# NOW!

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## THE TELOS ALLIANCE™

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# Z/IP ONE IP CODEC

## The IP Codec that drops jaws. Not audio.

## OVERVIEW

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

# FEATURES

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

# IN DEPTH

## Advanced caller management and superior sound

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

With Z/IP ONE (the "Z/IP" stands for "Zephyr IP"), you don't have to compromise audio quality for a solid connection.

Z/IP ONE helps you get the best possible quality from public IP networks and mobile data services — even from connections behind NATs and firewalls. Telos collaborated with Fraunhofer (the developers of MP3 and many AAC breakthroughs) to develop a unique coding control algorithm that adapts to changing Internet conditions on the fly, helping you maintain quality and stability.

# IN DEPTH

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

To make certain your remote broadcast has excellent audio quality even when IP connections are not-so-excellent, Telos engineers employed a new codec based on low delay AAC. It's called AAC-ELD (Advanced Audio Coding-Enhanced Low Delay), and it produces excellent fidelity at low bitrates with nearly inaudible loss concealment and very little delay. Standard high-performance codecs are a part of the

Z/IP ONE toolkit as well, such as AAC-HE, AAC-LD, MPEG4 AAC-LC, MPEG2 AAC-LC, G.711, G.722 and even linear PCM. And if apt-X® is part of your codec cache, you can add it to your Z/IP ONE as a small extra-cost option.

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

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

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

# SPECIFICATIONS

- Input Levels (Menu Selectable): +4dBu line level, -50dBu nominal microphone level
- THD+N: < 0.03% @ +12dBu, 1 kHz sine
- Frequency Response: +/- 1dBu 25-22kHz
- Headroom: 18dB
- Dynamic Range: >87dB unweighted, >90dB "A" weighted
- Crosstalk: >80dB
- Output Clipping: +22dB
- Calculated Output Impedance: 50 Ohms, Differential
- Calculated Input Impedance: 6k Ohms, Differential
- Analog to Digital Converter: 24bits
- Digital to Analog Converter: 24bits
- Conformance and Compatibility: Conforms to N/ACIP (Open) Standards. Fully supports Session Initiation Protocol 2.0 (SIP). Compatible with TCP, UDP, DNS, Zephyr Xstream, Uncompressed PCM and other Internet Protocols.
- Codecs: SIP: G.711, G.722, MPEG Layer2, MPEG AAC, MPEG 4 AAC LC, MPEG 2 AAC LC, Linear PCM, MPEG AAC-Enhanced Low Delay (ELD), High Efficiency AAC. Optional: apt-X® from CSR.
- Input Power: 0.12A @ 120VAC 14.2 Watts.
- Power Supply: Internal, auto-ranging, 90 – 132 / 187 – 264 VAC, 50Hz/60Hz. 100 Watts.

# ZEPHYR XSTREAM ISDN CODEC

## The Best Way To Hear From There.

## OVERVIEW

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

# FEATURES

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

# IN DEPTH

## Advanced caller management and superior sound

The Telos Zephyr is the best-loved broadcast codec in the world, and for good reason: Zephyr saves you time and money. A Zephyr Xstream at your studio becomes a "universal codec," connecting with every popular ISDN codec for full-duplex, 20kHz stereo audio. Using ISDN you can transmit and receive two mono channels to and from separate locations, even transmit and decode streaming AAC and MP3 audio over Ethernet. And in the field, Zephyr Xstream is a powerful remote tool, with intuitive step-by-step operation, context sensitive help, and a simple user interface that eases operation for non-technical personnel. Zephyr pioneered the concept of the ISDN codec — which is why you'll find more Zephyrs in studios and remote kits around the world than any other codec.

# IN DEPTH

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

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

In addition to providing superior coded audio via ISDN, Zephyr Xstream's "Netcoder" mode lets you produce excellent streaming audio, using familiar HTTP, RTP/UDP, or SIP connection methods.

## Ready for your rack

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

# IN DEPTH

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

## Ready for the road

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

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

# IN DEPTH

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

# SPECIFICATIONS

## General

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

## Frequency Response

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

## THD+N

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

# SPECIFICATIONS

## Dynamic Range

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

## Send Input

- Active balanced with RF protection.

Zephyr Xstream:

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

Zephyr Xstream MX and MXP:

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

## Limiter

- Zephyr Xstream MXP:
  - Internal DSP-based AGC/limiter with Omnia® audio processing. Includes presets for music & voice, selectable per channel.
- Zephy Xstream:
  - Analog soft-clipper prevents A/D converter overload without loss of dynamic range.

## Line Bit Rates (ISDN)

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

## Bit Rates (V.35/X.21)

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

# SPECIFICATIONS

## Receive Output

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

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

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

## Inverse Multiplex/Demultiplex

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

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

## Optional V.35/X.21 Direct Digital Interface

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

## ISDN Interface

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

## LAN Interface

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

# SPECIFICATIONS

## ISDN Voice Telephone Mode

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

# ZEPHYR XPORT POTS + ISDN CODEC

## The "Go-Anywhere" Zephyr.


## OVERVIEW

Zephyr Xport is a portable broadcast codec that allows use of an ordinary analog telephone line in the field to connect with the Zephyr Xstream ISDN codec at your studio. Your remote site has ISDN? Great! Zephyr Xport connects to ISDN phone lines as well. No matter which connection you use, you'll get great-sounding remotes using the highest fidelity low-bitrate coding available: AAC plus (MPEG AAC + Spectral Band Replication enhancement) Or, choose AAC-LD coding for exceptional quality with amazingly low delay. Features include a mixer with mic and line inputs, an internal power supply and selectable DSP processing by Omnia. The built-in ISDN interface has an internal NT-1 for use worldwide.

# FEATURES

- Full duplex, high-fidelity monaural POTS + ISDN field transceiver with aacPlus and MPEG-4 AAC-LD coding.
- Superior sound: aacPlus coding gives you the best-quality audio of any POTS codec, even at low bit rates.
- POTS and ISDN capability, standard. Plug Xport into any POTS jack and dial your studio's Zephyr Xstream; connect to digital phone lines when available for ultimate flexibility.
- Built-in mixer with mic and line inputs, and selectable DSP processing by Omnia. Control knobs push-in flush with front-panel for safety during travel.
- I/O includes mic input with provision for 12-volt Phantom power, direct output of received audio, headphone and monitor mix outputs and line-level input.
- Feed PCM audio directly into Xport from any Windows®-based computer.
- AUX INTERFACE provides convenient input/output to a cellular phone's headset jack for wireless transmission, or to send audio cues from auxiliary sources (such as a secondary mixer) directly to talent's headphones. May also be used as a dedicated record output.
- MIX control lets you blend IFB audio with send audio for headphone and PA feeds. Front-panel headphone level adjustment controls rear-panel output.
- Graphical display with backlight provides monitoring of send and receive levels simultaneously.
- Navigation keys surrounding the display provide fast access to all menu settings.
- Two bi-directional contact closures for remote operation of connected devices.
- 100Base-T Ethernet port provides web-based remote control using a local computer, LAN or WAN connection.
- AUTO key gives one-button access to 100 stored Autodial numbers and 30 Location setups.
- Ultra mobile: Xport is light-weight, portable, durable, and so compact it tucks easily into most flight bags.
- Self-contained design: No wall-warts to lose or worry about; internal auto-ranging power supply works anywhere in the world.
- ISDN Interface with Internal NT-1, includes RJ-45 "S" connector for use worldwide and RJ-11 "U" connector for use in the USA and Canada.

# IN DEPTH

## POTS, or ISDN? With Xport, you're prepared for both.

When you're traveling, "small and light" is the name of the game. Of course, excellent audio quality is a must-have, too. Zephyr Xport gives you both.

With Zephyr Xport, you get ISDN audio quality with POTS economy. Xport is the world's only POTS codec that talks to the Zephyr Xstream ISDN codec. So you get the most reliable connections, and the best audio, too. A Zephyr Xport in your remote kit makes your studio's Zephyr Xstream a "universal codec," since you can Xport with either POTS or ISDN to connect with your studio. You save the money, rack space, training time and telephone lines needed to support multiple dedicated POTS and ISDN codecs -- not to mention console audio inputs and mix-minus outputs. Inside Xport's light aluminum case - only seven pounds (3kg) - you'll find a custom DSP-based modem that's optimized for Xport's high-performance audio codecs.

Zephyr Xport's friendly front-panel makes it as easy for talent to use as a cell phone. During transmission, operators can monitor send and receive levels simultaneously with modem performance using the backlit graphical display. Navigation keys surrounding the display help speed your way through graphical menu selections; the NAV key provides fast access to all menu settings, while Function, scroll and SEL keys make it easy to choose options quickly. The AUTO key gives one-button access to 100 stored Autodial numbers, plus 30 Location setups to recall settings for frequently-visited places.

Zephyr Xport's mixer couldn't be easier to use. Controls are right underneath the display screen, with Mic & Line input controls as well as a MIX control that lets you blend IFB audio with send audio for headphone and PA feeds. There's also a front-panel headphone level adjustment that controls the rear-panel output. All mixer controls are "stowable" to prevent accidental level changes or transport damage; push the knobs in to lock your settings; push them again to extend them for use.

Zephyr Xport's rear panel includes a mic input with provision for 12-volt Phantom power, direct output of received audio, headphone and monitor mix outputs, and a line-level input. The AUX INTERFACE provides convenient input/output to a cellular phone's headset jack for wireless transmission, or to send audio cues from auxiliary sources (such as a secondary mixer) directly to talent's headphones; it can also be used as a dedicated record output.

You'll also find a 10/100Base-T Ethernet port that can be used to control your Zephyr Xport remotely using a any laptop with a Web browser, or via LAN or WAN connection. You can even feed PCM audio directly into Xport from any Windows®-based computer.

# IN DEPTH

Finally, the telephone interface includes connections for an analog (POTS) phone line, a telephone handset for making voice calls, and ports for ISDN connections.

Zephyr Xport uses the highest fidelity low-bitrate coding available, AAC plus (MPEG AAC + Spectral Band Replication enhancement), along with AAC-LD (Advanced Audio Coding – Low Delay) and G.722 coding. A full-featured mixer with mic and line inputs (and selectable audio processing by Omnia) completes the package.

# SPECIFICATIONS

## Frequency Response (+0/-3dB)

- aacPlus (POTS): 20Hz - 15kHz
- MPEG-4 AAC-LD (ISDN): 20Hz - 15kHz

## THD+N

- Mixer: Loop Audio: 0.0043% @ 1kHz, 22Hz - 22kHz
- Codec: End-to-end connection at 24 kbps, aacPlus (line in to direct receive output), 0.04% @ 1kHz, 22Hz - 22kHz

## Delay

- aacPlus (POTS): <700 ms.
- MPEG-4 AAC-LD (ISDN): < 90 ms.

## Transmission Modes

- ISDN: MPEG4 AAC-LD @ 15kHz
- POTS: aacPlus @ 15kHz
- POTS Hybrid Mode: 300 - 3.4kHz

# SPECIFICATIONS

## Microphone Input

Balanced XLR female

- Clip Point:
    - Gain High: -37dBu
    - Gain Low: -27dBu
- Available Gain:
    - Gain High: 50dB
    - Gain Low: 60dB
- Impedance: ≥ 3.8K Ohms
- CMMR @ 1kHz: 75dB
- Phantom Power: 12 volt @10mA

## Line Input

- Balanced ¼" TRS
- Level: +4dBu (+20 clip)
- Impedance: ≥ 19K Ohms
- CMMR @ 1kHz: 62dB

## Auxiliary Interface

- Unbalanced input/output, ¼" phone jack
- Tip input, Ring output
- Input:
    - Level: Unity Gain to mix out, 20dBu clip point
    - Impedance: ≥ 9k Ohms
- Output:
    - Level: -26dBu nominal
    - Impedance: ≤ 12k Ohms

# SPECIFICATIONS

## Receive Direct Output

- Balanced XLR male jack
- Level: +4dBu, 20dBu clip
- Digital Operating Level: -18dBfs, nominal
- Impedance: ≤ 50 Ohms
- Noise Floor: -75.5dBu (referenced to +4dBu, no decoder lock)
- Dynamic Range: No decoder lock: 94dB
- Encoder/Decoder loopback: 91dB
- THD (Line input, 100 Hz to 10 kHz @ +4dBu): 0.05%

## Headphone Output

- Level (level control @ maximum and nominal input level at line input): ±103dBa SPL

## Monitor Mix Output

- Floating, balanced ¼" TRS
- Level: +4dBu, 20dBu clip
- Impedance: ≤ 50 Ohms
- Noise floor: -85dBu (referenced to +4dBu, Mic & Line inputs off)
- THD: 0.01% (Line input, 100 Hz to 10 kHz @ +4dBu)

## Limiter

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

## POTS Interface

- 6-Position/4-Pin miniature modular connector (RJ-11 style) with connections on the center pins (3 & 4). User-selectable configuration allows International operation.

