processing

AES67 Livewire+

Continued Features List...

- **Wideband AGC 1/2/3** – Additional wideband compressors that can be "stacked" atop the Input AGC, located after the multiband processing stage or used as dedicated bass compressors
- **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
- **Multiband AGC/Limiters** – User-selectable between two and seven bands
- **Band Mix** – The final means of adjusting spectral balance before the final peak limiters, providing further customization of the sound

Omnia® MPX Node

Encoder/Decoder

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

- Omnia µMPX™ transports full FM composite (MPX) and RDS/RBDS at selectable rates from 320 to 576 kbps
- Internal switches let you set unit to either Encoder or Decoder
- 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
- A single Encoder can send unicast or multicast streams, providing duplicate signals to multiple decoders
- Dual Ethernet design supports redundant streaming over two separate networks for highest fault tolerance
- Decoders feature built-in alignment delays
- 5 In x 5 Out software-configurable GPIO
- Dual bank firmware update design, for secure upgrading
- Simple HTML5-based web GUI
- Half RU, fanless design

TelosAlliance.com/Omnia/MPX-Node

Omnia® VOCO™ 8

Premium Networked Mic Processor

AES67 Livewire+™ | Route & Monitor with Pathfinder

Manage, route, and process all your mics.

- Up to eight processed mics networked throughout your entire facility
- All mic channels offer a studio-grade microphone preamp, phantom power, and controls for phase, pad, and gain
- Advanced natural-sounding de-esser
- Three-band noise gate with ultra-fast attack that catches every syllable of even soft-spoken talent
- Three-band dynamics processor with our "HQ192" algorithm
- Four bands of parametric EQ and a brick wall final limiter
- "Dominate it!" feature allows host mic to always rise above the rest
- Outputs include analog, AES (with AES sync input), and Livewire+™ AES67

Omnia.9sg

Stereo Generator & Final Stage Processor

AES67 Livewire+

Your processor's secret weapon.

- Psychoacoustically controlled distortion masking clipper offers significant loudness and quality improvements when paired with any processor
- Features AutoPilot, which automatically switches off the stereo pilot during mono programming, allowing up to a 12dB increase in signal-to-noise
- Selectable SSB (single sideband) stereo encoding for potential multipath reduction
- Includes Omnia Toolbox instrumentation and full remote control
- Internal audio playback with built-in processing
- Options include local audio insertion and RDS
- Exclusive "One Louder" (embedded pilot) feature, which achieves a 10% modulation increase within the normal modulation level
- Built in stream receiver allows for a web stream to serve as an additional audio source
- Optional Kantar-certified watermarking support

Axia Quasar®
AoIP Mixing Console

Control with Pathfinder | AES67 Livewire+

The new star in IP consoles.

- Designed for any size radio studios and small TV installations
- Super-reliable 6th-generation native AoIP mixing platform
- Easy-to-operate touchscreen-based UI. No ext. display required
- New, intuitive Web UI with integrated HTML5 remote control
- Ethernet-connected and self-contained surface modules
- Built-in, modular fanless PSUs with redundant option
- Up to 4 user-assignable buttons per channel strip
- Up to 4 user-programmable Surface Layers
- 4 banks of 8 programmable Master buttons with Touch Sense
- Extensive metering built right into the surface, including fader bargraphs
- 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)
- 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

TelosAlliance.com/Axia/Quasar-AoIP-Mixing-Console

Axia Quasar®

Console Engine

AES67 Livewire+™ | Control with Pathfinder

A native AoIP powerhouse for Axia Quasar console.

- Super-reliable DSP platform based on proven native AoIP processing technology
- 64 stereo channels
- Variable slope Hi-Pass and Lo-Pass filters on every channel
- 4-band fully parametric EQ on every channel
- Powerful dynamics processing and De-Esser on every channel
- Gain-Sharing Automixer on every channel
- Four program buses and eight auxiliary buses per channel
- Virtual Mixer (VMix) with 16 independent 5-channel V-Mixers that extend the mixing capacity and flexibility of your Quasar console
- Support for AES67
- Talent headphone processing
- Redundant power is standard

Axia Fusion®
AoIP Mixing Console

Control with Pathfinder | AES67 Livewire+™

Who says consoles can't be smart and sexy?

