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.

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# 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 |

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# ICON LEGEND

**TV**
**Application**

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**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.

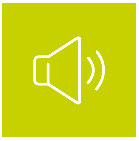
## Measurement & Monitoring

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.

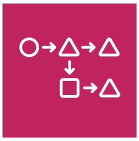


## On-Air TV Audio Processing

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 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 | 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


AES67 Livewire+™

Route & Monitor with Pathfinder

## 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


AES67 Livewire+

Route & Monitor with Pathfinder

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

- First fully-featured Cloud-based intercom system
- Supports Cloud-based media production workflows
- Panels can be deployed on any modern device, whether it be a smartphone, laptop, desktop, or tablet, through an HTML5 browser or our Android and iOS App
- 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-premises installations
- For both On-Premises 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
- Available on **Viz Now**, a tool for automating deployments of live productions in the cloud. Viz Now is a software-as-a-service (SaaS) portal hosted by Vizrt

# 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 VX® Duo
## Broadcast VoIP Phone System

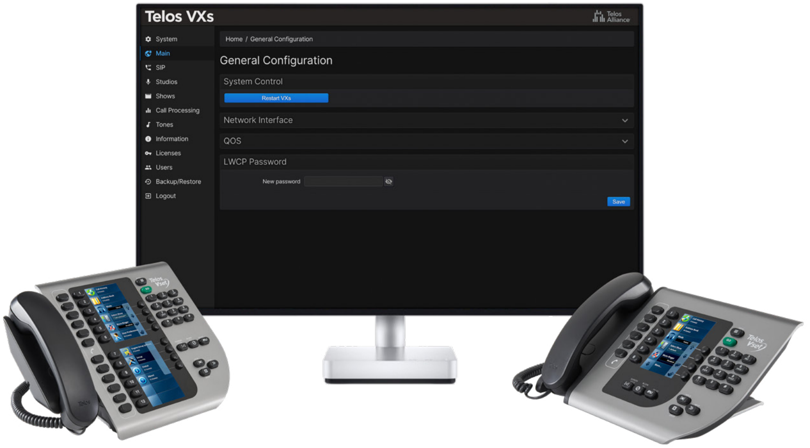

AES67 Livewire+ | Route & Monitor with Pathfinder

## Legendary VX performance, compact platform, affordable price.

- The same user experience and audio performance as our flagship VX phone system
- Includes two channels/hybrids, expandable to eight channels/hybrids via two-channel expansion licenses
- Smart AGC, three-band adaptive digital dynamic EQ, three-band spectral processor, noise gating, and wideband acoustic echo cancellation
- Compact, fanless, silent design for in-studio installation
- Place it on any flat studio or control room surface, or place up to three units side-by-side on a standard 1RU rack shelf
- Two Ethernet ports provide access to Livewire+ AES67 AoIP networks and VoIP services
- Works with Telos® VSet phones, integrated console call controllers, and VX-compatible call screening software
- Can be used with Axia® Pathfinder Core Pro Broadcast Controller to build HTML5-based control panels.

TelosAlliance.com/VXDuo

# 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
- Deployed on-premises or cloud-hosted as a Docker container, or pre-installed on the Telos Alliance AP-3000 hardware platform
- 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, plus support for SMPTE ST 2022-7
- 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/BRIISDN 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