# SPECIFICATIONS

## ISDN Interface

- Supports National ISDN-1, DMS Custom, 5ESS Custom PTP, ETS 300 (Euro ISDN), INS 64 (Japan) protocols.
- International version supports 4-wire ISDN “S” interface on an 8-position/8-pin miniature modular connector (RJ-45 style); USA & Canada version supports 4-wire ISDN “S” interface on an 8-position/8-pin miniature modular connector (RJ-45 style) and 2-wire ISDN “U” interface on a 6-position/4-pin connector (RJ-11 style).

## LAN Interface

- 10/100Base-T Ethernet port using RJ-45 connector. Supports TCP/IP (Telnet and FTP).

## Remote Control and Ancillary Data

- Remote control supported using popular web browsers over 10/100 Base-T Ethernet port using RJ-45-style connector.
- Bi-directional ancillary data: Serial connection at 9600bps; two contact closures.

### Control Ports

Two bi-directional inputs/outputs for end-to-end contact closure emulation.

- Inputs: Closure to ground; integral 10k ohm pull-up to 5 volts. External pull-ups can be used to support higher voltages.
- Outputs: Open collector, sinks up to 250mA to ground.

### Resolution

- Send Input: 24-bit.
- Receive Output: 24-bit.

### POWER SUPPLY

- Auto-ranging, 90 – 132 / 187 – 264 VAC, 50Hz/60Hz.
- Power consumption: 75 Watts.

# SPECIFICATIONS

## DIMENSIONS

- 17 1/8” (43cm) wide
- 12 1/2” (32cm) deep
- 3 1/2” (9cm) high

# ZEPHYR IPORT PLUS MULTI-CODEC GATEWAY

## 8 MPEG Codecs in a Livewire Gateway

## OVERVIEW

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

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

Coding algorithms include AAC, AAC-LD, HE-AAC (plus v2), MP2, MP3, linear, and optional aptX® Enhanced*. Bit rates range from 24 to 320 kbps for MPEG codecs, plus standard fixed rates for aptX and linear to over 2 Mbps. In addition, iPort offers dual, parallel-path end-to-end streaming for ultra-reliability and redundancy. For network operators, a unique Content Delay feature allows independent delay of any or all coded audio channels for up to six hours.

# FEATURES

- Connects two or more Livewire-equipped facilities over a wide-area network with QoS.
- Transmits/receives as many as 8 bi-directional channels, each with GPIO and PAD.
- Encodes up to 16 streaming audio channels for Internet transmission to the public, or for internal distribution, via SHOUTcast, Steamcast or compatible stream replication server.
- Wide choice of genuine Fraunhofer codecs, including Standard AAC, high-efficiency AAC-HE (aacPlus), AAC-HEv2, low-delay AAC-LD, and MP3, with a choice of bit rates from 24 kbps to 320 kbps, definable per stream.
- Optional aptX® Enhanced audio coding may be ordered at time of purchase or added later, as required.
- When used as part of a Livewire network, allows audio from remote facilities to be used as if it were a local source.
- Two 5-input Virtual Mixer (VMIX) channels each allow combining up to 5 networked Livewire audio streams on a single channel.
- Eight Virtual Mode (VMODE) channels allow audio to be split into left/right channels, summed L+R, and more, prior to encoding and transmission.
- Content Delay option enables delayed playout of any or all selected receive audio channels, along with their ancillary data, for up to six hours. Each playback delay time is independently configurable on a per-channel basis, making Zephyr iPort PLUS ideal for network operators, program distribution networks, or delayed playout of received audio at network-affiliated stations.
- Remote control/configuration via any computer with a standard Web browser.
- Separate LAN and WAN jacks help ensure network security.
- Fanless, convection-cooled DSP-powered platform with dual-redundant, auto-switching power supplies for maximum uptime. Power supply modules are field-replaceable in minutes.

# IN DEPTH

## Powerful, advanced program distribution and facility connection.

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

# IN DEPTH

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

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

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

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

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

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

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

# SPECIFICATIONS

## Audio

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

- Analog Line Inputs
  - Input Impedance: >40 k ohms, balanced
  - Nominal Input Range: Selectable, +4 dBor -10dBv
  - Input Headroom: 20 dB above nominal input
- Analog Line Outputs
  - Output Source Impedance: <50 ohms balanced
  - Output Load Impedance: 600 ohms, minimum
  - Nominal Output Level: +4 dBu
  - Maximum Output Level: +24 dBu
- Digital Audio Inputs and Outputs
  - Reference Level: +4 dB(-20 dB FSD)
  - Impedance: 110 Ohm, balanced (XLR)
  - Signal Format: AES3 (AES/EBU)
  - AES3 Input Compliance: 24-bit with selectable sample rate conversion, 32 kHz to 96 kHz input sample rate capable.
  - AES3 Output Compliance: 24-bit
  - Digital Reference: Internal (network timebase) or external reference 48 kHz, +/- 2 ppm
  - Internal Sampling Rate: 48 kHz
  - Output Sample Rate: 44.1 kHz or 48 kHz
  - A/D Conversions: 24-bit, Delta-Sigma, 256x oversampling
  - D/A Conversions: 24-bit, Delta-Sigma, 256x oversampling
- Frequency Response
  - Any input to any output: +/- 0.5 dB, 20 Hz to 20 kHz

# SPECIFICATIONS

## Network

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

## Codecs

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

## Power

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

# VX BROADCAST VoIP

## The Whole-Plant Broadcast Talkshow System


## OVERVIEW

VX is the world's first VoIP (Voice over IP) talkshow system — a broadcast phone system that's so powerful it can run all of the on-air phones for your entire plant, but economical enough for stations with just two or three studios. VX connects to traditional POTS and ISDN telephone lines via standard Telco gateways. But it also connects to VoIP-based PBX systems and SIP Trunking services to take advantage of low-cost Internet-delivered phone services.

VX also weds modern networking to the remarkable power of digital signal processing. VX uses Ethernet as its connection backbone, significantly cutting the cost of phone system installation, maintenance and cabling. Ethernet is a powerful, yet simple way to share phone lines among studios and connect system components. This also makes VX naturally scalable, capable of serving even the largest of facilities — while remaining surprisingly cost-effective for even single stations with more modest needs.

# OVERVIEW

VX plugs right into Axia IP-Audio networks, connecting multiple channels of audio and control via a single CAT-5 cable. Don't have an IP-Audio network yet? No problem; optional VX interfaces break out audio into analog and digital formats, along with GPIO logic commands. And with informative VSet phones, talent finds it easier than ever to take control of their callers, moving and sharing lines between studios at the touch of a button.

# FEATURES

- World's first VoIP telephone system designed and built specifically for broadcasting.
- Works with POTS, T1/E1, ISDN and SIP Trunking telco services for maximum flexibility and cost-savings.
- Standards-based SIP/IP interface integrates with most VoIP-based PBX systems to allow transfers, line sharing and common telco services for business and studio phones.
- Standard Ethernet backbone provides a common transport path for both studio audio and telecom needs, resulting in cost savings and a simplified studio infrastructure. Connection of up to 100 control devices (software or hardware) is possible.
- Modular, scalable system can be easily expanded to manage a network of up to 20 studios, each with a dedicated Program-On-Hold input – truly a "whole-plant" solution for on-air phones.
- System capacity of up to 48 standard phone lines; supports up to 250 SIP numbers.
- Up to 16 hybrids, with as many as 48 active calls (up to 4 per hybrid), may be placed on-air concurrently.
- Each call receives a dedicated hybrid for unmatched clarity and superior conferencing.
- Native Livewire integration: one connection integrates caller audio, program-on-hold, mix-minus and logic directly into Axia AoIP consoles and networks.
- Connect VX to any radio console or other broadcast equipment using available Analog, AES/EBU and GPIO interfaces. Audio interfaces feature 48 kHz sampling rate and studio-grade 24-bit A/D converters with 256x oversampling.
- Powerful dynamic line management enables instant re-allocation of call-in lines to studios requiring increased capacity.
- VSet phone controllers with full-color LCD displays and Telos Status Symbols present producers and talent with a rich graphical information display. Each VSet features its own address book and call log.

# FEATURES

- Drop-in modules can integrate VX phone control directly into your Axia mixing consoles.
- Included XScreen Lite screening software with built-in soft-phone allows a "phone" connection on any networked PC. Integrated recorder/editor simplifies recording of off-air conversations.
- Clear, clean caller audio from fifth-generation Telos Adaptive Hybrid technology, including Digital Dynamic EQ, AGC, adjustable caller ducking, and send- and receive-audio dynamics processing by Omnia.
- Wideband acoustic echo cancellation from Fraunhofer IIS completely eliminates open-speaker feedback.
- Support for G.722 codec enables high-fidelity phone calls from SIP clients.

# IN DEPTH

## VoIP for Broadcast. From Telos, Naturally.


VX is the world's first VoIP (Voice over IP) talkshow system. It's incredibly powerful, very flexible, and highly scalable — a powerful whole-plant broadcast phone system that's also economical enough for stations with just two or three studios.

VoIP has already taken the business world by storm, increasing the flexibility of office phone systems and PBXs while simultaneously lowering maintenance and equipment costs. In fact, most Fortune 500 companies have replaced their older PBX systems with VoIP for just these reasons. There's no reason broadcasters shouldn't take advantage of this cost-saving technology as well.

# IN DEPTH

Sure, VX can connect to traditional POTS and ISDN telephone lines, using standard Telco gateways. But we've built VX around the VoIP standard, so that it can connect natively to VoIP-based PBX systems and modern SIP Trunking services, allowing you to take advantage of low-cost Internet-delivered phone services. In addition to cost savings from digital phone service provisioning, VX significantly eases the cost of installation, maintenance and cabling by using standard Ethernet as its data backbone.

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

## Why VoIP for broadcast?

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

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

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

# IN DEPTH

## VX Components

### The VX Engine


The VX Engine, a fan-free 2RU rack-mount device with enormous processing power, is the heart of the system. It provides all the call control and audio processing needed for the system, and supports up to 30 active calls on-air simultaneously, across as many as 20 studios. Its two Gigabit Ethernet ports provide a cost-effective interface to both telephone lines and studio audio via proven Livewire AoIP. VX is Web-based, so remote control and configuration are a snap — engineers can work with it from any place they can get online.

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

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

You'll notice that there are no audio I/O or telco ports on the VX Engine. All connections to the Engine are via the two Ethernet jacks that connect to your system's Ethernet switch to support a wide variety of peripherals: telephone lines, Livewire studio audio, VSet phones, VX Producer PC applications, console-integrated controllers, etc.

# IN DEPTH

For traditional phone services, you can choose standard telco gateways from Patton, Cisco, Grandstream and others to connect to T1/E1, ISDN, and POTS providers. And, if you have a VoIPbased PBX or SIP Trunking telco service, the VX uses standard SIP (Session Initiation Protocol) and RTP (Real-time Transport Protocol).

## The coolest broadcast phone controllers ever

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

### VSet12

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

# IN DEPTH

## VSet6


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

## VSet1


VSet1 is a single-line phoneset that's perfect for news booths or production facilities where multiple

# IN DEPTH

lines are not required. As with its multi-line brethren, VSet1 has a big LCD color display that helps users intuitively navigate through available options, and provides information such as Caller ID time ringing and time on-hold, and even screener comments from software screening applications."

## On-Console Control


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

They can: IP-Audio networking technology provides the ideal way to integrate broadcast phones into the on-air console — the control center of every studio. VX connects directly to Axia Audio mixing consoles using Livewire IP-Audio to eliminate the cost and complexity of old-style inputs, outputs, and

# IN DEPTH

mix-minuses. Multiple phone lines – each with a dedicated hybrid – can automatically map to individual console faders for complete control of caller audio. And users enjoy seamless console integration, with phone controls right on the board so that talent can dial, answer, screen, and drop calls without ever diverting their attention from the console. Information about line and caller status can be displayed right on the console as well.

There are plenty of other advantages to melding phones with consoles. Like ease of installation: IP-Audio consoles with built-in phone controllers don't need any additional wires or connections. Their control signaling, caller audio and backfeeds ride on the network connection that's already there. Bringing caller audio into the IP-Audio domain makes it routable like any other audio source. With the Virtual Mixers built into Axia consoles, you could even choose to dynamically conference multiple lines and control their gain with a single fader. And since the console now communicates directly with the phone hybrid, mundane tasks such as mix-minus generation, starting recording devices, and playback of recorded off-air  conversations can all be automated.

## Audio Interfaces


VX Audio and Logic Interfaces let you connect VX to any non-networked radio console or other broadcast equipment, using standard analog or AES/EBU interfaces. A GPIO Logic interface provides control logic where needed.

