



LOVE WHAT YOU HEAR



Telos®

Omnia®



L LINEAR ACOUSTIC®



M 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

TelosAlliance.com/Dealers

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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.



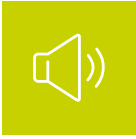
Measurement & Monitoring

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.



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.



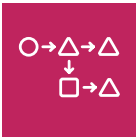
On-Air TV Audio Processing

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 Pathfinder 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



The ATSC 3.0 standard specifies the use of two audio codecs to deliver the audio bitstream 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.



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.



Tap Into the Power of the Livewire AoIP Ecosystem

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



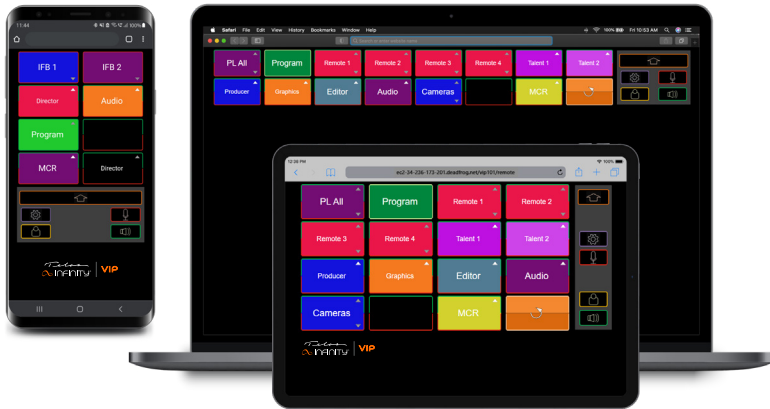
The IP Intercom Solution that has everyone talking.

- 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
- Telos Infinity Dashboard displays the entire Intercom system as a single page view; manages hardware & software elements; monitors network performance and fault logging
- 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



Telos Infinity® VIP

Virtual Intercom Platform



Route & Monitor with
Pathfinder 

First we broke the matrix, now we're putting Intercom in the cloud.

- First fully-featured Cloud-based intercom system
- Cloud-based media production workflows available on any device—smartphone, laptop, desktop, or tablet
- Use with third-party control devices, like Elgato's Stream Deck®
- Award-winning performance, scalability, ease of integration, and operational/cost efficiencies
- Use Telos Infinity VIP hardware appliance or your own server for on-prem installations
- For both On-Prem or Cloud versions, VIP integrates with Telos Infinity hardware comms, or any third-party intercom or audio subsystem using AES67 or SMPTE 2110-30 connectivity
- Available on **GV | AMPP (Agile Media Processing Platform)** as SaaS, a professional intercom solution for cloud-based media production workflows



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



Telos VX[®] Enterprise

Broadcast VoIP Phone System



Route & Monitor with
Pathfinder 

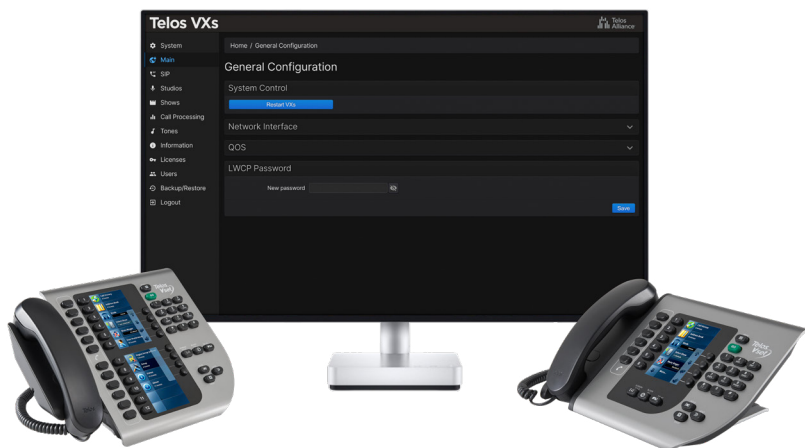
The only VoIP Phone System you'll ever need.

- 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



Telos® VXs

Virtual VoIP System



Route & Monitor with
Pathfinder 

Control with
Pathfinder 

Legendary Telos VX® performance, virtualized.