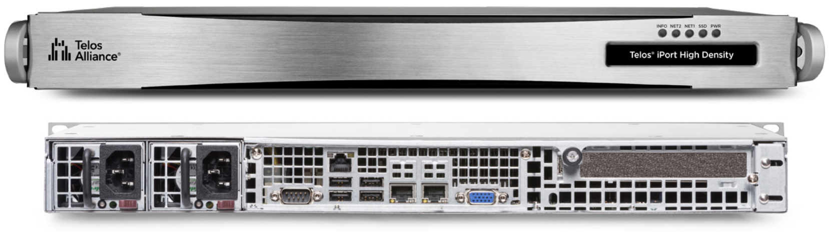

Control with Pathfinder

## 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



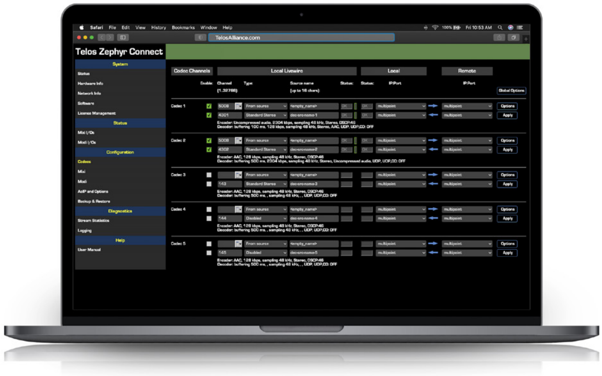

Control with Pathfinder

## 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

# Telos® Zephyr Connect

## 2-to 64-channel virtual codec, fully compatible with Telos iPort HD




AES67 Livewire+™ | Control with Pathfinder

## Advanced program distribution and facility connection virtualized.

- Transports multiple channels of PCM or coded stereo, mono, and dual-mono audio across IP networks
- Container deployment on-premises using COTS server or cloud-hosted
- Comes with 2 bi-directional stereo codecs, licensable up to 64 codecs per instance
- Each codec is independently configurable
- Up to four redundant IP stream destinations per encoder
- Unicast UDP and TCP, or UDP Multicast stream types, independently configurable per WAN stream
- Pair with Telos Alliance® xNodes and an adequately configured Ethernet switch for use as a standalone multi-stream codec
- Optional Enhanced aptX™ encoding
- Optional Content Delay feature

TelosAlliance.com/ZephyrConnect

# Omnia.11

## FM & FM+HD Audio Processor

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

- "Silvio" clipper (now shipping with Version 4.0) provides an even cleaner, brighter, and more open sound with no loudness penalty
- 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 µMPX™ 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.9sg

## Stereo Generator & Final Stage Processor

## 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

# Omnia Enterprise™ 9s

## Omnia.9 Processing, Virtualized


### Legendary Omnia.9 processing in a Windows software-based solution for AM, FM, HD/DAB, and streaming audio.

- The sound and features of Omnia.9 - including "Undo" and NfRemote - optimized for Windows server deployment.
- AM/FM version includes a single audio input ("station"), one AM processing core, one FM processing core with pre-emphasized L/R output and stereo generator with RDS, and two µMPX outputs, plus one stereo L/R processing core for HD-1, DAB, or streaming, including streaming encoders.
- HD/DAB/Streaming version includes a single audio input ("station") and one stereo L/R processing core for HD-2/3/4 or streaming, including streaming encoders.
- Optional Nielsen and Kantar audience measurement watermarking for each station.
- Available as a buyout or a subscription.
- Additional stations and options are available as field-installed license keys.

# Omnia® Forza
## Audio Processing Software

## A brand-new approach to the multiband audio processor for FM, HD, DAB, and streaming audio.

- Available as Forza FM for FM/HD-1 and Forza HDS for HD/DAB/Streaming audio applications
- Wideband AGC, 5 bands of multiband AGC, 5 bands of multiband limiting, plus soft and hard bass clippers and comprehensive bass management controls
- LUFS target-driven ITU-R BS.1770 loudness controller and True Peak limiter for compliance with streaming platform requirements (Forza HDS)
- Dual AoIP inputs for mixing two sources or for automatic failover switching (Forza HDS)
- Frank Foti-designed Silvio FM clipper and stereo generator with integrated RDS (Forza FM)
- FM/HD-1 diversity delay (Forza FM)
- Omnia's highly regarded Sensus® conditioning for low bitrate streams and HD Radio
- Carefully crafted Omnia presets ensure great-sounding audio on all types of formats and programs
- "Smart Controls" combine the adjustment of multiple, interrelated parameters into a single knob or slider
- Innovative single-page GUI welcomes less-experienced users while still providing the tools processing experts expect and need
- Browser-based HTML5 UI works on any device, including Windows and MacOS computers and Android and iOS smartphones and tablets
- Optional Kantar and Nielsen watermarking