- Anodized metal work surface, plus high-res OLED displays
- Available in sizes from eight to 28 positions, expandable up to 40 faders
- Four stereo program buses, four send buses, and two return buses
- One-plug connection to Axia® network via a single Ethernet cable
- Automatic mix-minus for phones and codecs, plus built-in talkback
- Custom source profiles with detailed control over every input
- 99 show profile presets
- T-Series faders
- Loudness measurement
- Livewire+™ AES67 I/O
- Audio delays on input
- Motorized faders
- Pathfinder integration for AFV (audio follow video)
- Auto mixer

TelosAlliance.com/Axia/Fusion

Axia® Fusion®
Studio Control Modules

AES67 Livewire+ | Control with Pathfinder

Give your talent the power.

- Six accessory control panels for convenient talent / guest control of frequently changed options
- Headphone and mic control; GPIO closures; routing scene changes tied to Axia Pathfinder; talkback to CR board op or guest positions
- Easy RJ-45 connection to console
- Can be flush-mounted in any flat or vertical solid surface

TelosAlliance.com/Axia/Studio-Control-Panels

Axia® PowerStation®

Integrated Console Engine

AES67 Livewire+™ | Control with Pathfinder

Power/mix engine for Fusion consoles.

- Fanless design with heavy machined heat sinks. Completely silent
- 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
- Built-in, zero-configuration network switch with Gigabit and SFP for long-distance fiber connection
- Add more I/O with PowerStation Aux, or à la carte using Axia Audio xNodes™
- Supports AES67, including Livewire+™ AES67

Axia® iQx

AES67 Mixing Console

AES67 Livewire+™ | Control with Pathfinder

Look below the surface.

- Mixing console and DSP engine in one chassis
- Optimized for AES67, including Livewire+™ AES67, with the ability to mix streams to and from the network with no local I/O required
- Standards-based console supports AES67 and SMPTE ST 2110-30
- Configurable from 8 to 24 faders, each with instant access to any source
- Assign any type of source to any channel
- Four main stereo outputs (Program-1 through Program-4)
- Built-in three-band EQ available for each source
- Channel-input confidence meters assure operator of audio presence
- Automix capabilities allow operators to balance levels of on-air-microphones when more than one mic is open at a time in a studio
- Unique Record Mode enables one-button setup of record mixes for phone bits or off-air interviews
- Four custom Show Profile “snapshots” can be saved to instantly recall frequently used console setups
- All functions can be accessed remotely via Web
- Fanless power supply standard; second redundant power supply optional
- Loudness measurement
- Audio delays on input
- Pathfinder integration for AFV (audio follow video)
- Automix

TelosAlliance.com/Axia/iQx

Axia® iQ

AoIP Mixing Console (Expandable Up to 24-Fader)

Base model eight-channel with optional iQ 6-Fader Telco Expansion Frame

AES67 Livewire+™ | Control with Pathfinder

The smarter IP console.

- Plugs into the QOR.32 engine using a single cable
- Operates as a standalone console; can also connect to Axia networks
- With QOR v2.1, iQ supports AES67
- Three dedicated stereo program buses, plus a stereo utility bus
- Automatic mix-minus on each fader, plus talkback functions, one-button off-air record mode, and show profile functions
- Expand faders and control capabilities by adding iQ expansion frames
- Loudness measurement
- Livewire+™ AES67 I/O
- Audio delays on input
- Pathfinder integration for AFV (audio follow video)

TelosAlliance.com/Axia/iQ

Axia® Radius

8-Fader AoIP Mixing Console

AES67 Livewire+™ | Control with Pathfinder

Small console for small studios, small price.

- Designed for small standalone or networked studios
- Powered by the Axia QOR.16 console engine via a single cable
- Includes four stereo program buses: three dedicated program, audition, and utility mixing outputs plus a stereo utility bus
- Automatic mix-minus for each fader, plus talkback functions, off-air record mode, show profile functions
- With QOR v2.1, Radius now supports AES67, including Livewire+™ AES67

Axia® DESQ
6-Fader Compact Desktop AoIP Console

AES67 Livewire+™ | Control with Pathfinder

The perfect desk companion.

- Cost-effective, with small footprint for small production studios, remote vehicles, content ingest stations
- Compact 16” square footprint; does not require countertop cutout
- Has two stereo mixing buses and a preview bus
- Powered by the Axia QOR.16 console engine via a single cable
- Automatic mix-minus, EQ for voice sources, show profile functions
- With QOR v2.1, DESQ supports AES67, including Livewire+™ AES67

Axia® RAQ

6-Fader Rack AoIP Console

AES67 Livewire+™ | Control with Pathfinder

The power of Axia, in your RAQ.