- Powerful VoIP call management for Radio, TV, and Pro AV workflows
- Software deployed on a server in the rack room, in the data center, or in the cloud
- Scalable from a single line system, to a facility-wide system, to a multi-facility system
- Plug-and-play connectivity to Axia® Livewire®, or other AES67, or compatible SMPTE-2110 networks
- Native support of G.722 'HD Voice' codec
- Clear, clean caller audio from fifth-generation Telos Adaptive Telephony technology, including Digital Dynamic EQ, AGC, adjustable caller ducking, and audio dynamics processing by Omnia®.
- Connects with all modern VoIP PBXs and UCaaS providers supplying SIP endpoints (extensions)
- Connects to legacy traditional POTS and PRI/BRI/ISDN circuits telephone lines via Telos-configured Asterisk servers
- Dial-up IFBs
- Remote engineering coordination
- On-air and straight-to-tape contribution



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®



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® iPort High Density

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 Telos Alliance xNode™ audio interfaces for use as standalone multiple-stream codec
- Comes with eight bi-directional stereo codecs
- License additional codecs up to a maximum of sixty-four
- 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



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



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 μ MPX™ encoding and optional Kantar-Certified Watermarking Support available
- Livewire+™ AES67 available for even more flexible IO (requires MKII platform)



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



Omnia® SST

Audio Processing Software



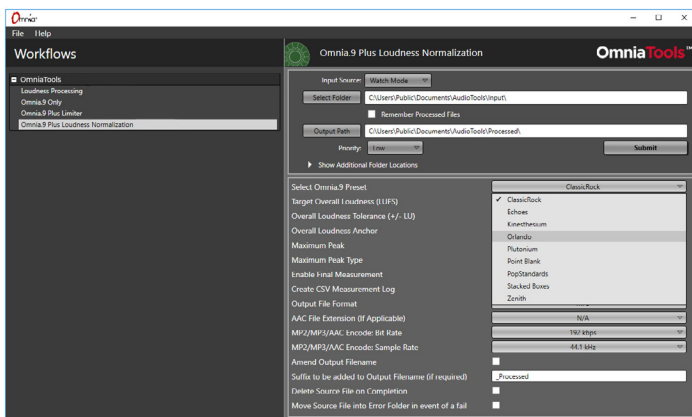
Transform your PC into an Omnia processor.

- Full-featured, pro software transforms a broadcast PC into a feature-rich Omnia audio processor
- First processor to support Omnia μ MPX™, 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



OmniaTools™

Enterprise-Class File-Based Loudness Control & Audio Processing Windows Service



Podcast and on-demand audio that stands out and keeps your audience engaged.

- Make your on-demand content sound great with Omnia 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
- License purchase or available on a subscription basis

Omnia Enterprise™ 9s

The Custom Solution that puts Omnia.9 Processing in the Cloud.



All the best features of our Omnia.9 hardware processor in software. Deployable on a server or in the cloud. Features include:

- **Undo** - Award winning process reconstructs clipped audio peaks and a multiband expander that adds dynamic range to highly compressed material
- **Phase Processing** - Adjustable phase rotator and scrambler eliminates distortion in asymmetrical voices, minimizes distortion.
- **Final Limiter** - A two-band "brick wall" look-ahead limiter to deliver precise peak control
- **Downward Expanders** - Fully adjustable multiband expanders minimize noise from poor environments
- **Input AGC** - Super clean input gain stage compensates for varying input levels
- **Wideband AGC 1/2/3** - Additional wideband compressors that can be "stacked" atop the Input AGC
- **Parametric EQ** - Six bands of fully adjustable equalization to customize the EQ curve ahead of the multiband AGC stage
- **Stereo & Bass Enhancers** - Multiband enhancer for full stereo image control. Bass Enhance for deep, tight bass
- **Multiband AGC/Limiters** - User-selectable between two and seven bands
- **Band Mix** - The final means of adjusting spectral balance before the final peak limiters



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



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™ XR

AoIP Console



AES67
Livewire+

Control with
Pathfinder 

The ultimate control of your AoIP universe.