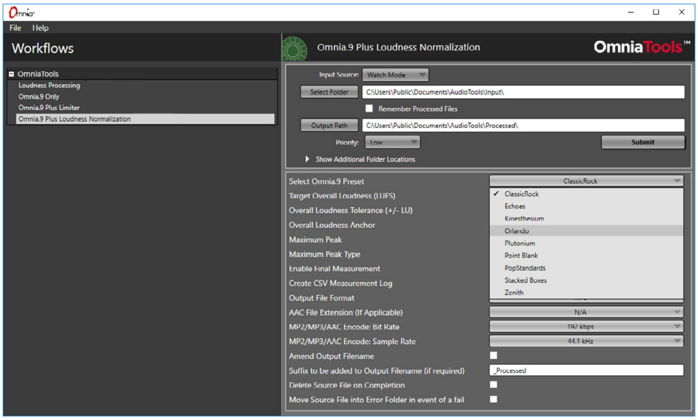
# Omnia® SST
## Audio Processing Software

## 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

# 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® 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

# 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 optional
- Telos Infinity® IP Intercom Remote control
- Call controller for Telos Hybrids and VX System
- Hotkeys touchscreen
- Up to 64 faders, each with high-resolution bar graph meters
- Dynamic assignment of fader modules to any surface
- 4 User Keys per channel strip
- Available as Table-Top or Flushmount, in single or split configurations
- New soft Call Controller switchable from 8-lines to 12-lines
- Support for backup control surfaces in Quasar Engine

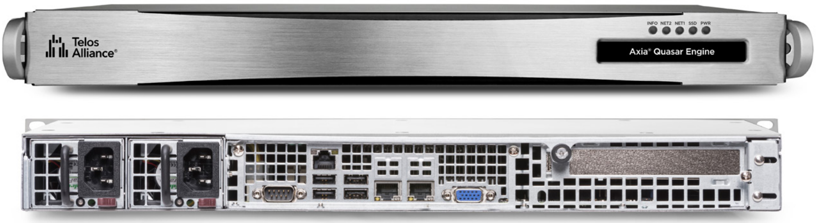
# Axia Quasar™ SR
## AoIP Console

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
- 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
- Up to 32 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
- 1 User Key per channel strip
- New soft Call Controller switchable from 8-lines to 12-lines
- Support for backup control surfaces in Quasar Engine

# 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 plus variable position insert sends and returns on every channel
- 4-band fully parametric EQ and powerful dynamics processing including talent headphone processing plus a de-esser and a low-latency peak limiter on every channel
- Filters, dynamics processing, and True Peak limiting on Program, Record, and Phone buses
- 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
- Redundant power is standard

# Axia Quasar™ Engine RPS
## Console Engine


Control with Pathfinder | AES67 Livewire+

### A Quasar Engine and a standalone mixer with no physical console.

- Based on the tried and true Quasar Engine
- Includes all of the same features and specifications as the standard Quasar Engine
- The addition of the RPS card provides the same functionality found on the Master Module inside every Quasar XR and SR surface
- Creates a standalone mixer without the need for an XR or SR hardware surface
- Quasar Soft provides the HTML5 user interface
- Quasar Cast provides low-latency audio monitoring
- Ideal for situations where a physical console is unnecessary or impractical such as remote broadcasts, small studios, or backup facilities
- Provides a low-cost backup for a main hardware surface
- Can be used in tandem with XR or SR surfaces
- Existing Quasar Engines can be field-upgraded to Quasar Engine RPS

# Axia Quasar™ Accessory Modules

## Studio Control Modules


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
- SmartKey Module adds extra programmable user keys
- 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® Altus
## Virtual Mixing Console

AES67 Livewire+

Control with Pathfinder

## Put yourself in the center of your production.

- Full function mixing for your distributed & remote workforce
- Control in a browser - no need for a physical surface
- Capable of running on a local server or data center
- Enabling concurrent collaboration from multiple users on a single live (or recorded) program
- Allowing temporary studios to be deployed anywhere, in minutes
- Controlled by computers and tablets that your team already has
- Not taking up counter space in the studio
- Providing easy low cost option for disaster recovery
- Broadcast from anywhere, on any device
- Deployed on-premises or cloud-hosted as a Docker container, or pre-installed on the Telos Alliance AP-3000 hardware platform
- Available either as a one-time buyout or as a subscription (12 month)
- Starts with a base Altus 4-fader software module and grows in increments of 4 faders (instead of 6 or 8), up to 24 faders
- Comes with 8 virtual auxiliary mixers