- Convenient way to add a physical mixing surface anywhere, despite space constraints
- Includes six rotary faders with OLED channel options displays
- Two stereo mixing buses and preview bus
- 2RU heavy-duty construction, with aircraft-grade switches and all-LED lighting
- Automatic mix-minus and show profile functions
- Powered by the Axia QOR.16 console engine via a single cable
- With QOR v2.1 upgrade, RAQ supports AES67, including Livewire+™ AES67

Axia® QOR.32 and QOR.16

Integrated Console Engines

AES67 Livewire+™ | Control with Pathfinder

Brains and brawn for your iQ, Radius, DESQ, or RAQ consoles.

- Fanless design with heavy machined heat sinks. Completely silent
- Front-panel LED display monitors power and network status
- Telecom-grade power supplies are designed for maximum uptime
- Add an Axia Console Power Supply 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
- Automix
- Large variety of built-in audio I/O; boasts studio-grade audio performance specs
- Add more I/O à la carte using Axia xNodes™
- AES67, including Livewire+™ AES67

Axia® Control Panels
Routing Control

Control with Pathfinder

Fingertip control where you need it.

- Create large, IP-based routing networks of up to 10,000 streams
- Six accessory control panels offer convenient talent / guest control
- A variety of studio operations: routing scene changes, GPIO closures, XY control of inputs/outputs, high visibility OLED router control, and more
- Slim panel design mounts in any 1RU rack space
- Fanless, convection-cooled

TelosAlliance.com/Axia/Routing-Panels

Axia® IP-Tablet

Virtual Radio Software

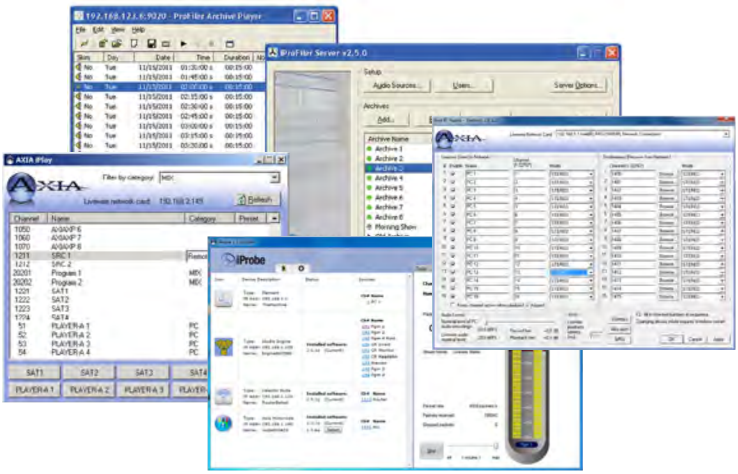
Virtualize your studio.

- Adds value to an Axia console purchase with virtualization of control and hardware resources
- Simplifies/aggregates control of several Telos Alliance devices
- Manages user rights for device access
- Minimizes need for external console monitors, freeing up space in the studio
- Optional aluminum mount fits Fusion consoles, taking up four console slots
- Designed by Livewire® partner IP-Studio for use with HTML5-equipped gear and various Telos Alliance products
- Available for Axia PowerStation, StudioEngine, QOR.32/16, xNodes, & Pathfinder button; Omnia VOCO 8 & Omnia.9; Telos VX & Z/IP One; Metadata Tools
- Works on Windows Tablets and PCs

TelosAlliance.com/Axia/IP-Tablet

Axia® Software

There's an app for that.



Comprehensive line of software solutions include:

- SoftSurface Virtual Console Software
- iProbe Network Management Software
- Axia Livewire+™ AES67 IP-Audio Driver for Windows
- Axia Livewire+ AES67 IP-Audio Driver for Linux
- Axia SoftCom Intercom for Windows
- iProFiler Automated Program Archiving
- iPlay Network Stream Player for Windows
- IP-Tablet Virtual Radio Software

Axia Livewire+™ AES67 Driver

IP-Audio Driver

Easy IP audio, fully AES67-compliant.