- 6th-Gen Axia console featuring sophisticated AoIP technology
- Part of the vast Livewire+™AES67 ecosystem
- Ergonomic design featuring rugged avionics-grade components
- No overbridge for easier installation on work surfaces. No OLEDs
- 12.1-inch touchscreen built-in, no external monitor required
- Automixing and mix-minus available on each channel
- Built-in modular fanless PSUs with redundant option
- Source Profiles, Show Profiles, and User Presets
- Motorized high-quality faders
- Quasar Soft remote control & Quasar Cast remote monitoring standard
- Telos Infinity® IP Intercom Remote control
- Call controller for Telos Hybrids and VX System
- Hotkeys touchscreen
- Extensive metering built next to each fader, with high-resolution bargraphs
- Dynamic assignment of fader modules to any surface
- Up to 64 Faders
- 4 User Keys per channel strip
- Rugged and robust modular chassis frames with a compact footprint
- Available as Table-Top or Flushmount, in single or split configurations

Axia Quasar™ SR

AoIP Console



Control with
Pathfinder 

AES67
Livewire+™

Where power meets control.

- Replaces Axia Fusion® best-selling console
- 6th-Gen Axia console featuring sophisticated AoIP technology
- Part of the vast Livewire+™AES67 ecosystem
- Ergonomic design featuring rugged avionics-grade components
- No overbridge for easier installation on work surfaces. No OLEDs
- 12.1-inch touchscreen built-in, no external monitor required
- Automixing and mix-minus available on each channel
- Rugged and lightweight modular chassis frames with a compact footprint
- Available as Table-Top or Flushmount, in single or split configurations
- Built-in modular fanless PSUs with redundant option
- Source Profiles, Show Profiles, and User Presets
- Non-motorized high-quality faders
- Quasar Soft remote control & Quasar Cast remote monitoring optional
- Telos Infinity® IP Intercom Remote control
- Call controller for Telos Hybrids and VX System
- Hotkeys touchscreen
- Confidence Class Metering an all channel strips
- Fewer and larger buttons than XR that are easy to reach
- Up to 32 faders
- 1 User Key per channel strip



Axia Quasar™

Console Engine



Control with
Pathfinder 

 **AES67**
Livewire+

A native scalable AoIP powerhouse for Axia Quasar consoles.

- Super-reliable DSP platform based on proven native AoIP processing technology
- Pay for only what you need with scalable channel count
- Starts at 16 stereo channels and scales up in blocks of 16 channels, up to 64 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 Quasar™ Accessory Modules

Studio Control Modules



AES67
Livewire+

Control with
Pathfinder 

Give Your Talent the Power

- Accessory control modules for convenient talent / guest control of frequently changed options
- Motorized Fader Module is functional as an additional channel strip for your surface
- Can be flush-mounted on any flat or vertical solid surface
- Easy installation to a switch or to an open GPIO port
- Axia Console GPIO Accessory Module works with legacy Fusion consoles, the iQ Family of consoles, and Quasar XR and SR consoles



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)
- Automixer available on all channels



Axia® iQs

AES67 Mixing Console Software



AES67
Livewire+

Control with
Pathfinder

Mix control. Anywhere. On Any Device.

- iQx console capability without the physical surface
- For distributed / remote workforce, temporary studio, studios lacking physical space, & multiple users concurrently collaborating on a single mix
- First soft console controlled by a full HTML-5 interface, allowing you to control the iQs mix from anywhere, on any device!
- Your local Telos Alliance Integration Solution Consultant will provide you with expert assistance, in real-time to deploy the Axia iQs Docker Container.
- More flexible than its physical counterparts, starting with a base iQs 4-fader software module and growing in increments of 4 faders (instead of 6 or 8), up to 24 faders.
- Stay up to date with the latest iQs version and get expert service and support with TelosCare™ PLUS Service-Level Agreement (SLA)
- Supports AES67 and SMPTE 2110-30; works with any AES67, Livewire+™ AES67, or Livewire® source
- Assign any AoIP source to any channel, like having more than 16 million patch points!