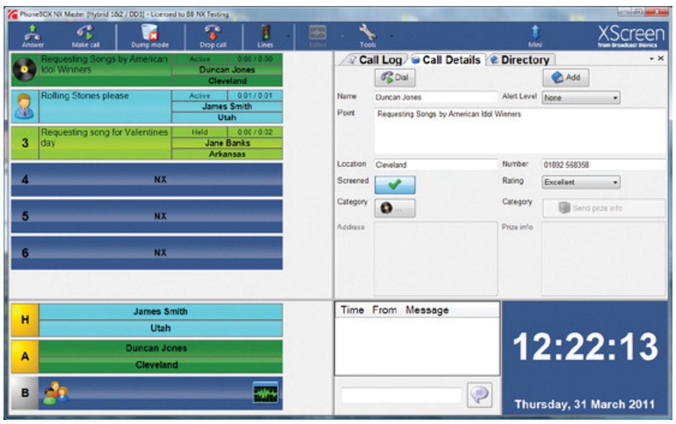
The VX Analog audio interface has eight balanced stereo inputs and eight balanced stereo outputs, presented on easy-to-install RJ-45 connectors. The inputs are switchable to accommodate consumer level -10dBv or professional level +4dBu. Outputs are short-circuit protected and capable of delivering up to +24dBu before clipping. Superior performance specs include 102dB of dynamic range, <0.005% THD.

The VX AES/EBU audio interface provides eight digital AES3 inputs and outputs, each on a separate RJ-45 connector. Studio-grade performance specs, like 138dB of dynamic range and <0.0003% THD.

# IN DEPTH

Each VX GPIO logic interface has eight assignable logic ports. Each port contains 5 opto-isolated inputs and 5 opto-isolated outputs, which can be associated with audio input peripherals and/or output destination devices to provide machine start/stop pulses, lamp drives, and transport controls. Once a port is configured to be associated with a particular device, it automatically activates with that device.

## Call Screening Software Included




VX comes complete with XScreen Lite call screening software, from Broadcast Bionics. XScreen Lite's interface gives screeners and hosts tons of information and control using sophisticated visual talkback, including a drag and drop database of all calls for your show as well as a phonebook and visual warnings for persistent or nuisance callers. A fully-functional copy of XScreen Lite is provided to all Hx6 customers, but an upgrade to the full XScreen client software adds even more features, including extended call history, an enhanced phonebook, prize management, powerful GPIO functionality and more. XScreen, deployed as part of a Livewire network, also enables call recording, editing and console integration directly over the network.

# SPECIFICATIONS

## General

- Telos 5th-generation Adaptive Digital Hybrids.
- Maximum number of phone lines: 48, when used with a-Law or u-Law codecs for VoIP lines. (Higher quailty codecs, such as G.722, consume more system resources and result in a decreased number of total lines available.)
- Maximum number of SIP numbers: 250
- Maximum active on-air calls: 48
- Maximum number of simultaneous audio connections (Livewire I/O channels): 16 systemwide
- Maximum on-air calls on one fader: 4

### Analog Inputs (with VX Interface):

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

### Analog Outputs (with VX Interface):

- Output Source Impedance: <50 Ohms balanced
- Output Load Impedance: 600 Ohms, minimum
- Nominal Output Level: +4 dBu
- Maximum Output Level: +24 dBu

### Digital Audio Inputs And Outputs

- Reference Level: +4 dBu (-20 dB FSD)
- Impedance: 110 Ohm, balanced (XLR)
- Signal Format: AES-3 (AES/EBU)
- AES-3 Input Compliance: 24-bit with selectable sample rate conversion, 32 kHz to 96kHz input sample rate capable.

# SPECIFICATIONS

- AES-3 Output Compliance: 24-bit Digital Reference: Internal (network timebase) or external reference 48 kHz, +/- 2 ppm
- Internal Sampling Rate: 48 kHz
- Output Sample Rate: 44.1 kHz or 48 kHz
- A/D Conversions: 24-bit, Delta-Sigma, 256x oversampling
- D/A Conversions: 24-bit, Delta-Sigma, 256x oversampling
- Latency <3 ms, mic in to monitor out, including network and processor loop

### Frequency Response

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

### Dynamic Range

- Analog Input to Analog Output: 102 dB referenced to 0 dBFS, 105 dB "A" weighted to 0 dBFS
- Analog Input to Digital Output: 105 dB referenced to 0 dBFS
- Digital Input to Analog Output: 103 dB referenced to 0 dBFS, 106 dB "A" weighted
- Digital Input to Digital Output: 138 dB

### Total Harmonic Distortion + Noise

- Analog Input to Analog Output: <0.008%, 1 kHz, +18 dBu input, +18 dBu output
- Digital Input to Digital Output: <0.0003%, 1 kHz, -20 dBFS
- Digital Input to Analog Output: <0.005%, 1 kHz, -6 dBFS input, +18 dBu output

### Crosstalk Isolation, Stereo Separation and CMRR

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

# SPECIFICATIONS

## VX ENGINE

### IP/Ethernet Connections

- One 100BaseT/gigabit Ethernet via RJ-45 LAN connection
- One 100BaseT/gigabit Ethernet via RJ-45 WAN connection

### Processing Functions

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

### Power Supply AC Input

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

### Operating Temperatures

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

# SPECIFICATIONS

### Studio Audio Connections

- Via Livewire IP/Ethernet. Each selectable group and fixed line has a send and receive input/output.
- Each studio has a Program-on-Hold input.
- Each Acoustic Echo Canceller has two inputs (signal and reference) and one output.
- Livewire-equipped studios may take the audio directly from the network. VX Interface nodes are available for pro analog and AES3 breakout.

## TELCO CONNECTIONS

- Audio: standard RTP. Codecs: g.711u-Law and A-Law, and g.722.
- Control: standard SIP trunking

# NX12 & NX6 TALKSHOW SYSTEMS

## Clean, consistent call quality. Every line – every time.

## OVERVIEW

Telos Nx12 and Nx6 Talkshow Systems are advanced multi-line telephone systems which each feature four independent digital hybrids. Callers can be managed in separate "screened" and "hold" queues, and taken to air on a dedicated hybrid or "button mashed" for quick on-air conferencing. Each hybrid has symmetrical wide-range AGC by Omnia, plus long-tail-echo cancellation to deal with VoIP and cellular calls.

Nx systems can also be configured for dual-studio operation, in which two separate studios may share or split incoming phone lines: each studio receives its own independent caller queue and command of two of the Nx system's hybrids, maximizing both your technology investment and the flexibility of your Nx system.

Nx systems may be ordered with AES/EBU or Analog I/O, and include Livewire IP-Audio connectivity, standard. Phone line support includes ISDN S or U interface and/or universal POTS. Each NX system includes XScreen Lite call screening software from Broadcast Bionics for easy caller management.

# FEATURES

- 12- and 6-line talkshow systems, each with four (4) DSP-powered telephone hybrids. Analog or AES/EBU I/O connection allows use of two hybrids; connection to Livewire IP-Audio Network or Telos VX Interface enables use of additional two hybrids for a total of four.
- Works with POTS (analog) telephone lines, or ISDN connections (two BRI channels per connector). Supports ISDN U (2B1Q 2-wire, USA & Canada) and ISDN S (4-wire, outside North America) digital connections.
- Standard audio connections includes analog and Livewire AoIP I/O. AES/EBU connections are optional at extra cost.
- Native Livewire connection integrates caller audio, program-on-hold, mix-minus and logic directly into Axia AoIP consoles and networks via single CAT-5 connection.
- Dual-Studio mode allows one Nx system to serve two studios, each with assigned or shared phone lines. Two separate Program-On-Hold inputs support different audio feeds in dual-studio mode.
- Default configuration supports connection of up to 4 control surfaces (Desktop Director phones, screener PCs or drop-in modules for Axia consoles). Optional accessory power supplies allow connection of up to 8 control surfaces.
- Telos 3rd-generation adaptive digital hybrids include send-to-caller Acoustic Echo Cancellation, high-pass filter, frequency shifter, AGC/limiter, Program-on-Hold AGC/limiter and sample-rate conversion for AES connections.
- Receive-from-caller processing includes a high-pass “hum” filter, smart AGC/platform leveler, noise gate, Telos DDEQ (Digital Dynamic Equalization) 3-band adaptive spectral control processor, and sample rate conversion for AES connections.
- Trans-Hybrid Loss of > 55 dB.
- Built-in Web server for configuration and software updates, plus Telnet capability for command line control and diagnostics.
- GPIO for recording, delay, tally functions provided on standard 15-pin D-Sub connector with 5 status outputs and 4 control inputs. Livewire GPIO available when connected to Livewire IP-Audio systems.
- Desktop Director controllers with dot-matrix LED Status Symbols display report line and caller status at a glance.
- Drop-in modules can integrate Nx phone control directly into your Axia mixing consoles.
- Includes XScreen Lite call screening software from Broadcast Bionics.

# IN DEPTH

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

Find a radio facility where caller audio quality is important, and chances are you'll find a Telos Nx broadcast phone system. Nx12 twelve-line and Nx6 six-line systems deliver the cleanest, most consistent call quality possible from even the most challenging calls. Nx systems combine multiple advanced telephone hybrids (each with their own AGC, noise gate, and caller override dynamics) with Telos' famous Digital Dynamic EQ, a sophisticated multiband equalizer which analyzes and adjusts received audio spectral characteristics so that calls sound smooth and consistent despite today's wide variety of phone sets and connection types. Nx systems also feature caller audio sweetening by Omnia, special echo cancellation to tame tricky VoIP and cellular calls, and anti-feedback routines to tackle the acoustic feedback that plagues open speaker applications.

Both talent and producers love working with Nx, thanks to unique Telos features that help make shows run smoother, faster and more error-free. Exclusive Telos Status Symbols visual call management icons clearly show line and caller status; special-function keys on Desktop Director phones let you command GPIO-style outputs for push-button command of profanity delay systems and recorders.

Nx6 and Nx12 systems can be ordered to work with your choice of POTS or ISDN (BRI) phone lines, and work with the Telos Desktop Director phoneset and drop-in modules for Axia Audio mixing consoles. The Ethernet port makes short work of connecting to PCs with call screening software (and provides a one-click connection to Axia IP-Audio networks, as well). Setup is easy too: each Nx system has a built-in Web server for fast configuration and remote monitoring functions.

# IN DEPTH

Nx6 works with up to 6 telephone lines, and Nx12 with up to 12 lines, Each have four hybrids, for extra flexibility in fast-paced talk environments; two hybrids may be used with analog or AES I/O while connecting Livewire I/O allows use of all four advanced hybrids. Nx systems feature a useful "dual studio mode" that allows a single system to power phones for two studios at once, each with its own Program-On-Hold input. Out of the box, you can connect 4 control surfaces (phones, screener PCs or console directors) for flexibility in commanding your calls – or up to 8 surfaces using accessory power supplies.

Nx6 and Nx12 work with any mixing console. Both come standard with analog I/O, and Nx12 can be outfitted with an optional AES interface that allows direct access to all four hybrids individually. But should you happen to have Axia IP consoles, Nx talkshow systems connect directly using just a CAT-5 cable. That one connection takes care of all audio I/O, on-hold inputs, hybrid control and GPIO. Drop-in modules available for Axia Element consoles let users easily take control of their Nx6 or Nx12 system right from the console. Choose the Call Controller module with onboard hybrid controls with Status Symbol displays, or the standard four-fader module for "fader-per-hybrid" European operating style.

Most importantly, Nx systems make your call-in segments sound great thanks to a host of sophisticated DSP and audio processing routines. There's our famous Digital Dynamic EQ and a symmetrical wide-range AGC and noise gate from the audio processing experts at Omnia Audio for caller consistency. Adjustable caller ducking to help your hosts keep control of the conversation. And a sophisticated pitch shifter and studio adaptation routines that help keep feedback from appearing when taking calls with open speakers. Additionally, Nx6 and Nx12 feature Caller ID on both analog POTS and ISDN telephone lines, and feed that data over Ethernet to your call screening application.

Nx systems will work happily with either POTS, ISDN-S, or ISDN-U lines — just tell us when you order. Or, split the difference: with Nx12, you can specify half POTS, half digital line interfaces.

## Desktop Director Phone Controllers

# IN DEPTH

Telos Desktop Directors are sophisticated, yet easy-to-use phonesets that make fast-paced production a snap. Desktop Directors help you screen calls quickly and efficiently using deluxe features like the built-in handset and speakerphone — there's even a port to connect your screener's favorite headset. And caller management has never been simpler, thanks to intuitive Telos Status Symbols — clear, easy to read graphical icons that convey immediate information about line availability, on-hold and ready-for-air queue status with just a glance.

## On-Console Control

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

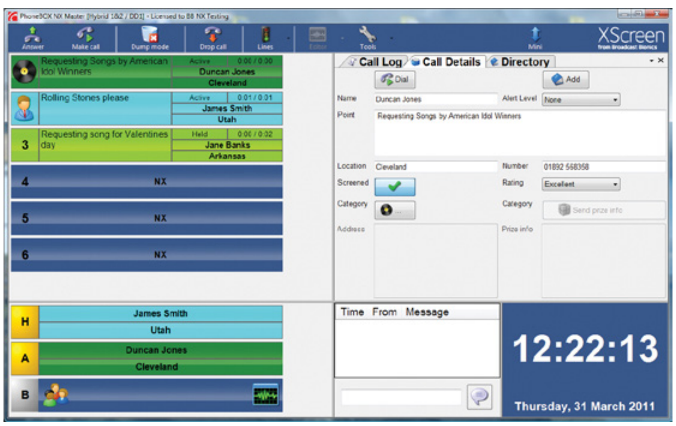
Now, they can. Nx12 and Nx6 can connect directly to Axia mixing consoles using Livewire IP-Audio to eliminate the cost and complexity of old-style inputs, outputs, and mix-minuses. IP-Audio networking technology provides the ideal way to integrate broadcast phones into the on-air console — the control

# IN DEPTH

center of every studio. Users enjoy seamless console integration, with phone controls right on the board so that talent can dial, answer, screen, and drop calls without ever diverting their attention from the console. Information about line and caller status can be displayed right on the console as well.