- Fully AES67-compliant IP audio driver
- Send and record up to 24 channels of stereo PC audio directly to/from Axia networks via Ethernet–no sound cards needed
- Receives audio from the Livewire® / Livewire+ AES67 network to destinations on the PC/Windows system
- GPIO function conveys “button-press” data from the Livewire / Livewire+ AES67 network to destination applications
- Single and Four stream versions emulate standard sound cards with inputs and outputs appearing in your Windows Control Panel
- Multichannel OEM versions emulate 8 or 24 stereo channel sound cards with support for most radio automation systems
- Supports 5.1 surround audio streams as well as stereo, configurable on a per-stream basis

Axia® Pathfinder Core™ PRO

Broadcast Controller / VM

See more. Do more. Control more.

- Reliable, redundant, system-wide broadcast controller/VM
- Automatic router table generation; graphical interface with real-time state reporting, logic gates for creation of complex logic, and control protocol for third-party integration
- PC-platform-independent; you can use almost any device to interface
- Next generation of Livewire® control builds upon our extensive experience controlling Livewire systems
- Linux-based network-attached appliance with a Web-UI provides route control and custom logic events
- Includes 1,000 points for use as crosspoint or Logic Flow endpoints
- Add-on licenses available for additional 100 points and 500 points for use as crosspoints or Logic Flow endpoints
- VML comes with 300 points for use as crosspoints or Logic Flow endpoints

TelosAlliance.com/Axia/Pathfinder-Core-Pro

Telos Alliance® xNodes™

IP-Audio Interfaces

AES67 Livewire+ | Control with Pathfinder

The most advanced AoIP interfaces on the planet.

- Third-generation IP-Audio interfaces
- One-button configuration for ultra-simple setup
- Fanless, noiseless
- Versatile mounting options, including two xNodes in 1RU
- Dual network interfaces and dual power inputs for redundant operation
- Available in Analog, AES/EBU, Microphone, Mixed-Signal, SDI and GPIO versions
- V2.0 adds a fully configurable Mixing Matrix that allows you to mix both physical and virtual inputs for added flexibility
- First and only AoIP I/O device that is Livewire+, RAVENNA, SAP, and AES67 compliant

TelosAlliance.com/Telos/xNodes

Z/IPStream® Hardware

Streaming Audio Processing and Encoding

Route & Monitor with
Pathfinder

Stream like you mean it.

- Pro-grade, reliable hardware for streaming and encoding, featuring genuine Omnia® processing
- Convenient single rack unit
- R/1 features wideband AGC, a three-band compressor- limiter, EQ, low-pass filter, and a precision look-ahead final limiter
- R/2 features optional Omnia.9 processing, which includes selectable two to seven bands of multi-band processing for consistent spectral balance and a final two-band look-ahead limiter for clarity and loudness. Includes proprietary 'Undo' with de-clipper to prevent listener fatigue plus six-band parametric EQ for creation of a perfect signature sound
- R/2 features xHE-AAC for very low bitrates, plus adaptive streaming for rock-solid connection
- R/1 accepts a single audio input, can encode at two different bitrates and can send the stream to up to four media servers
- R/2 accepts up to 8 audio inputs, can encode each at multiple bitrates and can send the stream to potentially hundreds of destinations

Z/IPStream® Software

Streaming Audio Processing and Encoding

Route & Monitor with Pathfinder

Stream like you mean it.

- All-in-one streaming-audio processors/encoders for Windows PCs featuring Omnia® processing
- X/2 for advanced streaming/encoding featuring adaptive streaming capability that allows media players to automatically adapt to changing network conditions by encoding at multiple bitrates; xHE-AAC for extremely low bitrates
- 9X/2 includes X/2 features PLUS Omnia.9 processing and other Omnia.9 advantages for the ultimate in streaming quality and manageability
- Simultaneous MP3/AAC/aacPlus encoding, compatible with Shoutcast, Icecast, Wowza, and RTMP servers
- Supports multiple simultaneous Wave audio interfaces
- Can be installed on-prem or deployed to AWS, Azure or servers from other cloud services

25-Seven® Voltair

Watermark Processor & Monitor

Knowledge is power.

- Monitors/analyzes robustness of watermark encoding across program content
- Gives visibility into how listening environments influence watermark decoding
- Includes advanced audio signal processing to enhance detectability of watermark codes within the context of programming objectives
- Empowers programmers to make informed decisions to address potential weaknesses in either encoding or decoding
- Data Export License allows downloadable spreadsheets for each broadcast day, with a confidence level for each minute of your programming

25-Seven® TVC-15

Broadcast Watermark Analyzer and Monitor

See what counts.