Telos Alliance® AE-1000

Application Engine



AES67
Livewire+

Control with
Pathfinder

AE-1000 Features

- AE-1000 supports a single instance of iQs, which will accommodate up to 24 faders
- Start with the base 4-fader module, add faders in increments of 4 up to 24, each with instant access to any AoIP source
- 34 inputs x 24 outputs total, all AoIP, including 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
- 2 network ports, 1 for a link to the AoIP network and 1 for a local xNode or other network device
- Fanless
- PoE or DC 12 volt



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)
- Automixer available on all channels



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
- Automixer available on all channels



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
- Automixer available on all channels



Axia® RAQ

6-Fader Rack AoIP Console



AES67
Livewire+

Control with
Pathfinder

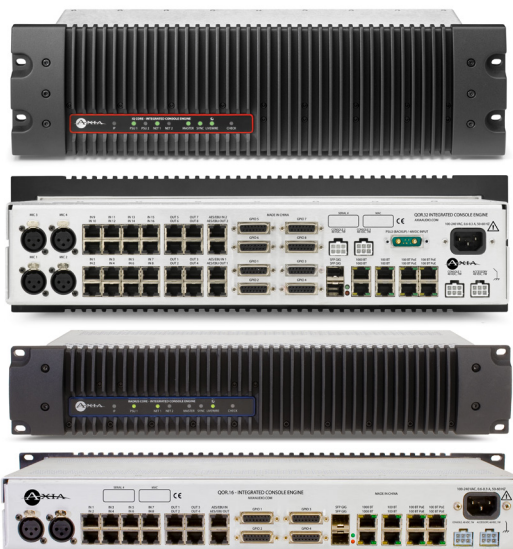
The power of Axia, in your RAQ.

- 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
- Automixer available on all channels



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



Axia® IP-Tablet

Virtual Radio Software



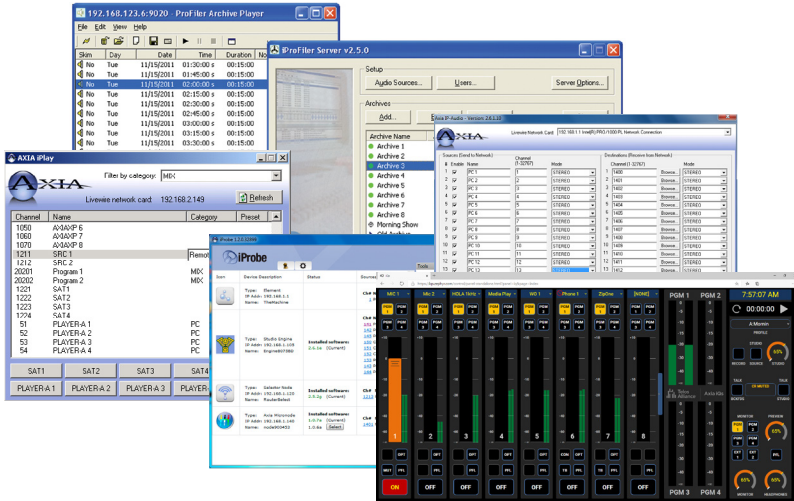
Virtualize your studio.

- 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
- Designed by Livewire® partner IP-Studio for use with HTML5-equipped gear and various Telos Alliance products
- Available for Axia PowerStation, StudioEngine, Axia iQ Family, xNodes, Omnia.9, and Telos VX Family.
- Works on Windows Tablets and PCs



Axia® Software

There's an app for that.



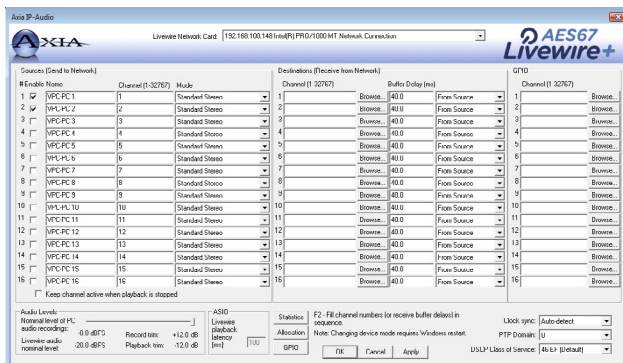
Comprehensive line of software solutions include:

- 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
- iProFile Automated Program Archiving
- iPlay Network Stream Player for Windows
- IP-Tablet Virtual Radio Software
- Axia® iQs AES67 Mixing Console Software



Axia Livewire+™ AES67 Driver

IP-Audio Driver



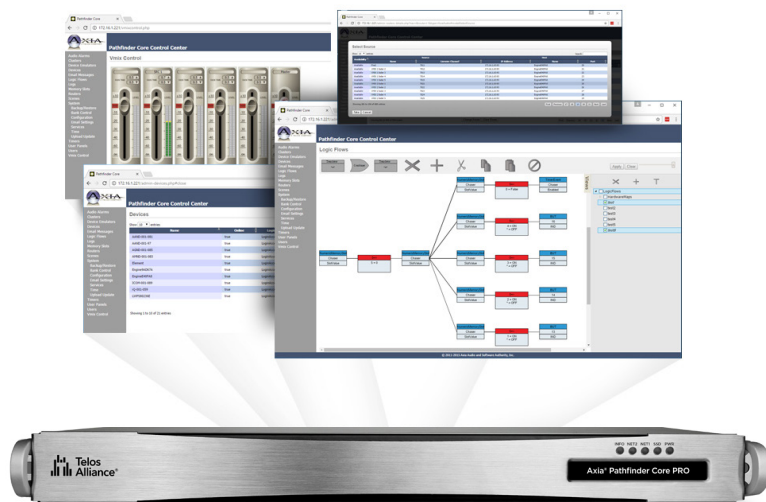
Easy IP audio, fully AES67-compliant.

- 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



Telos Alliance® xNodes™

IP-Audio Interfaces



AES67
Livewire+

Control with
Pathfinder 

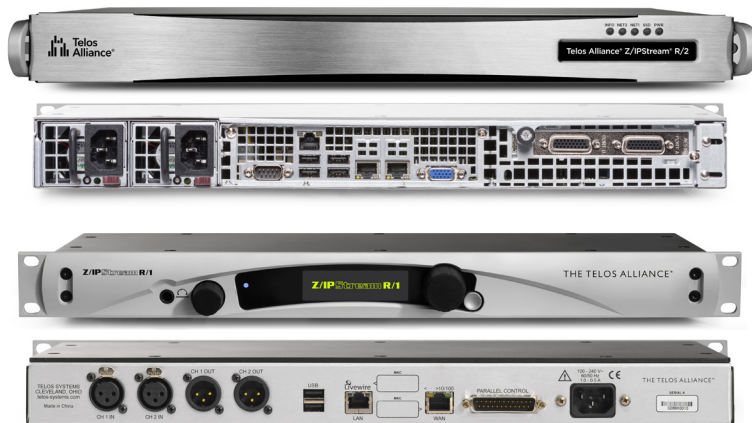
The most advanced AoIP interfaces on the planet.

- 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



Z/IPStream® Hardware

Streaming Audio Processing and Encoding



Route & Monitor with
Pathfinder 

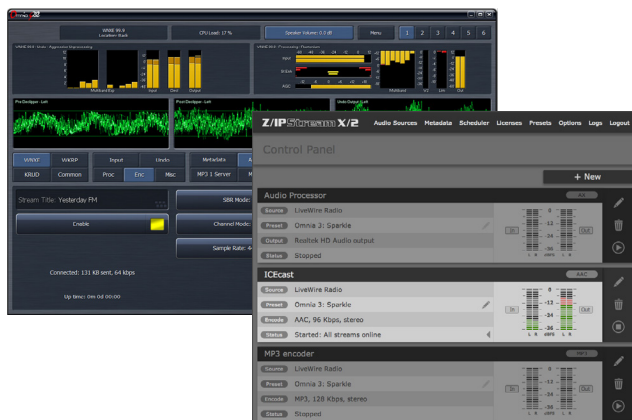
Stream like you mean it.