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

## Call Screening Software Included




Nx6 and Nx12 systems come complete with XScreen Lite call screening software, from Broadcast Bionics. XScreen Lite's interface gives screeners and hosts tons of information and control using sophisticated visual talkback, including a drag and drop database of all calls for your show as well as a phonebook and visual warnings for persistent or nuisance callers. A fully-functional copy of XScreen Lite is provided to all Nx clients, but an upgrade to the full XScreen client software adds even more features, including extended call history, an enhanced phonebook, prize management, powerful GPIO functionality and more. XScreen, deployed as part of a Livewire network, also enables call recording, editing and console integration directly over the network.

# SPECIFICATIONS

## General

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

## Analog Inputs:

- Send Analog Inputs: 2x
- Program-on-Hold Analog Inputs: 2x
- Connector : XLR Female, Pin 2 High (Active Balanced with RF Protection)
- Input Level: Adjustable from -10dBV to +8 dBu (nominal)
- Analog Clip Point : +24 dBu
- Impedance: Bridging, > 10K Ohms
- Analog-to-Digital Converter Resolution: 20 bits

## Analog Outputs:

- Receive Analog Outputs: 2x
- Connector: XLR Male, Pin 3 High
- Output Level: Adjustable from -10dBV to +8 dBu (nominal)
- Impedance: <60 ohms
- Digital-to-Analog Converter Resolution: 24 bits
- Headroom Before Clipping: 20 dB headroom above 4dBU nominal levels

# SPECIFICATIONS

## AES3 Digital Inputs/Outputs (optional)

The AES input/output option module substitutes 2x input and 2x output AES XLRs in place of the standard analog, providing 4x in/4x out channels.

- Conforms to AES3 standard
- Sample rates: 32kHz to 96kHz.
- Rate conversion: Input and output, independently selectable
- Output Clock: AES input or 48kHz internal.
- Input Level: Adjustable -27 to –9 dBFS nominal.
- Output Level: Adjustable –27 to –9 dBFS nominal.

## Livewire

LIVEWIRE CONNECTION IS STANDARD FOR ALL HYBRIDS IN/OUT AND POH AUDIO IN.

## Audio Performance

- Frequency Response: +/-0.5dB, 20Hz to 20kHz (swept sine procedure, measured from analog input to output with unit in loopback mode)
- THD+N: Analog to analog, studio loop mode, 1Khz 18dBu test level: <0.006%
- Dynamic Range: Analog in to Analog out, studio loop mode, 10Hz-20Khz. A-weighted: > 92 dB
- SNR: Analog output, referred to -12dBm phone line signal (+4dBu studioout), 10Hz-20Khz a-weighted: > 72 dB
- Trans-Hybrid Loss: Analog or ISDN phone line, ducking, gate, AGC, EQ all off, relative to +4dBu input level: > 55 dB

# SPECIFICATIONS

## Switching Matrix and Conferencing:

- Audio Routing and Switch: All Digital
- Telephone Lines: 6
- Hybrids: 4. All available when Livewire I/O is used, 2 available when analog or AES/EBU I/O is used.
- Studio Inputs: 4
- Studio Outputs: 4
- Program-on-Hold: 2

## ISDN Telephone Line Connectivity

- Protocol Compatibility: National ISDN-1 and 2, DMS-100 Custom Functional, AT&T 5ESS Custom Point-to-Point, Euro-ISDN conforming to the NET 3/ETS300 protocol
- Interface (one of the following):
    - USA & Canada Integrated NT1 for direct connection to ISDN line via the 2-wire U-interface (6-position/2-pin RJ-11 connector). 2B1Q line coding.
    - Worldwide 4-wire S interface (8-position/8-pin RJ-45 connector)
- Telephone Coding Modes
    - µLaw (ISDN Proto set to Natl I-1, AT&T Custom, Q.931mu or DMS Custom)
    - A-Law (ISDN Proto set to ETS-300)

## Analog Telephone Line Connectivity

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

# SPECIFICATIONS

## Desktop Director Ports

- Four ports, permitting connection of 4 Telos Desktop Directors. Eight directors using external RJ-45 splitters and power.

## Control Ports:

- Ethernet 100Base-T
    - Web server for configuration and software update
    - Telnet for command line control and diagnostics
    - Call Screening Interface server allows up to 8 instances of call screening software to connect simultaneously
- General purpose Input/Output: 15 pin D-Sub connector with 5 status outputs and 4 control inputs.

## Two-Studio Modes

Nx systems can be used with two studios independently. Lines may be independent or shared. Each studio controls two hybrids. There are two Program-on-Hold inputs, one for each studio. Two Broadcast Bionics Xscreen applications can be used independently.

- With analog audio in/out, there are only two total inputs and outputs. So the Talkshow System must be used in a mode that combines the two hybrids assigned to each studio into one input/ output.
- The AES option provides 4 input and 4 output channels total, so all hybrids are accessible with each studio using one AES connection's two channels. Program-on-Hold remains an analog input.
- Livewire provides the full 4 hybrid input channels, 4 hybrid output channels, and 2 Program-on-Hold input.

## Power Supply

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

# HX6 SIX-LINE TALKSHOW SYSTEM

## Give Your Phones an Instant Upgrade

## OVERVIEW

Hx6 is Telos' most advanced six-line digital Talkshow system. It features two high-performance digital hybrids and includes Telos' famous Digital Dynamic EQ, noise gate, caller ducking and acoustic echo cancellation. Works with analog or digital phone lines (6 POTS or 3 ISDN BRI). Single-cable Ethernet hookup via Axia Livewire I/O, or choice of analog or AES/EBU I/O with one input and one output per hybrid, and one Program On-Hold input. Includes complimentary XScreen Lite call screening software from Broadcast Bionics.

# FEATURES

- Six line capacity; works with POTS (analog) or ISDN (digital) phone lines.
- Our most advanced digital hybrids, with DSP algorithms optimized for superior performance with today's wide variety of far-end call types (VoIP, cell, POTS, app-based).
- Telos DDEQ (Digital Dynamic EQ) and adjustable smart-level AGC ensure spectrally consistent audio from call to call — even on notoriously tough cellular calls.
- Excellent trans-hybrid loss of >55dB.
- Smooth, proven, symmetrical wide-range AGC by the audio processing experts at Omnia Audio.
- Studio adaptation and a subtle, inaudible pitch shifter to prevent feedback in open-speaker studio environments.
- A sophisticated caller override that improves performance and allows precision adjustment of the degree to which talent audio "ducks" the caller audio.
- Striking Telos VSet6 six-line phone controllers with large, colorful VGA LCD displays that provide intuitive operation and setup. Telos-exclusive Status Symbols provide producers and talent with animated, high-contrast icons that communicate line and caller status at a glance.
- Three different Hx6 versions matched to your choice of analog POTS phone lines, ISDN-S (Europe), or ISDN-U (North American) digital phone lines.
- Caller ID for both analog and digital telephone connections, which is displayed on the VSet6 phoneset and the included XScreen Lite call screening application.
- Livewire IP-Audio allows fast, one-cable integration with Axia networks, and provides Axia board operators with seamless, on-console control of multiple lines and hybrids. Standard Ethernet backbone provides a common transport path for both studio audio and telecom needs, resulting in cost savings and a simplified studio infrastructure.
- Choice of standard Analog I/O or optional, extra-cost AES/EBU I/O.
- Easy setup and configuration via Ethernet using any PC and your favorite Web browser.
- XScreen Lite call screening software from Broadcast Bionics, provided at no cost.

# IN DEPTH

## Advanced caller management and superior sound

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

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

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

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

# IN DEPTH

Around back, you'll find audio I/O, GPIO, and Telco connections. Hx6 comes in versions that work with analog or digital phone lines, which connect directly to 6 POTS or 3 ISDN BRI lines. Separate analog or optional AES digital I/O is provided for each hybrid, as well as a Program-On-Hold input, GPIO connections for speaker muting, ring tallies, et cetera.

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

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

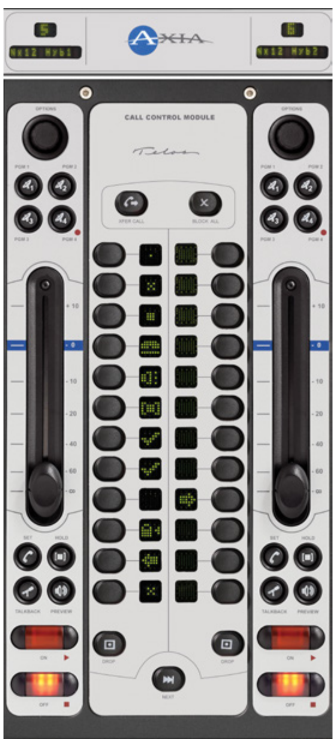
## Intuitive, easy-to-use controllers

This is the Telos VSet6 six-line phone controller, an IP-based phoneset with a large, high-contrast color LCD panel that provides line status and caller information. There's almost no learning curve; VSet

# IN DEPTH

phones work like traditional Telos controllers, with calls selected, held, and dropped in the way to which operators have grown accustomed. Exclusive animated Telos Status Symbol™ icons show line and caller status at a glance; easy VSet controls let talent manage incoming lines, lock calls on-air, start an external recording device, and take a queue of calls to air sequentially, for precise management of multi-call interviews or conferences. The LCD display delivers detailed line status, caller information, caller ID, time ringing-in or on-hold, and even comments entered in the included Xscreen Lite screening software. A built-in address book and call history log round out VSet6's features. And, just like the Hx6 itself, each VSet6 has its own web server for easy remote configuration and software upgrades.

## On-Console Control



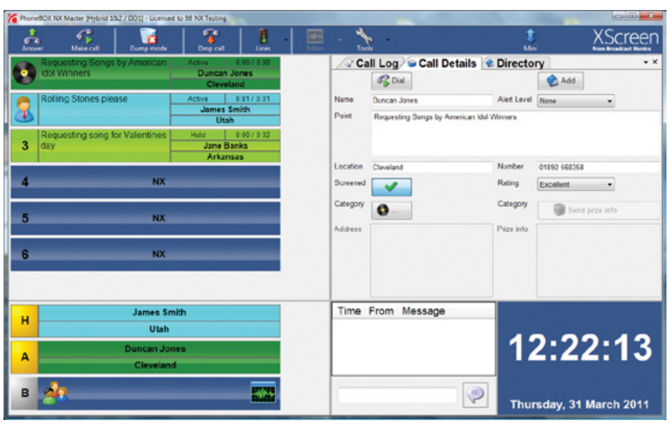

Hx6 works with any brand of broadcast console. But wouldn't it be great if talent could take control of phones without ever having to divert their attention from the board? Whether your shows consist of live calls or pre-recorded interviews, phone segments are usually fast-paced with little room for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

# IN DEPTH

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

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

## Call Screening Software Included




Hx6 comes complete with XScreen Lite call screening software, from Broadcast Bionics. XScreen Lite's interface gives screeners and hosts tons of information and control using sophisticated visual talkback, including a drag and drop database of all calls for your show as well as a phonebook and visual warnings for persistent or nuisance callers. A fully-functional copy of XScreen Lite is provided to all Hx6 customers, but an upgrade to the full XScreen client software adds even more features, including extended call history, an enhanced phonebook, prize management, powerful GPIO functionality and more. XScreen, deployed as part of a Livewire network, also enables call recording, editing and console integration directly over the network.

# SPECIFICATIONS

## General

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

## Analog Inputs:

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

## Analog Outputs:

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

## Switching Matrix and Conferencing:

- Audio Routing and Switch: All Digital

# SPECIFICATIONS

- Telephone Lines: 6
- Hybrids: 2
- Studio Inputs: 2
- Studio Outputs: 2
- Program-on-Hold: 1

## Control Ports:

- Ethernet 100Base-T
  - Web server for configuration and software update
  - Telnet for command line control and diagnostics
  - Call Screening Interface server allows up to 8 instances of call screening software to connect simultaneously
- General purpose Input/Output: 2x 15-pin D-sub with status outputs and control inputs.
- Control Interface: Up to 12 attached controllers (any mix of VSet6 phones, Console Controllers or screening software) via Ethernet connection.