- Detect, monitor, and analyze how well each element in your programming supports watermarking
- Every 400 milliseconds TVC-15's tone verification codec analyzes the actual code symbols in any audio you feed it
- Measurements happen in real time, right off the air, without depending on or being connected to a particular encoder
- Front-panel timer updates every time station broadcasts a complete watermark message
- Downloadable reports and remote readouts
- Dynamically control Voltair processing based on moment-by- moment analysis of your actual air signal

TelosAlliance.com/25-Seven/TVC-15-Watermark-Analyzer-Monitor

25-Seven® Program Delay Manager

Profanity Delay

Route & Monitor with
Pathfinder

Profanity delay, reinvented.

- Ease-of-use, transparent audio quality, and program-director- friendly features take an old process to a new level
- PD-Alert™ feature archives and emails two time-stamped audio files capturing what took place when material was dumped
- Provides an instant log record establishing action and intent to keep the airwaves clean
- 99 seconds of stereo audio delay and a customizable dump button standard
- Overkill lets you select file to play over the dump buffer instead of collapsing the delay

Minnetonka AudioTools® Server

File-Based Content Production Solutions & Audio Workflow Orchestration

ATSC 3.0 | MPEG-H AUDIO | DOLBY ATMOS® | NEXTGENTV™

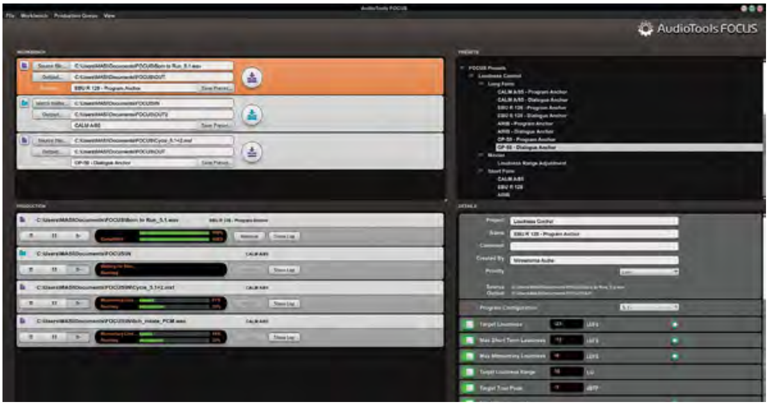
Quality-first audio workflow orchestration in flexible, focused packages of audio specialization.

- Custom, efficient file-based workflows
- Unrivaled loudness tools & compliance to broadcast standards
- Interoperability with all major workflow environments
- Modular, scalable platform: on premise, VM, or in the cloud
- Flexible license deployment in a multi-node configuration
- V5 adds RESTful web services API, Netflix compliance profiles, new QC features, Kantar watermarking, and AutoMix
- Use cases include: Loudness Measurement & Correction, Dialog Intelligibility, Encode/Decode, Pitchshift/Time Compression, Channel Assignment Detection, Audio QC
- Global NEXTGEN TV support for Dolby Atmos® and MPEG-H Immersive Audio formats
- Immersive formats for UPMAX™. Coupled with new Dolby Digital Plus Atmos encoding, upmixing stereo or 5.1 to the new immersive formats (5.1.2, 7.1.4, 9.1.6, etc.) makes for a "complete Atmos toolkit"
- Omnia.9 processing for podcasts on-demand

TelosAlliance.com/MinnetonkaAudio/AudioTools-Server

AudioTools® FOCUS for Radio

Standalone Windows File-Based Loudness Control & Podcast Processing Software Application



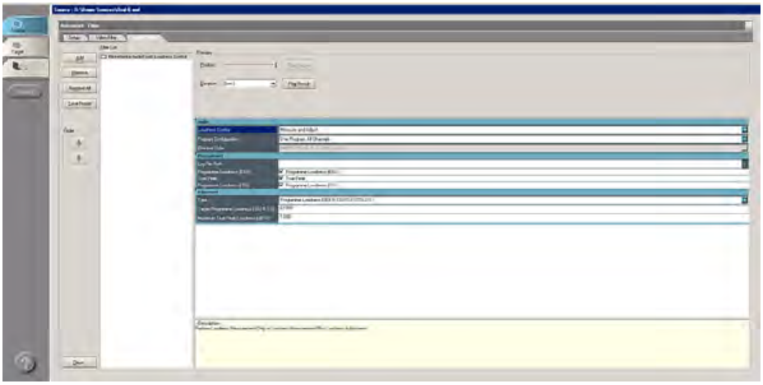
Great sound makes your podcasts stand out!