- 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® Program Delay Manager II

Profanity Delay, Perfected



Route & Monitor with
Pathfinder 

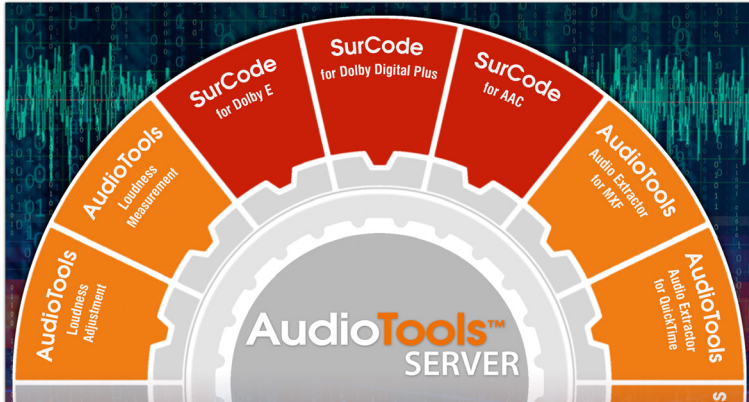
It's About Time, Again!

- PD Alert™ instantly emails time-stamped audio files whenever Dump is pressed
- Files capture what took place both on-air and off-air
- Seamlessly builds and exits delays of up to 99 seconds; supports multiple build, exit and dump modes
- Delays IP data, serial streams, and GPIO while maintaining sync to audio
- Analog, AES3 digital and Livewire+AES67 AoIP
- HTML5 GUI supports remote control from multiple devices
- Dual Network ports, redundant power, fanless design



Minnetonka AudioTools® Server

File-Based Content Production Solutions & Audio Workflow Orchestration



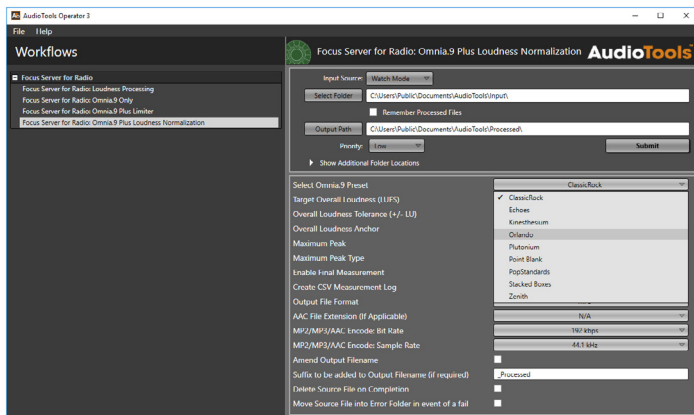
Quality-first audio workflow orchestration in flexible, focused packages of audio specialization.

- 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



AudioTools® FOCUS Server for Radio

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



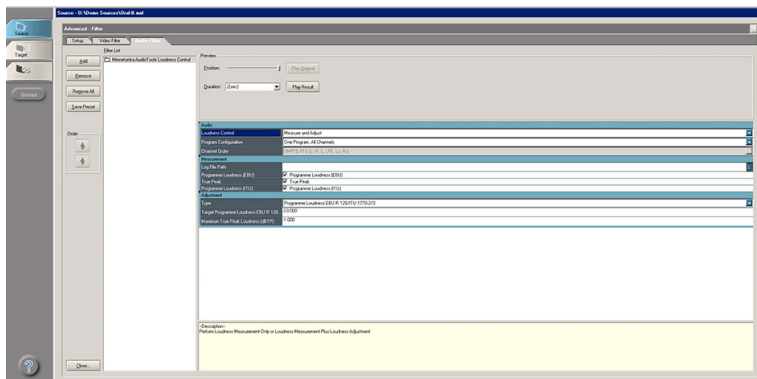
Great sound makes your podcasts stand out!