## POWER SUPPLY

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

## ISDN TELEPHONE CONNECTIVITY:

- Protocol Compatibility: National ISDN 1 and 2, DMS-100 Custom Function, AT&T 5ESS Custom Point-to-point, Euro-ISDN conforming to the Net 3/ETS300 Protocol
- Interface: USA & Canada — Integrated NT1 for direct connection to ISDN line via the two-wire U-interface (6-position/2-pin RJ-11 connector) 2B1Q Line Encoding; Worldwide — 4-wire S-interface (8-position / 8-pin RJ-45 Connector)

# SPECIFICATIONS

- Telephone Coding Modes: μLaw (ISDN Proto set to Natl I-1, AT&T Custom, Q.931mu or DMS Custom), A-Law (ISDN Proto set to ETS-300)

## ANALOG TELEPHONE CONNECTIVITY:

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

# iQ6 SIX-LINE TELCO GATEWAY

## Talkshow System for Axia IP-Audio Networks


## OVERVIEW

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

iQ6 is built around the Telos Hx6, our most advanced six-line digital Talkshow system. It features two high-performance digital hybrids and includes Telos' famous Digital Dynamic EQ, noise gate, caller ducking and acoustic echo cancellation. iQ6 works with analog or digital phone lines (6 POTS or 3 ISDN BRI). Includes complimentary XScreen Lite call screening software from Broadcast Bionics.

# FEATURES

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

# IN DEPTH

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

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

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

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

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

# IN DEPTH

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

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

The back panel likely looks much different from any other phone system you've seen. There are no discrete audio I/O, GPIO, PoH or output connections -- the RJ45 connection to your Axia Livewire network handles all of that.. Like its brother, the Telos Hx6, iQ6 comes in versions that work with analog or digital phone lines, which connect directly to 6 POTS or 3 ISDN BRI lines (ISDN-U and ISDN-S interfaces, for worldwide use).

# IN DEPTH

## On-Console Control

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

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

# IN DEPTH

since the console now communicates directly with the phone hybrid, mundane tasks such as mix-minus generation, starting recording devices, and playback of recorded off-air conversations can all be automated. All of which means faster, more precise phone segments — since operators' eyes never need to leave the console.

## VSet6 Phone Controllers

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

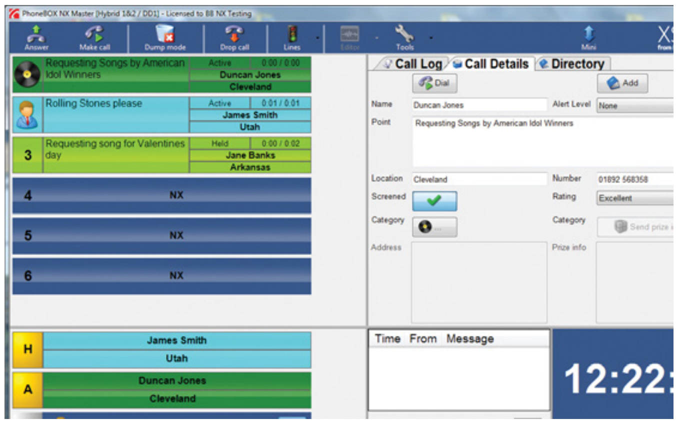
VSet6 is an IP-based phoneset that also connects to Axia networks with a single Ethernet connections. There's almost no learning curve; VSet phones work like traditional Telos controllers, with calls selected, held, and dropped in the way to which operators have grown accustomed. Easy VSet controls let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

## Call Screening Software Included

iQ6 comes complete with XScreen Lite call screening software, from Broadcast Bionics. XScreen Lite's interface gives screeners and hosts tons of information and control using sophisticated visual talkback, including a drag and drop database of all calls for your show as well as a phonebook and visual

# IN DEPTH

warnings for persistent or nuisance callers. A fully-functional copy of XScreen Lite is provided to all Hx6 customers, but an upgrade to the full XScreen client software adds even more features, including extended call history, an enhanced phonebook, prize management, powerful GPIO functionality and more. XScreen, deployed as part of a Livewire network, also enables call recording, editing and console integration directly over the network.




# SPECIFICATIONS

## General

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

## Input / Output Channels

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

# SPECIFICATIONS

## Switching Matrix and Conferencing:

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

## Control Ports:

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

## Power Supply

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

# SPECIFICATIONS

## ISDN TELEPHONE CONNECTIVITY:

- Protocol Compatibility: National ISDN 1 and 2, DMS-100 Custom Function, AT&T 5ESS Custom Point-to-point, Euro-ISDN conforming to the Net 3/ETS300 Protocol
- Interface: USA & Canada — Integrated NT1 for direct connection to ISDN line via the two-wire U-interface (6-position/2-pin RJ-11 connector) 2B1Q Line Encoding; Worldwide — 4-wire S-interface (8-position / 8-pin RJ-45 Connector)
- Telephone Coding Modes: μLaw (ISDN Proto set to Natl I-1, AT&T Custom, Q.931mu or DMS Custom), A-Law (ISDN Proto set to ETS-300)

## ANALOG TELEPHONE CONNECTIVITY:

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

# HX1 & HX2 DIGITAL HYBRIDS

## POTS phones never sounded so good.


## OVERVIEW

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

# FEATURES

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

# IN DEPTH

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

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

# IN DEPTH

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

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

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

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

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

# SPECIFICATIONS

## General

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

## Analog Inputs:

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

## Analog Outputs:

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

# SPECIFICATIONS

## AES3 Digital Inputs/Outputs (optional)

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

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

## Audio Performance

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

## ANALOG TELEPHONE LINE CONNECTIVITY

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

## POWER SUPPLY

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

# TELOS PROSTREAM

## The do-it-all, one-box streaming solution.


## OVERVIEW

Telos ProStream combines audio processing with MP3 and AAC encoding in one convenient, single-rack unit. The AAC encoder supports AAC-LC, HE-AAC and HE-AAC v2 formats, and is fully managed and configured remotely with any standard Web browser. ProStream features a wideband AGC, 3-band compressor/limiter, EQ, low-pass filter and a precision look-ahead final limiter; processed audio can then be encoded directly to MP3 or AAC streams to feed a remote replication server at your ISP. Streams can be "tagged" with "now-playing" information received from automation systems. Analog and Livewire I/O are standard.

# FEATURES

- Audio pre-processing, stream encoding and delivery to remote replication server, all in a professional 1RU appliance.
- Pro-grade 24-bit A/D converter for studio-reference quality audio.
- Choice of MP3 or AAC-LC, HE-AAC, HE-AAC v2 stream coding, with output bit rates from 16 kbps to 320 kbps (dependent upon active codec).
- Omnia processing includes wideband AGC, 3-band compressor/limiter, EQ, low-pass filter and precision look-ahead final limiter.
- Metadata support for all popular playout platforms allows streams to be dynamically tagged with "now-playing" information from automation systems.
- Studio-grade analog and Livewire IP-Audio I/O, with separate LAN & WAN Ethernet ports.
- Directly supports ICEcast, SHOUTcast, SHOUTcast v2, Adobe Flash Media Server as well as Adobe RTMP, RTP streams (including RTP multicast), as well as LimeLight, Akamai and other popular streaming servers.
- Dual encoder support can be used to provide high and low bitrate streams, or MP3 and AAC at the same time.
- Can accept metadata over RS-232 (using USB to RS232 adapter).

# IN DEPTH

## Plug. Play. Netcast.

For years, the way to stream audio to Internet listeners included unbalanced mini-jacks, poor-quality sound cards, one or more PCs to maintain, and a collection of software that didn't always play nicely together. Broadcasters asked for a professional, PC-free Web streaming solution — and Telos delivers.

Telos ProStream takes the hassle out of netcasting. There's no PC needed; ProStream takes just 1RU of rack space. Slide it in and it's ready to go – no more running your mission-critical audio over crash-prone PC hardware and operating systems. Just send audio to ProStream, make a few setup selections and, within minutes, you'll be streaming your programming to your favorite stream server or streaming service for worldwide distribution.

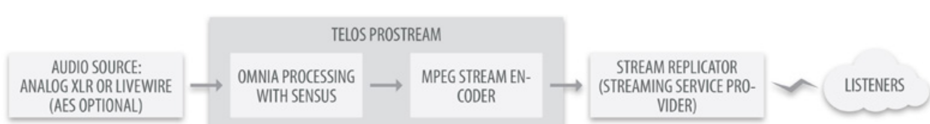
# IN DEPTH

Broadcasters know Telos is the codec expert. ProStream puts all our expertise into one integrated streaming appliance. First, incoming audio gets treated to pre-processing from Omnia Audio, using algorithms that work hand-in-glove with ProStream's codecs to shape and optimize audio prior to encoding. Then, genuine MPEG encoding algorithms from FhG, the inventors of MP3, ensure the most artifact-free sound quality at whatever bit rate you choose. Encode directly to an MP3 or MPEG-AAC stream, then send it to a Shoutcast, Wowza, Icecast, LimeLight, Akamai, Adobe Flash Media server, or other popular streaming server for distribution to your waiting listeners.

Setup is a breeze. Log in with a laptop and Web browser for easy setup or remote control, or tweak the front-panel controls -- there's a convenient built-in headphone amp with 1/4" jack and volume control for last minute in-the-rack fine tuning.

ProStream comes with studio-grade analog inputs and outputs, plus Livewire IP-Audio I/O. On the output side, ProStream delivers fully processed, unencoded audio as well as encoded audio, providing your studio with another source for processed sound. Full network connectivity is provided via two Ethernet jacks, one for the LAN (including Livewire) and the other for the WAN and streaming.

## The Professional Choice for Streaming Audio.




Optimizing sound quality is as essential on the web as it is on traditional formats. ProStream has a built-in processing section that works together with the streaming encoder, optimizing your audio for stunning sound — even after bit-reduction. This isn't just some cheap leveler – it's real processing by Omnia, complete with wideband AGC, a 3-band combined compressor/limiter, high-frequency EQ, an adjustable-bandwidth low-pass filter, and Omnia's famous anti-aliasing final Look-Ahead limiter. There are even a selection of presets, tailored to specific formats and bit rates, to help you get up and running quickly.

# IN DEPTH

Of course, the foundation for high fidelity audio distribution rests on professional encoding technology. The quality of the encoder directly affects the quality of the output. Telos has a long history of partnership with Germany's Fraunhofer Gesellschaft Laboratory (FhG), the world leader in audio compression research and the inventors of MP3; ProStream uses genuine MP3 and MPEG-AAC encoding algorithms to ensure the most artifact-free sound quality at any bit rate you choose, from 16 kbps all the way to 320 kbps. No other encoder has this pedigree, or achieves this level of quality and performance. Generic "mp3" encoders can't come close.

ProStream gives you a wide choice of genuine Fraunhofer encoding algorithms, which include MP3, the Standard for digital audio. It's the safest codec choice for compatibility with the widest variety of listening devices. Or choose AAC-LC, a high performance codec for excellent audio quality at lower bitrates. AAC-LC is in widespread use, most notably in Apple's iTunes. And then there's High Efficiency Advanced Audio Coding, or HE-AAC, a newer AAC codec which incorporates Spectral Band Replication (SBR) bandwidth expansion to improve audio at very low bitrates. HE-AAC v2 applies a Parametric Stereo feature to HE-AAC codec allowing for even further reduction in bandwidth.

When you're done processing and encoding, select your metadata source and feed your stream to any SHOUTcast or SHOUTcast v2-compatible media server, or a Wowza server for streaming to Flash clients. ProStream works with ICECast and Adobe Flash Media and Adobe RTMP servers too, as well as popular streaming services from LimeLight, Akamai, and other popular streaming service providers. You can feed directly to a streaming server on your LAN, to an Internet streaming relay service via the WAN port, or take processed audio from the rear-panel XLR outputs. No matter what your audio source or how you stream, ProStream delivers flawlessly optimized audio that sounds terrific.

# SPECIFICATIONS

## Audio Coding

- Codec Choices:
    - MP3: 16 to 320 kbps
    - AAC-LC: 24 to 320 kbps
    - HE-AAC: 24-96 kbps
    - HE-AAC v2 (aacPlus): 24-96 kbps

# SPECIFICATIONS

- AAC Transport Modes:
    - ADTS
    - ADTS-CRC
    - ADIF
    - RAW
- Metadata Formats:
    - Character Parser Sample
    - Line Parser Sample
    - Nexgen Audio Sense
    - Simian Template 1
    - XML Parser Sample
    - XML-Jazler
    - XML-Jazler2
    - XML-MediaTouch
    - XML-MediaTouch2
    - XML-Sample2
    - XML-Zetta
    - User-definable

## Input

- Analog: Balanced XLR, +4 dBu
- Input Impedance: 6K Ohm differential
- Analog to Digital Converter: 24bits
- Digital: Livewire AoIP, via LAN port

# SPECIFICATIONS

## Output

- Analog: Balanced XLR
- Output Clipping: + 22dBu
- Output Impedance: 50 Ohm differential
- Digital to Analog Converter: 24bits
- Digital: Livewire AoIP, via LAN or WAN port

## Audio Performance

- THD+N: < 0.03% @ +12dBu, 1 kHz Sine
- Frequency Response: +/- 1dB 25– 20 kHz
- Headroom: 18dB
- Dynamic Range: > 87dB Unweighted > 90 dB "A" Weighted
- Crosstalk: > 80 db

## Remote Control

- LAN via built-in Webserver

## Power

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

# OMNIA.11 FM AND FM/HD

## Maximum firepower for an extremely competitive environment

## OVERVIEW

Available in two models: Omnia.11 FMHD with separate processing paths for FM and HD/DRM, and Omnia.11 FM without HD/DRM. FM-only model is upgradeable to FM/HD at a later date. Switchable Single Sideband Suppressed Carrier (SSBSC) technology for potential reduction of multipath is standard feature. A front panel touch screen GUI, on a 10.5" diagonal screen, provides ease of use and enhanced metering and diagnostics. Remote access is via any web browser, as well as a local onboard WI-FI connection. Laptops to iPads will have access. Livewire, AES/EBU digital and analog I/O are standard. Fanless cooling. Rugged 4 RU chassis.