- Make your podcasts on-demand sound like great radio with our Omnia.9 radio processing!
- Use your favorite Omnia.9 presets on all your file-based content
- Consistent loudness with quality-first results means your fans listen longer
- Complete compliance with international loudness control standards
- Hot Folder, file-based, or API-driven processing
- PCM, MP2, MP3, and AAC support
- Loudness adjustment or pure logging
- File type auto-detection
- Clearly labeled, proven presets right out of the box
- WAV, BWF, and AIFF file-handling
- Purchase or low monthly subscription fee

TelosAlliance.com/MinnetonkaAudio/AudioTools-Focus

AudioTools® Loudness Control for Harmonic ProMedia™ Carbon Loudness Control Plug-in

Measurement and adjustment is performed in accordance with EBU R 128, A/85 & BS.1770-3



Transcode with industry leading, globally compliant Loudness Control.

- Measures:
  - Program Loudness
  - Maximum Momentary Loudness
  - Maximum Short Term Loudness
  - Loudness Range
  - Maximum True Peak Level
- 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
- Fully supported in Harmonic's WFS environment
- 32 program configuration presets–up to 4 additional custom program configs

SurCode® for Dolby® Pro Logic II

Certified LtRt Encoder, Decoder and Monitoring Plug-Ins for Workstations with Built in Loudness Measurement

The world's most widely used surround format with Loudness Control!

- Leading brand of broadcast standard plug-ins, applications, and SDKs with certified support for all major codecs
- Available for AVID and VST-based work- stations, and as modules for AudioTools Server and OEM
- 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
- v2.5 includes an RTAS plug-in running on Pro Tools 8, 9 HD, 10 or PT 8 LE with Complete Production Toolkit; VST 32bit, 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 or higher and Media Composer 8.1 or higher, and 64 bit VST for qualified 2.4 surround capable hosts
- Built-in loudness measurement for simplified workflows & EBU R 128 & CALM-compliant deliverables

TelosAlliance.com/MinnetonkaAudio/SurCode-for-Dolby-Prologic-II

SurCode for Dolby® E

Certified Dolby E Encoder, Decoder, and Monitoring Plug-Ins for Workstations

The industry standard for Dolby E on your workstation.

- 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

TelosAlliance.com/MinnetonkaAudio/SurCode-for-DOLBY

SurCode for Dolby® Digital Plus

Certified Dolby Digital Plus (E-AC-3) and Dolby Digital (AC-3) Encoder, Decoder and Monitoring Plug-ins for Workstations

Complete, end-to-end Dolby Digital Plus surround sound plug-in suite for post- production.

- Encoding, decoding, monitoring and QC in one unified environment
- Encode & decode simultaneously in real time to easily optimize your metadata
- Perfect complement to SurCode for Dolby E
- Cross–platform AAX & AudioSuite, for Mac OS & Windows, all on iLok
- 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

TelosAlliance.com//MinnetonkaAudio/SurCode-for-Dolby-Digital-Plus-Bundle

SurCode for Dolby® Digital Plus for Adobe®

Creative Cloud Certified Dolby Digital Plus (E-AC-3) and Dolby Digital (AC-3) Encoder

Dolby Digital Plus surround sound encoder plug-in for post-production.

- Only Dolby encoding solution for the Adobe® platform
- Encode directly from the Premiere Pro timeline
- Perfect complement to SurCode for Dolby E
- Cross-platform for Mac OS & Windows
- Fully compatible with all Dolby Digital and Dolby Digital Plus decoders, either in software versions, or as hardware in CE electronics or stand-alone pro decoders

TelosAlliance.com/MinnetonkaAudio/SurCode-for-DOLBY

Telos Alliance® SDI AoIP Node

SDI IP Audio Interface

Route & Monitor with Pathfinder

SDI/AES67 conversion for AoIP workflows.