# 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
- Four custom Show Profile "snapshots" can be saved to instantly recall frequently used console setups
- Optional web browser control surface for remote control of the console
- Fanless power supply standard; second redundant power supply optional
- Loudness measurement
- Audio delays on input
- Pathfinder integration for AFV (audio follow video)

# Axia® iQs

## AES67 Mixing Console Software


## Mix control. Anywhere. On Any Device.

- iQx console capability without the physical surface
- Pre-installed on a 1RU Telos Alliance AE-1000 Application Engine
- 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!
- 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® StudioCore™

## Integrated Console Engine

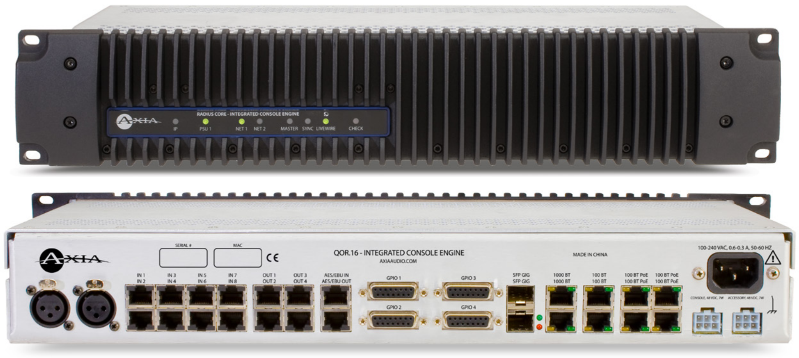

AES67 Livewire+

Control with Pathfinder

## Our newest integrated console engine for iQ, Radius, RAQ, and DESQ consoles.

- Fanless design for silent in-studio use.
- 5” color IPS LCD touchscreen display and front panel UI for local control of routing, I/O, and audio levels
- Single internal power supply (second internal PSU optional)
- Dedicated 5-port AoIP network switch with PoE
- 24-channel mixing engine with Livewire+ AES67 stream capacity of 32 inputs and 32 outputs
- 4 selectable mic/line inputs, 8 dedicated line inputs/line outputs, 3 digital inputs/outputs (user-configurable combinations of AES/EBU, USB Audio, and S/PDIF), USB Audio I/O, and 2 headphone outputs with independent DACs and built-in amplifiers
- 4 GPI/O ports
- Optional CAN board with 3 CAN connections
- Built-in audio file player via USB data port
- 8 output monitor matrix system
- Optional Core Soft license provides browser-based control of StudioCore, allowing it to function as a standalone mixer, control a connected iQ surface, and serve as an in-browser monitor for the control room monitor output.