- 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



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



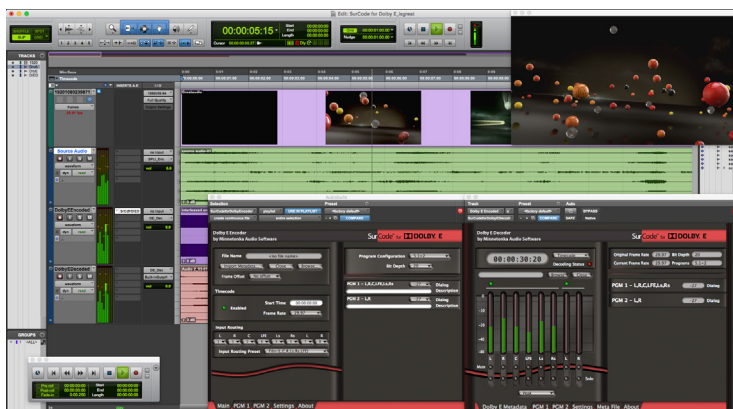
The world's most widely used surround format with Loudness Control!

- Leading brand of broadcast standard plug-ins, applications, and SDKs with certified support for all major codecs
- Available for AVID and VST-based workstations, 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



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



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



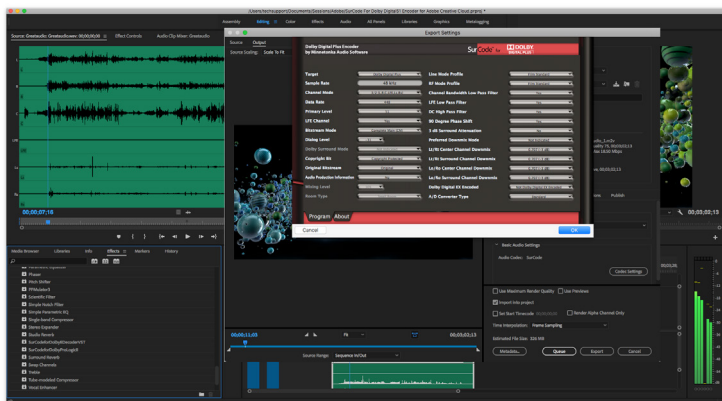
Complete, end-to-end Dolby Digital Plus surround sound plug-in suite for post- production.

- 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



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



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



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

Dolby® AC-4

nextgenTV

DOLBY ATMOS

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

- Dolby® AC-4 encoding from PCM
- Optional Dolby Digital / Dolby Digital Plus encoding from PCM
- 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
- Telos Alliance APTO™ real-time loudness control when encoding to Dolby Digital / Dolby Digital Plus
- 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



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 7.1.2 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



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



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



Linear Acoustic® AMS

Authoring and Monitoring System



Leading the way to ATSC 3.0 audio.

- Real-time authoring of MPEG-H control track
- Monitoring and emulation/rendering of MPEG-H immersive 3D audio
- Features Next Generation Audio (NGA) technologies including MPEG-H
- Simultaneously delivers discrete immersive audio output with MPEG-H control track, 5.1-ch, 2-ch, and control room monitoring outputs
- Linear Acoustic APTO® loudness control for high quality audio and compliance with global loudness standards
- Also available as a monitoring- and rendering-only solution



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 AlXpressor

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 4x4 or 2x6 channels), Analog I/O (switchable between 8x8 or 4x12 channels), Sync in
- I/O modularity via 4 interface slots
- Native Livewire+™, AES67, and SMPTE ST 2110-30, -31 support
- Livewire+™ AES67, RAVENNA, and (optional) 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



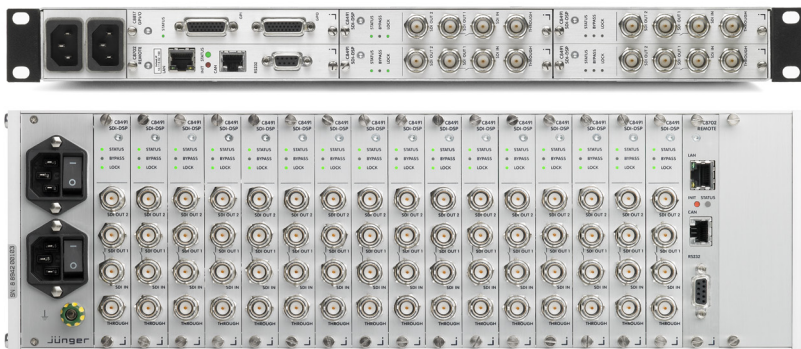
Monitoring and Dolby® workflow solution.

- 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