- Two 3 Gb/s HD/SD-SDI inputs with de-embedding of 16 audio channels per input to AES67
- Re-embedding of AES67 audio to two SDI outputs
- Video compensating delay for A/V sync
- Two Gigabit Ethernet ports for control and AES67
- Dual internal redundant power supplies

Linear Acoustic® LA-5291

Professional Audio Encoder

DOLBY ATMOS®

Audio encoding, transcoding, and decoding for Dolby Atmos® workflows.

- Decoding from Dolby ED2 to PCM
- Transcoding from Dolby ED2 and Dolby E to Dolby Digital Plus JOC and Dolby Digital Plus
- Encoding PCM to Dolby Digital Plus JOC, Dolby Digital Plus, and Dolby ED2
- 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 4K video workflows or MADI I/O (mutually exclusive options)
- Dual internal redundant auto-ranging power supplies
- Browser-based remote control

Linear Acoustic® LA-5300

Broadcast Audio Processor

ATSC 3.0 | MPEG-H AUDIO | Dolby® AC-4 | NEXTGEN TV | DOLBY ATMOS®

The complete audio processor for NEXTGEN TV/ ATSC 3.0 audio.

- 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 to support 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

TelosAlliance.com/Linear/LA-5300

Linear Acoustic® UPMAX® ISC

Immersive Soundfield Controller

Upmixing for Immersive and NextGeneration Audio.

- 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

TelosAlliance.com/Linear/UPMAX-ISC

Linear Acoustic® AERO.2000

Loudness Management Platform for DTV

High-quality, compliant audio with the convenience of a front-panel GUI.

- 2RU loudness management platform with front panel controls and display
- Can host one or two AEROMAX® processing instances in AMX5.1, AMX2.0, or AMX5x2 configurations (Min. 1 required)
- UPMAX®-II upmixing/downmixing with automatic detection and downmix replacement
- Available Dolby® Digital and Dolby Digital Plus transcoding
- Available Nielsen® watermark encoding
- 8 audio pairs via SDI I/O with SDI video delay, 8 audio pairs via AES
- Dual power supplies and failover bypass relays
- Extensive TCP/IP remote control

Linear Acoustic® AERO.100

Loudness Management Platform for DTV

High-quality, compliant audio in a compact 1RU design.

- Space-efficient 1RU loudness management platform
- Can host one or two AEROMAX® processing instances in AMX5.1, AMX2.0, or AMX5x2 configurations (Min. 1 required)
- UPMAX®-II upmixing/downmixing with automatic detection and downmix replacement
- Available Dolby® Digital and Dolby Digital Plus transcoding
- Available Nielsen® watermark encoding
- 8 audio pairs via SDI I/O with SDI video delay, 4 audio pairs via AES
- Dual power supplies and failover bypass relays
- Extensive TCP/IP remote control

Linear Acoustic® AERO.10

Loudness Management Platform for DTV

Cost-effective loudness control for PCM audio without compromising quality.

- Space-efficient 1RU loudness management platform with front panel controls and display
- Hosts one AEROMAX® processing instance in AMX5.1, AMX2.0, or AMX5x2 configurations (user-selectable)
- UPMAX®-II upmixing/downmixing with automatic detection and downmix replacement
- PCM audio only (no Dolby® coding)
- 8 audio pairs via SDI I/O with SDI video delay, 4 audio pairs via AES, 2-channel audio via analog
- Dual power supplies and failover bypass relays
- Extensive TCP/IP remote control

TelosAlliance.com/Linear/AERO10

Linear Acoustic® AERO.8000

Loudness Management Platform for DTV

AES67 Livewire+™

The future of Audio over IP loudness control – today.

- 1RU high-density loudness management platform featuring Livewire+™ AES67 AoIP for enterprise-wide audio access
- Hosts up to eight AEROMAX® processing instances in AMX5.1, AMX2.0, or AMX5x2 configurations (Min. 1 required)
- UPMAX®-II upmixing/downmixing with automatic detection and downmix replacement
- Available Dolby® Digital and Dolby Digital Plus transcoding
- Available Nielsen® watermark encoding
- Extensive TCP/IP remote control
- Requires Livewire+ AES67 I/O interface such as the Telos Alliance SDI xNode

TelosAlliance.com/Linear/AERO8000

Linear Acoustic® AMS

Authoring and Monitoring System

Leading the way to ATSC 3.0 audio.