# Axia® QOR.16
## Integrated Console Engine



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

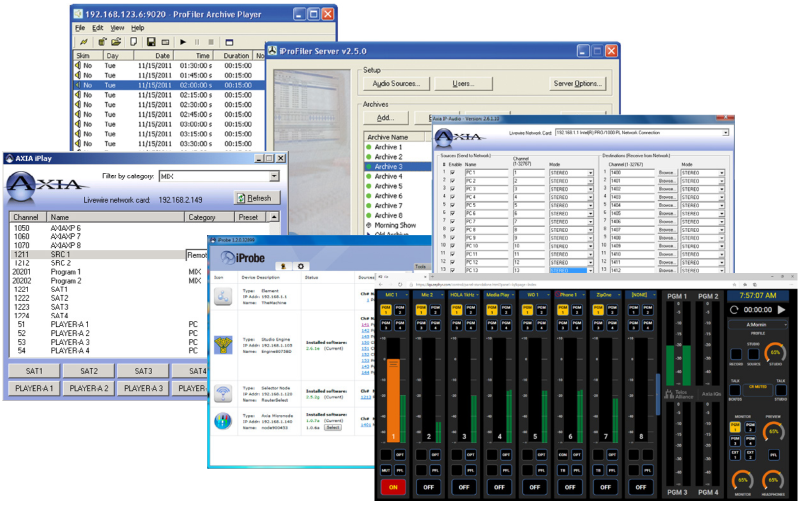
Control with
Pathfinder

### 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® Software

## There's an app for that.

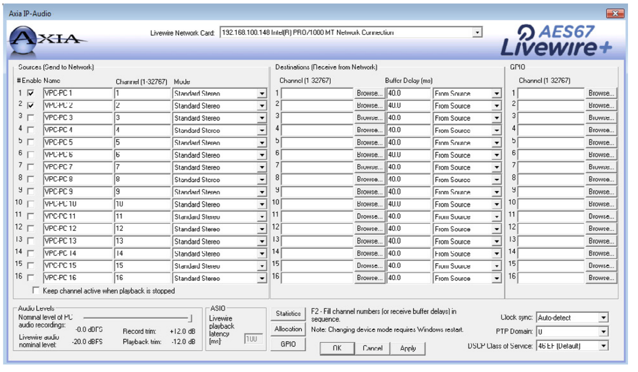



## Comprehensive line of software solutions include:

- 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® iQs AES67 Mixing Console Software

# Axia Livewire+™ AES67 Driver

## IP-Audio Driver




AES67 Livewire+

Control with Pathfinder

## 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® xNode2™

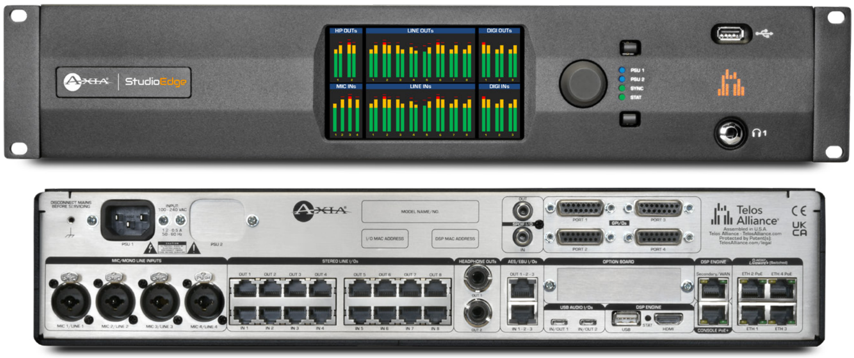
## IP-Audio Interfaces

AES67 Livewire+™ | Control with Pathfinder

## The next generation of the industry's most trusted AoIP interface.

- Latest generation AoIP interfaces
- Easy setup and configuration
- Fanless and noiseless
- High-resolution front-panel multi-function LCD color display
- Versatile mounting options, including two xNode2s in 1RU
- Available in Analog, AES/EBU, Microphone, Mixed Signal, and GPIO versions
- Livewire+ AES67 compliant, including SAP
- Dual NICs with SMPTE 2022-7 seamless protection switching and automatic failover
- Redundant power plan via AC and PoE+ (IEEE 802.3at) as supplied by compliant Ethernet switches

# Axia® StudioEdge™
## High-Density I/O Edge Platform




AES67 Livewire+

Control with Pathfinder

## Abundant I/O and an integrated network switch, all in one package.

- 2RU fanless design for silent in-studio use
- 5” color IPS LCD touchscreen display and front panel UI for local control of routing, I/O, and audio levels
- Single internal power supply (second internal PSU optional)
- Dedicated 5-port AoIP network switch with PoE
- 4 selectable mic/line inputs, 8 dedicated line inputs/line outputs, 3 digital inputs/outputs (user-configurable combinations of AES/EBU, USB Audio, and S/PDIF), USB Audio I/O, and 2 headphone outputs with independent DACs and built-in amplifiers
- 4 GPI/O ports
- Built-in audio file player via USB data port
- 8 output monitor matrix system