# FEATURES

## Chameleon Processing Technology

Enables Omnia.11 to handle rapidly changing, hyper compressed source material.

## Ultra-Multiband Limiter System

Self-adjusting attack/release functions guarantee crystal clear music and voice

## Bass Management

Manages harmonics for a natural and undistorted bottom end.

## Ultra LoIMD Distortion Controlled Clipper System

Dramatically reduces intermodulation distortion (IMD) for more loudness headroom

## SSBSC Technology

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

## Extra Wide Touchscreen

10.5" diagonal screen clearly shows all controls.

## Looks Cool and Stays Cool.

Military-grade industrial design stays cool due to robust heatsinks in rugged 4RU chassis.

## Once You Get One You Won't Let Go

Built-in, retractable handles make it easy to transport and install

# IN DEPTH

## Chameleon Processing Technology

A major part of this technology, the new Density Detector, enables Omnia.11 to properly handle hyper compressed content. The AGC system cannot be fooled due to heavy density, or by older source material which contains high peak-to-average levels. The density-detector keeps Omnia.11 operating on-target, at all times.

## Ultra-Multiband Limiter System

Traditional limiting technology has often resulted in various forms of audio corruption. Omnia.11's new LoIMD technology coupled with smart gain reduction algorithms, now have limiters which sound amazingly transparent.

All AGC and limiting algorithms employ an auto acceleration/deceleration mechanism, which tunes out perceptible intermodulation distortion. The attack/release functions adjust themselves based upon content density. This breakthrough method literally analyzes the audio content in both the amplitude and frequency domain, then adapts the timing networks - on the fly - to transparently control the signal, without the control being heard. The result is revealed in added detail, clarity, and quality, yet maintaining the desired competitive loudness level.

Special attention was paid to the behavior of live voice quality. The improved performance of the AGC and limiter functions generate live voice clarity and impact far beyond that which was previously possible.

## Bass Management

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

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

# IN DEPTH

## Ultra LoIMD Distortion Controlled Clipper System

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

For those who feel the need to use it, there's also a composite clipper embedded in the stereo generator. However, to date, all of our testing has been done without any composite clipping. Pilot protection is on the order of magnitude close to 90 dB, which is considerably more protection than necessary for even the best FM receiver. Integrated laboratory-grade stereo generator with dual MPX outputs, 19 kHz reference output for external RDS/RBDS systems and pilot protection that provides >80 dB pilot protection - with or without composite clipping. MPX spectral low-pass filter to protect RDS/RBDS and SCA signals if composite clipping is employed. Multiple ways to adjust the system to achieve the exact sound you're looking for. An installation wizard will guide anyone through a simple step-by-step setup to on-air operation. Using the answers to a series of simple questions, Omnia.11 adapts itself, based upon the answers, to craft a preset which delivers the desired end result quickly for an effortless out of the box experience.

## Unprecedented Access

A front panel touch screen GUI, on a 10.5" diagonal screen, provides ease of use and enhanced metering and diagnostics. Remote access is via any web browser, as well as a local onboard WI-FI connection. Laptops to iPads will have access.

Livewire, AES/EBU digital and analog I/O is standard. Headphone soft "patch points" are available for listening through the processing chain.

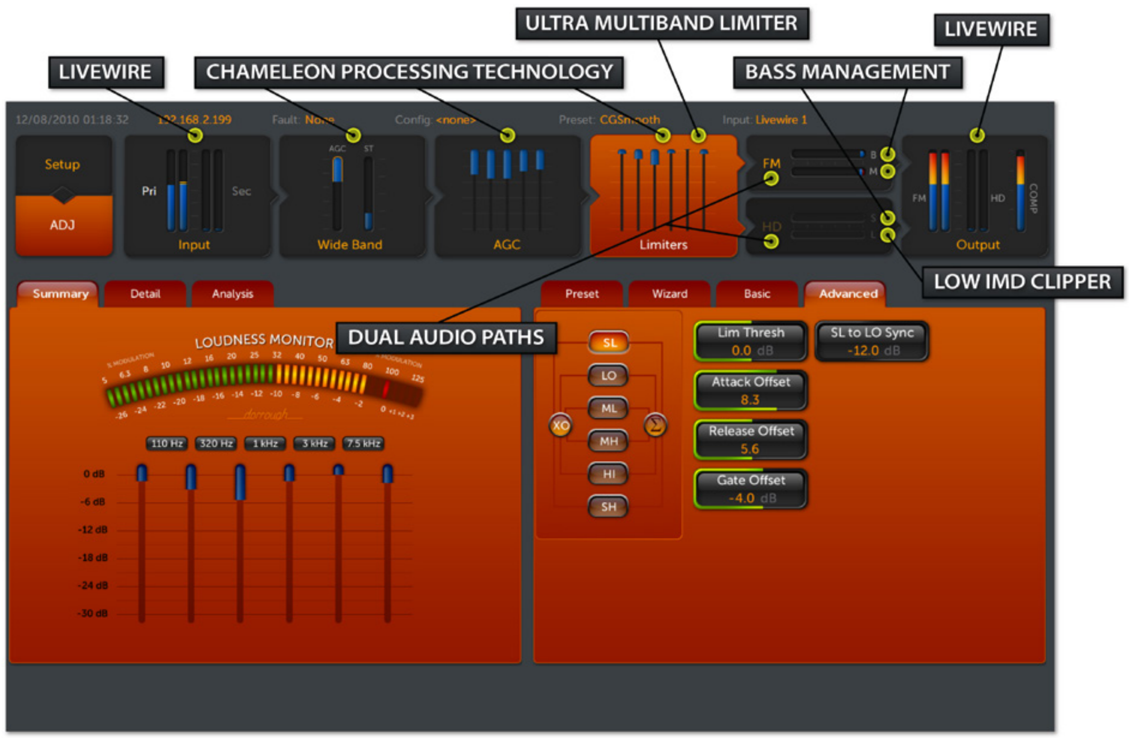
Diversity-Delay.

# IN DEPTH

## Additional Features

“A front panel touch screen GUI, on a 10.5” diagonal screen, provides ease of use and enhanced metering and diagnostics. Remote access is via any web browser, as well as a local onboard WI-FI connection. Laptops to iPads will have access.

Livewire, AES/EBU digital and analog I/O is standard. Headphone soft “patch points” are available for listening through the processing chain.



# SPECIFICATIONS

## General

Non-linear Crosstalk: > -80 dB, main to sub or sub to main channel (referenced to 100% modulation).

38 kHz Suppression: > 70 dB (referenced to 100% modulation).

76 kHz Suppression: > 80 dB (referenced to 100% modulation).

Pilot Protection: > -65 dB relative to 9% pilot injection, ± 1 kHz.

57 kHz (RDS/RBDS) Protection: better than -50 dB.

Connectors: Two EMI suppressed female BNC, floating over chassis ground

Maximum Load Capacitance: 5nF (at 10 ohms source impedance).

Maximum cable length: 100 feet/30 meters RG-58A/U.

## Analog Audio Input

Left/Right Stereo. Electronically balanced.

Input impedance 10k ohms resistive.

Maximum Input Level: +22 dBu.

Nominal Input Level: +4dBu, which nets a -18dBFS input meterreading on a steady-state signal when the Input Gain controlis set to 0.0dB. Program material with a nominal average level(VU reading) of +4dBu will typically produce peak readings on the input meter in the range of -12 dBFS to -6dBFS. This is thecorrect operating level.

# SPECIFICATIONS

## A/D Conversion

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

Connectors: Two, EMI-suppressed XLR-female. Pin 1 chassis ground, Pin 2 "Hot".

## Analog Audio Output

Left/Right Stereo. Electronically balanced.

Output Impedance 20 ohms.

Minimum load Impedance: 600 ohms.

Output Level adjustable from -2 dBu to +22dBu peak
in 0.1dB steps.

## D/A Conversion

Crystal Semiconductor CS4391, 24 bit, 128x oversampled.

Connectors: Two, EMI-suppressed XLR-male. Pin 1 chassis ground, Pin 2 "Hot".

## Frequency Response

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

## System Distortion

Less than 0.01% THD, 20 Hz – 7.5 kHz. Second harmonic distortionabove 7.5 kHz is not audible in the FM system.

* Signal-Noise Ratio: > -80 dB de-emphasized, 20 Hz – -15 kHzbandwidth, referenced to 100% modulation.

# SPECIFICATIONS

* The measured noise floor will depend upon the settings of the Input and Output Gain controls and is primarily governed by dynamic range of the Crystal Semiconductor CS5361 A/D Converter which is specified as >110 dB. The dynamic range of the internal digital signal processing chain is >144 dB.

## Stereo Separation

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

## Crosstalk

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

## System Latency

36ms. "FM" channel, as measured from the analog inputs through the composite MPX output.

## Composite Outputs

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

## D/A Conversion

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

## Configuration

Two electrically independent outputs. Software based level adjustment.

## Load Impedance

50 ohms or greater load is suggested.

## Pilot Level

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

# SPECIFICATIONS

## Pilot Stability

19 kHz, ± 0.5 Hz.

## Signal-to-Noise Ratio

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

## Distortion

< 0.02% THD 20 Hz – 15 kHz bandwidth, 75 _S
de-emphasized, referenced to 100% modulation.

Stereo Separation: > 65 dB, 30 Hz – 15 kHz.

Linear Crosstalk: > -80 dB, main to sub or sub to main channel (referenced to 100% modulation).

## Connector

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

## External Sync Range:

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

## Remote Control:

Via Ethernet using built-in Java (TM) based remote control program integrated into web page interface. All software is served from the built-in web server to any standard web browser; there is nothing to install on the user's computer.
Connectors: Ethernet - Industry standard EMI-suppressed RJ-45 connector.

# SPECIFICATIONS

## GPI Interface:

Connector: EMI suppressed DB-15 female connector.

## Power Requirements:

Voltage: 100-250 VAC, 47-63 Hz.

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

Power Supply: Internal. Overvoltage and short circuit protected.

## Digital Audio Input:

Configuration: Stereo per AES/EBU standard, CS8420 Digital
Audio Transceiver with 24 bit resolution, software selection of stereo, mono from left, mono from right or mono from sum.

Automatically accepts and locks to input sample rates between 30 and 108 kHz.

Connector: XLR-female, EMI-suppressed. Pin 1 chassis ground, pins 2 and 3 transformer isolated, balanced, and floating – AES3 standard 110 ohm impedance.

## Digital Audio Output #1:

Stereo per AES3 standard. Output can be configured in software for flat or pre-emphasized response at 50 or 75 microseconds.

Digital Sample Rates: Output sample rates software selectable for 48kHz, Sync to Input or Sync to External.

Connector: XLR-male, EMI-suppressed. Pin 1 chassis ground, pins 2 and 3 transformer isolated, balanced, and floating. Standard AES3 specified 110 ohm source impedance.

Digital Output Level: -22.0 to 0.0 dBFS software adjustable.

# SPECIFICATIONS

## Digital Audio Output #2:

Stereo per AES3 standard. Output can be configured in software for flat pre-emphasized response at 50 or 75 microseconds.

Digital Sample Rates: Output sample rates software selectable for 48kHz, 44.1kHz or Sync to External.

Connector: XLR-male, EMI-suppressed. Pin 1 chassis ground, pins 2 and 3 transformer isolated, balanced, and floating. Standard AES3 specified 110 ohm source impedance.

Digital Output Level: -22.0 to 0.0 dBFS software adjustable.

## External Sync Input:

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

# OMNIA.9

## FM or AM processing. Optional HD-1, HD-2, HD-3 processing and streaming/encoding, RDS. Also available in Dual Path model.

## OVERVIEW

FM or AM processing on demand is standard. Simultaneous FM+AM if programming is 100% simulcast and at same transmitter site. Studio output with very low latency for talent monitoring is also standard. Optional HD-1, HD-2, HD-3 processing and streaming/encoding, RDS. Dual path model available for two separate program feeds broadcasting on FM+FM, FM+AM or FM+FM+AM simulcast with HD output, IP-Streaming with separate processing and encoding and low latency studio processing, all built-in for both feeds.