- ATSC 3.0-compliant real time authoring, rendering and monitoring of immersive 3D audio
- Features Next Generation Audio (NGA) technologies including MPEG-H
- Simultaneously delivers both ATSC 3.0 and 5.1/2-channel audio for ATSC 1.0 using smart metadata
- Linear Acoustic APTO® loudness control for high quality audio and compliance with global loudness standards
- Includes two Telos Alliance SDI xNode AoIP interfaces for Livewire+ AES67 I/O
- Also available as a monitoring- and rendering-only solution

TelosAlliance.com/Linear/AMS

Linear Acoustic® ARC

Television Loudness Processor

Linear Acoustic quality with the right features at the right price.

- 1RU next-generation television loudness processor with APTO® loudness adaptation algorithm
- Processing for two PCM stereo or mono programs
- User-defined target output loudness and true peak values
- Compliant with all global loudness standards including EBU R128 and ATSC A/85 (CALM)
- 3Gb/s HD/SD-SDI, AES-3, and AES67 I/O for SMPTE ST 2110 workflows
- Rolling six-hour loudness logging for each program, plus separate loudness event log to easily identify loudness issue and system event log
- SNMP
- Dual redundant internal power supplies
- Web-based user interface

Jünger Audio AIXpressor

Next Generation Audio Processor

The audio processing revolution.

- Powerful x86 based processing core
- flexAI ecosystem–software defined processing
- On board interfaces: 4x Ethernet/Audio over IP, 2x MADI or tieLight with SFP modules, AES3 I/O (switchable between 8x8 or 4x12 channels), Analog I/O (switchable between 8x8 or 4x12 channels), Sync in
- I/O modularity via 4 interface slots
- 2x USB host, 1x USB client
- Native AES67 and SMPTE ST2110 support
- Livewire+™ AES67, RAVENNA, and Dante® AoIP
- tieLight interface for cascading with up to 1,024 channels
- 6.6-inch high-resolution touch sensitive display
- Powerful headphone output
- 19”, 1RU device, redundant PSU, relay bypass

Jünger Audio EASY LOUDNESS
Dual Stereo Level Magic Audio Processor

Set-and-forget loudness control.

- Level Magic loudness management
- Loudness measurement
- Compatible to ITU-R BS.1770 (all revisions), EBU R128, ATSC A/85,
- ARIB TR-B32, Free TV OP-59 and Portaria 354
- True peak limiter
- Dual stereo audio processing
- Fail Over with signal loss detection
- Dante®/AES67 Audio over IP or 3G-SDI
- On board interface: 1x AES3 I/O (XLR & BNC), Sync in, Sync out, 8 GPI/O
- Optional loudness logging software (J*AM)
- External control via network, Ember+ or GPI/O

Jünger Audio D*AP8 MAP

Multichannel Monitoring Audio Processors

Monitoring and Dolby® workflow solution.

- Multichannel/multi-format audio-monitoring system (up to 7.1)
- Loudness measurement supporting all worldwide standards
- Multiple speaker connectivity with extensive solo/mute functions
- Speaker alignment: parametric EQ, delay, bass management, and downmix
- Dedicated log ports for network based measurement and logging
- Officially featured Dolby Digital Plus, Dolby Digital, Dolby E workflow solution
- Dolby encoding, decoding and metadata emulation
- Perfect replacement for Dolby's no longer available devices DP570, LM100, DP563, and DP571
- Intuitive touch-capable user interface
- Optional loudness logging software (J*AM)
- On board interfaces: 4x AES3id I/O, Sync in, Sync out, Metadata I/O, 8 GPI/O
- I/O modularity via 2 interface slots
- External control via network, Ember+ or GPI/O
- 19", 1RU device, redundant PSU, relay bypass

Jünger Audio COMPACT
High-Density SDI Audio Processing

3G-SDI powerhouse for up to 256 audio channels.

- COMPACT 256 and COMPACT 64
- Configurations for up to 4 (C64) or 16 (C256) independent SDI-Streams
- 3G-HD, HD, and SD
- Level Magic loudness management
- Compatible to ITU-R BS.1770 (all revisions), EBU R128, ATSC A/85, ARIB TR-B32, Free TV OP-59, and Portaria 354
- Dynamics
- 5.1 Surround Upmix
- Fail-over switching with auto-mono
- True peak limiter
- External control via network, Ember+ or GPI/O
- 19", 1RU or 3RU devices, redundant PSU, relay bypass

Telos Alliance

TelosAlliance.com | inquiry@telosalliance.com | +1.216.241.7225