# Telos Alliance® Z/IPStream® X/20

## Streaming Audio Processing and Encoding


Route & Monitor with Pathfinder

## Stream like you mean it.

- All-in-one streaming encoding and processing for Windows PCs and servers
- Standard Omnia 3-band processing plus optional Omnia Forza HDS and Omnia.9 processing
- Optional Dèjá Vu upmixing providing, 5.1-channel surround sound via MPEG-Surround encoding
- Optional Nielsen or Kantar Watermarking
- HLS and MS Adaptive streaming capabilities allow media players to automatically adapt to changing network conditions by encoding at multiple bitrates, including xHE-AAC for extremely low bitrates
- Simultaneous MP3/AAC/HE-AAC encoding, compatible with Shoutcast, Icecast, Wowza, and RTMP servers
- Accepts input from any Windows sound source, including physical and virtual (AoIP) sound cards such as the Livewire driver
- Can be installed on-premises or deployed to AWS, Azure, or other cloud-hosting platforms

# Z/IPStream Hardware

## Streaming Audio Processing and Encoding


Route & Monitor with
Pathfinder

## Stream like you mean it.

- Reliable 1RU pro-grade hardware for streaming and encoding, featuring genuine Omnia processing and genuine Fraunhofer IIS codecs
- R/1 features 3-band Omnia processing
- R/20 offers the option of Omnia Forza HDS and Omnia.9 processing for more demanding situations
- R/20 offers optional Dèjá Vu upmixing, providing 5.1-channel surround sound via MPEG-Surround encoding
- R/20 features HLS and MS adaptive streaming capabilities that allow media players to automatically adapt to changing network conditions by encoding at multiple bitrates, including xHE-AAC for extremely low bitrates
- R/20 features optional Kantar and Nielsen watermarking
- R/1 accepts a single audio input, can simultaneously encode to two different bitrates, and can send the stream to up to four media servers
- R/20 accepts up to 8 audio inputs, can simultaneously encode each at multiple bitrates, and can send the stream to potentially hundreds of destinations

# 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

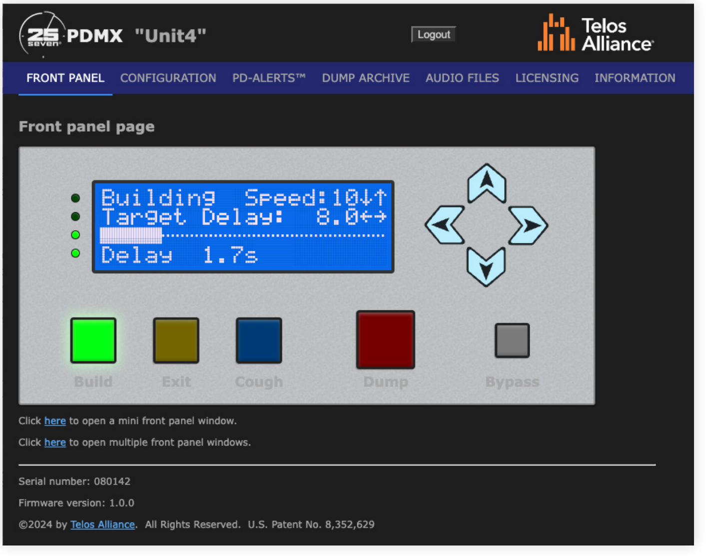

Route & Monitor with
Pathfinder

## 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

# 25-Seven® PDMX Program Delay Software

## Audio Program Delay For Virtual Workflows




AES67 Livewire+

Route & Monitor with Pathfinder

## Program Delay, Virtually

- All the functions of the renowned PDM II delay in a pure Livewire+AES67 AoIP software-based workflow
- Container-based instances enable high-density, server-based deployment
- Available on the Telos Alliance AP-3000 hardware platform
- PDAlert™ instantly emails time-stamped audio files whenever the Dump button is pressed
- Audio files capture what took place both on-air and off-air
- Seamlessly builds and exits delays of up to 99 seconds with support for multiple build, exit, and dump modes
- Delays IP metadata streams and GPIO to maintain audio sync
- HTML5 GUI supports remote control of multiple instances from multiple devices