# FEATURES

## Exclusive "Undo" Technology

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

## Psychoacoustically Controlled Distortion-Masking Clipper

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

## Omnia Toolbox

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

## Speaker Calibration

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

## Dry Voice Detector

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

## Remote Client

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

## Auto Pilot

A selectable feature that automatically turns off the 19kHz stereo pilot when Omnia.9 detects that the source material is mono, and turns it back on when the input is determined to be stereo.

# FEATURES

## Additional Features

Multiband downward expansion (source noise reduction)

3-stage wideband AGC with adjustable sidechain equalization

Program-dependent multiband compression

Multiband look-ahead limiting

Selectable phase linear high pass filter, 15, 30 or 45 Hz

(For Digital) Two-band final look-ahead limiting

Selectable phase linear (high latency) or low latency
(talent air monitoring capable) modes

7 inch front panel touch screen

Full remote control.

On-screen keyboard with several layouts (QWERTY, QWERTZ, AZERTY, Dvorak and ABC sequential) for easy setup and preset name typing

Selectable SSB (Single Sideband) stereo encoder

HTTP push support for automation, such as dynamic RDS and streaming song titles, preset recall

Studio Output with very low latency for talent monitoring

Dual independent power supplies

Composite pass-through (relay bypass) for your backup processor

# IN DEPTH

## "Undo"

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

## Psychoacoustically Controlled Distortion-Masking Clipper

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

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

# IN DEPTH

## Omnia Toolbox

When Leif Claesson was creating Omnia.9, he knew that having diagnostic and measurement tools would be necessary. The original plan was to keep them in place only for development, but he quickly realized that engineers would find great value in them as well, and decided to leave them in place. Audio processing is largely a "hearing" process, but there is much to be learned by seeing what your adjustments are doing to your sound as well. Some stations still have an oscilloscope on the test bench or a spectrum analyzer at the transmitter, but it's not always convenient (or possible) to hook up a processor to them while it's on the air.
Even if you did so, you're pretty much limited to monitoring only the composite output of your own station's processing. Omnia.9's built-in solution means there's no extra test equipment to buy ('scopes and analyzers aren't cheap) and no cables to hook up. It also means you to visually monitor the signal at the input, the output, and dozens of in-between points throughout the processing path so you can tell what's happening to the audio every step of the way. As an added bonus, Omnia.9's composite inputs can be fed from a calibrated tuner or frequency-agile mod monitor so that you can monitor the other signals in your market too!

The RTA and speaker calibration tools are included for similar reasons. While it is certainly good practice to listen to your station on a variety of radios and speakers as you adjust your processing, it is also good practice to have at least one set of calibrated speakers available. Otherwise, the changes you make to your processing will be influenced by listening to speakers that either under- or over-represent certain frequencies. By adding an inexpensive calibrated microphone and using the included pink noise generator and RTA, you can quickly and easily calibrate a set of speakers to use as a reference as you adjust your sound.

## Speaker Calibration

If you make decisions about your processing on uncalibrated monitors, you are making choices that are influenced by the deficiencies and exaggerations present in every speaker, not to mention the coloration imposed by the room in which you are listening. Simply put, you're dealing with subjective, not objective, information. By using the pink noise generator and RTA built into Omnia.9 and adding an inexpensive calibrated microphone, it is possible to calibrate any speaker system to deliver as flat a response as the speakers themselves will allow (small speakers still won't reproduce low frequencies as well as larger ones – the laws of physics still apply after all). With speaker and room influences removed from the equation, you are in a position to adjust your audio based only upon "the facts." When explaining this process to someone in person, this is the point in the conversation where they inevitably say, "But listeners aren't hearing my station on calibrated speakers! They're listening in their cars, at their computer, and through cheap ear buds, so I should too!" It's true - that's exactly how your listeners are hearing your station in the real world, and why it is always important to listen on a wide variety of radios

# IN DEPTH

in many different environments. But adjusting your processing this way is a shortcut to a lot of tail-chasing frustration and lousy audio.  Let's say you listen first in an inexpensive compact car with a typical factory stereo. You notice there isn't much bass, so you adjust your processing to deliver more low end. It sounds good. Then you move into a high-end luxury car with 10 speakers and a subwoofer, and the bass is muddy, boomy, and overwhelming. Why? Because you upped the bass in the processor to make up for deficiencies you thought were in your processing, but in fact were in your speakers! Having at least one pair of high quality, calibrated speakers to go back to as your reference will dramatically improve your on-air sound, save you valuable time, and help preserve your sanity at the same time. (Don't worry – there are still plenty of people at your station to chip away at your mental well-being – we just don't want to be among them!).

## Dry Voice Detector

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

## Remote Client

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

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

# IN DEPTH

## Auto Pilot

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

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

# SPECIFICATIONS

## Frequency Response

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

## Signal to Noise Ratio

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

## System Distortion

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

## Stereo Separation

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

## Digital Output Level

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

## Stereo Baseband Output

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

# SPECIFICATIONS

## A/D Conversion

Crystal Semiconductor CS5361, 24 bit 128x over-sampled delta sigma converter with linear-phase anti-aliasing filter.

Pre-ADC anti-alias filter, with high-pass filter at <10 Hz

## D/A Conversion

Crystal Semiconductor CS4391, 24-bit, 128x oversampled

## External Sync Input

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

## Analog I/O

Two balanced, EMI filtered XLR connectors

## Stereo Generator Connections

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

(FM style only) AES/EBU In & External Sync

## Digital I/O

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

## Ethernet

Shared RJ45 supporting 100 and 1000 BASE-T Ethernet connections

## Power Requirements

100-264 VAC, 47-63Hz autosensing

# SPECIFICATIONS

## Power Connector

IEC male, detachable 3-wire power cords supplied

## Power Supply

Internal dual redundant, hot-swappable

## Environmental

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

## Regulatory

North America: Tested to comply with the limits for a class A digital device pursuant to Part 15 of the FCC rules (CFR). Designed for U.S. and Canadian listing with UL.

Europe: Tested for CE and RoHS compliance.

# OMNIA.7

## A very useful slice of features and tools from the wildly successful Omnia.9.

## Delivers FM plus HD processing in a compact 2RU package.

## OVERVIEW

FM+HD processing is standard. Studio output with very low latency for talent monitoring is also standard. Optional basic RDS encoding.

# FEATURES

## Exclusive "Undo" Technology

The Omnia.7 includes a basic version  of "Undo" which is a two-step process that restores peaks and dynamic range to - and removes distortion from - source material that has been damaged by over-compression and clipping during the mastering process.

## Psychoacoustically Controlled Distortion-Masking Clipper

Omnia.7's final FM clipper takes into account how the human ear hears and perceives distortion and uses that information to effectively mask it, leaving only clean, distortion-free audio on the air.

## Omnia Toolbox

Every Omnia.7 comes with the standard Omnia "toolbox" which includes a digital oscilloscope, an FFT spectrum analyzer, and real time analyzer (RTA) to help you adjust your processing and "see what you're hearing".

## Speaker Calibration

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

## Dry Voice Detector

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

## Remote Client

Omnia.7's client software allows full remote control of the processor from any Windows-based PC or tablet, including touch screen devices, on the local network. The remote interface looks and functions just like the front panel screen.

# FEATURES

## Additional Features

Multiband downward expansion (source noise reduction)

3-stage wideband AGC with adjustable sidechain equalization
Program-dependent 5-band multiband compression

5-band look-ahead limiting

Selectable phase linear high pass filter, 15, 30 or 45 Hz

(For Digital) Two-band final look-ahead limiting

Selectable phase linear (high latency) or low latency
(talent air monitoring capable) modes

4.3 inch/10.9 centimeter front panel touch screen

Full remote control

On-screen keyboard with several layouts (QWERTY, QWERTZ, AZERTY, Dvorak and ABC sequential) for easy setup and preset name typing

HTTP push support for automation, such as dynamic RDS and streaming song titles, preset recall

Studio Output with very low latency for talent monitoring

Dual independent power supplies

Composite pass-through (relay bypass) for your backup processor

# IN DEPTH

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The Omnia.7 carries on the tradition!

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# SPECIFICATIONS

## Frequency Response

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

## Signal to Noise Ratio

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

## System Distortion

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

## Stereo Separation

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

# SPECIFICATIONS

## Digital Output Level

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

## Stereo Baseband Output

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

## A/D Conversion

Crystal Semiconductor CS5361, 24 bit 128x over-sampled delta sigma converter with linear-phase anti-aliasing filter.

Pre-ADC anti-alias filter, with high-pass filter at <10 Hz

## D/A Conversion

Crystal Semiconductor CS4391, 24-bit, 128x oversampled

## Analog I/O

Two balanced, EMI filtered XLR connectors

## Stereo Generator Connections

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

## Digital I/O

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

## Ethernet

Shared RJ45 supporting 100 and 1000 BASE-T Ethernet connections

## Power Requirements

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

# SPECIFICATIONS

## Power Connector

IEC male, detachable 3-wire power cords supplied

## Power Supply

Internal dual redundant

## Environmental

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

## Regulatory

North America: Tested to comply with the limits for a class A digital device pursuant to Part 15 of the FCC rules (CFR). Designed for U.S. and Canadian listing with UL.

Europe: Tested for CE and RoHS compliance.

# OMNIA ONE

## FM, AM, HD Radio, DAB, DRM, Stereo Generator, multicasting, podcasting, netcasting or satcasting unit as needed.

## OVERVIEW

10,000+ Omnia ONEs. That's how many are on the air around the world. Engineers tell us of reliability and ease of operation. Program Directors love the power and punch of genuine Omnia processing. General Managers love the price!

# FEATURES

## FM

Smart wideband AGC followed by advanced Four-Band AGC and selectable Four or Five-Band Peak Limiter sections. Omnia'sadvanced, fully distortion-controlled, pre-emphasized final limiter / clipper. A newly designed digital stereo generator with SCA convenience input, two independently adjustable composite MPX outputs and 19kHz pilot output for synchronization to external RDS generators.

## AM

Omnia's advanced NRSC compliant, distortion-managed final limiter / clipper, including selectable Low Pass Filter frequencies that support AM HD transmission installations...the same as used in the Omnia ONE's bigger siblings.

## Multicast

Features SENSUS ™, an audio conditioning technology to minimize codec artifacts as well as restore the fullness and depth that bit-reduction steals. See expanded explanation about SENSUS technology later in this brochure. Ultra low-distortion final limiting optimized for the HD codec.

## Studio Pro

Full-bandwidth processor for applications that require minimal delay and do not require absolute peak limiting. The first studio processor to include a four-band compressor / limiter allowing you precise and accurately defined control while pre-processing music, commercials, remote feeds, or sweetening audio. Applications include recording studios, mastering labs, TV stations, radio headphone feeds ... just abut any application where signal processing is needed.

## Additional Features

Price includes software downloads for FM, AM, internet/satellite, studio processing`, or Stereo Generator on demand if necessary

Balanced XLR Analog inputs and outputs, plus Digital AES/EBU input, output and external Sync

Browser-based remote control and configuration

# FEATURES

Automatic backup input switching

Livewire / Ethernet RJ45 jack

Universal Power Input

Robust headphone amp with rugged jack and front panel volume control

Single Jog-Wheel and Selection Button user interface with LED level metering and LCD screen

# IN DEPTH

## Time alignment.

The Omnia ONE is completely, 100% time-aligned. This means that all audio signals, no matter what frequency, have the exact same propagation time from Input to Output of the Omnia audio processor. This is a claim that other manufacturers cannot make, as they don't deem time-alignment to be important. Omnia processors sound more precise and less "smeary" due to this attention to time-alignment.

## Pre-emphasis placement.

All other FM audio processors employ FM pre-emphasis prior to their multi-band limiting. This placement of the pre-emphasis function results in a more convenient design for the manufacturer. However, it becomes this approach in the processor which ends up sounding more dense and "packed up" in the higher audio frequencies. Most engineers and listeners have come to accept this sound as "the FM sound". This audio aberration is not a function of FM transmission, but a bad result of the design of traditional, multiband FM audio processors. Omnia takes a different approach. In the Omnia ONE, the necessary FM pre-emphasis is applied after the multi-band limiting. This technique requires difficult attention to both the limiting and clipping algorithms. However, the audible result is a cleaner, much more detailed high-end in the transmitted audio. This advantage of Omnia processing architecture is most easily noticed on musical instruments such as cymbals, castanets, trumpets, and other sounds with a lot of high-frequency energy. You will notice that these musical instruments sound "fake" through other audio processors, but they sound very real and natural through an Omnia audio processor.

# IN DEPTH

## Separate multiband AGC and multiband limiter stages.

Most other audio processors take a shortcut in the execution of multiband AGC and limiting functions; they put both functions within the same audio processing "block". Omnia takes a more comprehensive approach. We designed the multiband AGC and multiband limiting blocks completely separate from each other. This allows us to give slightly different treatment as needed for the absolute cleanest and competitively loudest audio. For example, some bands of limiting are best served by feed-back servo control, while the higher bands are best served by feed-forward servo control. Our design, which breaks these multiband functions into individual processing blocks, allows for the absolute best treatment of each audio band.