# Minnetonka AudioTools® Server

## File-Based Content Production Solutions & Audio Workflow Orchestration

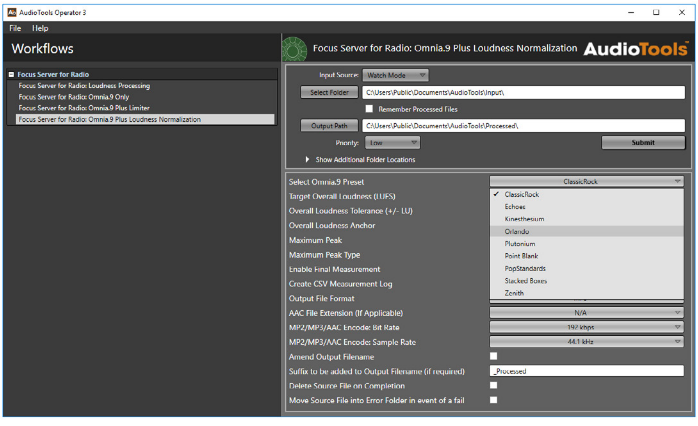

ATSC 3.0 | MPEG-H AUDIO | DOLBY ATMOS® | NEXTGEN TV

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

- Efficient file-based workflows based upon 1000+ editable custom templates or created entirely from scratch using WorkflowCreator
- 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

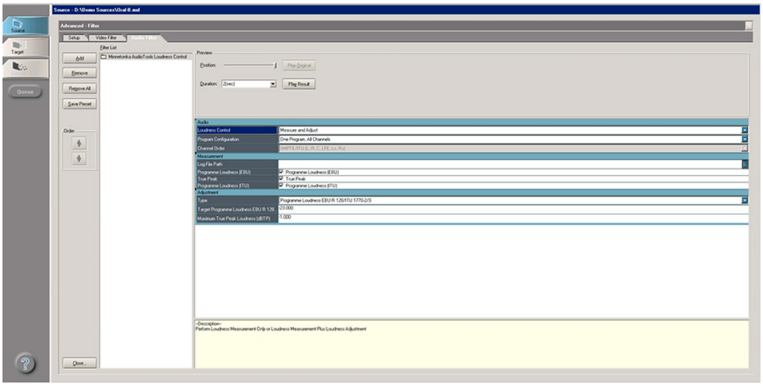



## Great sound makes your podcasts stand out!

- 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

# 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 | Dolby® AC-4 | NEXTGEN TV | DOLBY ATMOS®

## The complete audio solution for Next Generation Audio and ATSC 3.0.

- Linear Acoustic® UPMAX ISC upmixing to surround and Dolby® Atmos configurations
- Linear Acoustic APTO Loudness Control
- Input priority switching
- Optional Dolby AC-4 encoding (stereo, 5.1, Dolby Atmos)
- Optional Dolby Digital Plus / Dolby Digital Plus with Atmos, and Dolby Digital encoding (stereo, 5.1, Dolby Atmos)
- Dolby RTLL (Real Time Loudness Leveling) available when encoding to a Dolby format
- Optional Nielsen and/or Verance Aspect watermarking
- Dual 3Gb HD/SD-SDI, 5 pairs of AES3 I/O, and AES67 I/O in support of SMPTE ST 2110-30 and -31 (Level A) workflows
- Optional Quad-Link 3G SDI I/O for 4K workflows or MADI I/O (mutually exclusive)
- Dual Gigabit Ethernet connections for AES67 and remote control
- SNMP alarm and status reporting
- Browser-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.2400

## DTV Audio Processor


## 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 Plus transcoding, including Dolby Digital/ Dolby Digital Plus decoding to PCM and encoding to Dolby Digital/ Dolby Digital Plus for transmission
- Available Nielsen® or Verance® Aspect® watermark encoding
- 16 audio pairs via dual 3G/HD/SD-SDI I/O with included compensating video delay, and 8 audio pairs of AES3 I/O with reference input
- AoIP via AES67 for SMPTE ST2110-30 support as well as Livewire+
- Dual power supplies and failover bypass relays
- Extensive TCP/IP remote control

# Linear Acoustic® AERO.200

## DTV Audio Processor


## 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 Plus transcoding, including Dolby Digital/ Dolby Digital Plus decoding to PCM and encoding to Dolby Digital/ Dolby Digital Plus for transmission
- Available Nielsen® or Verance® Aspect® watermark encoding
- 16 audio pairs via dual 3G/HD/SD-SDI I/O with included compensating video delay, and 4 audio pairs of AES3 I/O with reference input
- AoIP via AES67 for SMPTE ST2110-30 support as well as Livewire+
- Dual power supplies and failover bypass relays
- Extensive TCP/IP remote control

# Linear Acoustic® AERO.20

## DTV Audio Processor


## 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)
- 16 audio pairs via dual 3G/HD/SD-SDI I/O with included video delay, 4 audio pairs via AES3 with reference input
- AoIP via AES67 for SMPTE ST2110-30 support as well as Livewire+
- 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


AES67 Livewire+ | ATSC 3.0 | MPEG-H AUDIO | NEXTGENTV

## 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

# Jünger Audio™ AIXpressor
## Processing Powerhouse


## Unparalleled I/O Flexibility and Audio Processing Power.

- Based on Jünger Audio's flexAI processing architecture
- Easily converts the most popular AoIP formats to/from traditional broadcast I/O
- Includes 4x Gigabit Ethernet connections
- Native support for Telos Alliance Livewire+ and AES67 (SMPTE ST 2110-30, -31, and -41 Metadata over Ethernet)
- Built-in interfaces for analog I/O, AES3 I/O, plus dedicated analog output
- 2x SFP slots support 64-channels of MADI or 1024-channels of tieLight when populated with optional SFP modules
- Four option card slots for additional analog, AES3, and MADI I/O, mic/line input, 3G/HD/SD SDI (including UHD), and Dante AoIP
- Licensed software modules add Jünger program and voice processing, voiceover mixing, upmixing, GfK watermarking, Dolby® E encoding/decoding, and MPEG-H 3D audio and S-ADM authoring and rendering
- Remote browser-based user interface
- Ember+ and NMOS
- SNMP

# Jünger Audio™ flexAIserver

## High-Density Server-Based Processing

## Maximum Processing Power for High Program-Count Applications.

- Based on Jünger Audio's flexAI processing architecture
- Offers server-class processing power using a Dell PowerEdge R450 for processing multiple programs
- Supports multiple SAS/SATA drives
- Use with traditional external multi-channel audio interfaces or leverage native support for AoIP streams including Telos Alliance Livewire+ and AES67 in support of SMPTE ST 2110-30, -31, and -41 Metadata over Ethernet)
- Includes 4x 1 Gigabit Ethernet connections; 2x 10 Gigabit Ethernet optional
- Optional PCIe card with 2x SFP slots support 64-channels of MADI or 1024-channels of tieLight each when populated with optional SFP modules
- Single or multi-mode fiber module for MADI or tieLight
- Micro-BNC module for MADI (tieLight not supported)
- Licensed software modules add Jünger program and voice processing, voiceover mixing, upmixing, GfK watermarking, Dolby® E encoding/decoding, and MPEG-H 3D audio and S-ADM authoring and rendering
- Remote browser-based user interface
- Ember+ and NMOS
- SNMP

# 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

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# Jünger™ Audio D*AP8 MAP

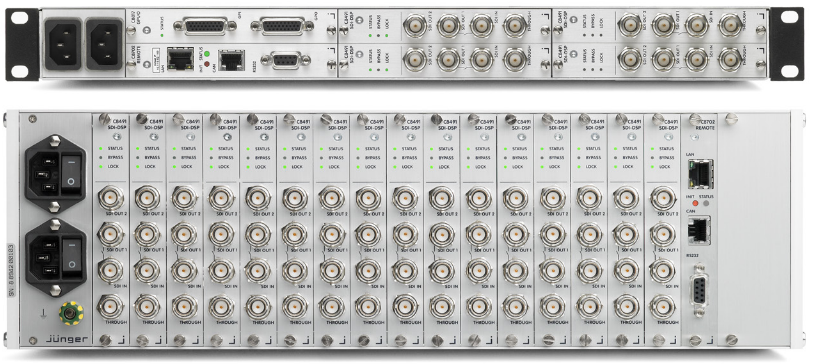
## Multichannel Monitoring Audio Processors



## 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

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