## Omnia ONE's Powerful and Comprehensive Clipper

In any FM audio processor, the final audio clipper presents the largest hindrance or benefit to the loudness and clarity of the on-air sound for a given modulation level. The FM clipper section in the Omnia ONE, originally available only in big brother Omnia-6, includes two distinct sections: a bass-management clipper and a main clipper. The bass clipper is quite sophisticated in its own right, but the main clipper offers two different clipping styles and the ability to "balance" between them, if desired. This flexibility gives the curious or competitive user the ability to finely tune this most important function.

## Sensus Overview

Until now, digital signal processing has been a more precise numeric implementation of well-known analog methods. Even relatively recently designed digital audio processors couldn't veer too far from the comparatively simplistic concepts that analog dynamics processing had utilized...until now!

Extremely high power DSP chips have become available and at relatively low cost, and they make it possible to build smarter and more complex processing algorithms that were too difficult or impossible (or too expensive) to do in the past.
Running on a platform of the latest high power DSP chips, the Omnia ONE and our new Sensus® technology takes digital dynamics processing into a completely new frontier. Instead of the two-dimensional static processing architecture of the past, Sensus® enables the audio processor to modify its own architecture in real time and in response to ever-changing program content. Simply stated,

# IN DEPTH

Sensus® has the ability to "sense" what must be done to a signal in order to best tailor it for output to a codec. As program content changes, it "rearranges the algorithms" to accomplish this goal. The uniqueness of the Sensus® technology makes it highly suitable not only for codec pre-conditioning (or provisioning), but also for a range of other highly specialized signal processing challenges. The following is a discussion of how Sensus® technology can be applied to a coded audio environment.

## Codec Provisioning

The codec is now a common denominator in the world of audio and broadcasting. Digital broadcasting (HDTV, HDRadioR, DAB, DRM), podcasting, webcasting, cellcasting, and downloadable music files all employ a form of codec-based data compression in order to minimize the bandwidth required to transmit audio data. The necessarily low bitrates utilized by these mediums presents a tough challenge for any audio processor used prior to a codec. Traditional dynamics processors are designed to fulfill the requirements of a medium where the functions are generally static. That is, they're well suited to the rather simplistic peak control and bandwidth limiting methods required for analog broadcasting, as well as for the signal normalization techniques used in recording and mastering. Audio codecs on the other hand are moving targets - each codec algorithm has its own set of artifacts.

Not only does the sonic quality vary depending on the algorithm and bitrate used, but more importantly they vary in their ability to mask their own coding action. This is why we call it a 'moving target' , and is why conventional audio processors fall short in a coded audio environment and can actually make coding artifacts worse due to their inability to adapt appropriately to the changing operation of the codec as the program content changes. Prior art in audio dynamics processing could only address some of the challenges of provisioning audio for coding. This hurdle existed because the codec adapts to the incoming program (so as to generate the least amount of output data representing the input audio) causing the sonic artifacts generated by the process to continually change. Unless the audio processor can predict these changing characteristics of the codec, it can't possibly create output audio that is perfectly tailored for the coding process.

Conventional processors utilize rather simplistic high frequency limiters and fixed low pass filtering that does not change with the program material. When these less intelligent processors feed a codec the audio might sound acceptable one moment and offensive the next. Because they cannot "know" what the codec will do next, the result is over-compensated, dull and lifeless audio... audio that still contains objectionable codec-generated artifacts!

# IN DEPTH

## Omnia ONE Multicast and HD® Radio

The advent of HD RadioR has introduced the capability to transmit multiple program streams, or "Multicast", within a single 96kbps digital broadcast data channel. To facilitate this, multicast relies on the use of codecs with comparatively low bitrates. A broadcaster can choose to transmit a number of multicast channels and select the bitrate for each one. However, the more multicast channels there are, the lower the bitrate each channel must have in order for them to all fit within the total available bandwidth.   To achieve maximum sound quality, the kind that attracts and holds listeners, those channels need specialized dynamics processing capable of creating great sound regardless of program content and bitrate.  They need Sensus®.

# SPECIFICATIONS

## Omnia ONE FM & SG

### Frequency Response

Complies with the standard 50 or 75 microsecond pre-emphasis curve within ± 0.50 dB, 30 Hz to 15 kHz. The analog left/right outputs and AES/EBU Digital outputs can be configured for flat or pre-emphasized output..

### System Distortion

Less than 0.01% THD, 20 Hz – 7.5 kHz. Second harmonic distortion above 7.5 kHz is not audible in the FM system.
*Signal-Noise Ratio: > -80 dB de-emphasized, 20 Hz –- 15 kHz bandwidth, referenced to 100% modulation).

*The measured noise floor will depend upon the settings of
the Input and Output Gain controls and is primarily governed
by dynamic range of the Crystal Semiconductor CS5361 A/D
Converter which is specified as >110 dB. The dynamic range of the internal digital signal processing chain is >144 dB.

### Stereo Separation

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

# SPECIFICATIONS

## Crosstalk

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

## Composite Outputs

Source Impedance: 5 ohms or 75 ohms, jumper-selectable. Singleended and floating over chassis ground.
Output Level: 0V to 10V in 0.05V steps, software adjustable.
D/A Conversion: Texas Instruments/Burr Brown PCM1798, 24-bit sigmadelta converter.
Configuration: Two electrically independent outputs. Software based level adjustment.
Load Impedance: 50 ohms or greater load is suggested.
Pilot Level: Adjustable from 4.0% to 12.0% in 0.1% steps and OFF.
Pilot Stability: 19 kHz, ± 0.5 Hz.
Signal-to-Noise Ratio: -85 dB typical, 75 μS de-emphasized, 15 kHz bandwidth, referenced to 100% modulation).
Distortion: < 0.02% THD 20 Hz – 15 kHz bandwidth, 75 μS
deemphasized, referenced to 100% modulation.
Stereo Separation: > 65 dB, 30 Hz – 15 kHz.
Linear Crosstalk: > -80 dB, main to sub or sub to main channel (referenced to 100% modulation).
Non-linear Crosstalk: > -80 dB, main to sub or sub to main channel (referenced to 100% modulation).
8 kHz Suppression: > 70 dB (referenced to 100% modulation).
76 kHz Suppression: > 80 dB (referenced to 100% modulation).
Pilot Protection: > -65 dB relative to 9% pilot injection, ± 1 kHz. 57 kHz (RDS/RBDS) Protection: better than -50 dB.
Connectors: Two EMI suppressed female BNC, floating over
chassis ground.
Maximum Load Capacitance: 5nF (at 10 ohms source impedance).
Maximum cable length: 100 feet/30 meters RG-58A/U.

## Analog Audio Input

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

# SPECIFICATIONS

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

## Analog Audio Output

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

## D/A Conversion

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

Digital Audio Input
Configuration: Stereo per AES/EBU standard, CS8420 Digital Audio Transceiver with 24 bit resolution, software selection of stereo, mono from left, mono from right or mono from sum. Automatically accepts and locks to input sample rates between 32 and 108 kHz.
Connector: EMI-suppressed RJ-45 female pinned according to StudioHub+® standards. Transformer isolated, balanced, and floating according to AES3 standard.

## Digital Audio Output

Stereo per AES3 standard. Digital Output sample rate can lock to the input, lock to an additional external sync source, or use the internal 48kHz rate.
Connector: EMI-suppressed RJ-45 female according to Studio- Hub+® standards. Transformer isolated, balanced, and floating according to AES3 standard.

## Digital Output Level

-24.0 to 0.0 dBFS peak, software adjustable in 0.1dB steps.

## External Sync Input

External Sync: Allows the output sample rate to be synchronized to an AES3 signal applied to the Ext.

# SPECIFICATIONS

Sync input connector.
(Does not accept Word Clock inputs)
Connector: EMI-suppressed RJ-45 female according to Studio- Hub+® standards. Transformer isolated, balanced, and floating according to AES3 standard.

## External Sync Range

Automatically accepts sample rates between 32kHz and 96kHz.
Connector: EMI-suppressed RJ-45 female pinned according to StudioHub+® standards, Transformer isolated, balanced, and floating according to AES3 standard.

## Remote Control Methods

External Modem or 10/100BaseTX Ethernet. Modem (support not yet available as of software version 2.2): Usersupplied Hayes command compatible external serial modem connected to rear-panel DB-9 male serial port.
Ethernet: TCP/IP control via web page interface and Java
(TM) remote control program included in the web pages. All
software is served from the built-in web server; there is nothing to install on the user's computer.
Ports Used: The defaults are TCP Ports 4545 and 4546 (for control and metering data, respectively
Connectors: Modem port - EMI-suppressed DB-9 male connector. Ethernet - Industry standard EMI-suppressed RJ-45 connector.

## GPI Interface

Support not yet available as of software version 2.2.
Connector: EMI suppressed DB-9 female connector.

## Power Requirements

Voltage: 100-250 VAC, 47-63 Hz., Less than 40 VA.
Power Connector: EMI suppressed IEC male. Detachable 3-wire power cords supplied for US and European use.
Power Supply: Internal. Overvoltage and short circuit protected. Meets EN55022, EN55011 Level B Conducted Emissions. EN61000-4-2, -3, -4, -5, -6 level 3 immunity compliant. Full international safety approval. CE marked.
Environmental: Operating Temperature: 32 to 122 deg. F / 0 to 50 deg. C for all operating voltage ranges.
Humidity: 0-95% RH, non-condensing.

# SPECIFICATIONS

## Omnia ONE AM

### Notes

Discrete I/O measurements have been made in "Bypass" mode (available in the Input/Output menu). All measurements made with the supplied "FACT_TEST" preset, which is available in the Preset Submenu.

### System Frequency Response

Complies with the NRSC emphasis curve within ± 0.50 dB, 30 Hz to 10 kHz. (At a setting of "10" on the HF EQ control)

### System *Signal to Noise Ratio

-80 dB de-emphasized, 20 Hz –– 10 kHz NRSC bandwidth,
referenced to 100% modulation).
*The measured noise floor will depend upon the settings of
the Input and Output Gain controls and is primarily governed
by dynamic range of the Crystal Semiconductor CS5361 A/D
Converter which is specified as >110 dB. The dynamic range of the internal digital signal processing chain is >144 dB.

### System Distortion

Less than 0.01% THD, 20 Hz – 5 kHz. (second order harmonic
distortion above 5 kHz is not relevant in the AM system due to the removal of harmonics by the system's 10 kHz low pass filter)

### System Stereo Separation

Greater than 65 dB, 20 Hz –– 10 kHz; greater than 70 dB typical.

### Analog Audio Input

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

# SPECIFICATIONS

### A/D Conversion

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

### Analog Audio Output

Left/Right Stereo. Electronically balanced.
Output Impedance 20 ohms.
Minimum load Impedance: 600 ohms.
Output Level adjustable from -2 dBu to +22dBu peak in 0.1dB steps.
D/A Conversion: Crystal Semiconductor CS4391, 24 bit, 128x oversampled.
Connectors: Two, EMI-suppressed XLR-male. Pin 1 chassis ground, Pin 2 "Hot".

### Digital Audio Input

Configuration: Stereo per AES/EBU standard, CS8420 Digital Audio Transceiver with 24 bit resolution, software selection of stereo, mono from left, mono from right or mono from sum. Automatically accepts and locks to input sample rates between 32 and 108 kHz.
Connector: EMI-suppressed RJ-45 female pinned according to StudioHub+® standards. Transformer isolated, balanced, and floating according to AES3 standard.

### Digital Audio Output

Stereo per AES3 standard. Digital Output sample rate can lock to the input, lock to an additional external sync source, or use the internal 48kHz rate.
Connector: EMI-suppressed RJ-45 female according to Studio- Hub+® standards. Transformer isolated, balanced, and floating according to AES3 standard.

### Digital Output Level

-24.0 to 0.0 dBFS peak, software adjustable in 0.1dB steps.

## External Sync Input

External Sync: Allows the output sample rate to be synchronized to an AES3 signal applied to the Ext. Sync input connector.
(Does not accept Word Clock inputs)
Connector: EMI-suppressed RJ-45 female according to Studio- Hub+® standards. Transformer isolated, balanced, and floating according to AES3 standard.

# SPECIFICATIONS

### External Sync Range

Automatically accepts sample rates between 32kHz and 96kHz. Connector: EMI-suppressed RJ-45 female pinned according to StudioHub+® standards, Transformer isolated, balanced, and floating according to AES3 standard.

### Remote Control Methods

External Modem or 10/100BaseTX Ethernet.
Modem (support not yet available as of software version 2.2): Usersupplied Hayes command compatible external serial modem connected to rear-panel DB-9 male serial port.
Ethernet: TCP/IP control via web page interface and Java
(TM) remote control program included in the web pages. All
software is served from the built-in web server; there is nothing to install on the user's computer.
Ports Used: The defaults are TCP Ports 4545 and 4546 (for control and metering data, respectively)
Connectors: Modem port - EMI-suppressed DB-9 male connector.
Ethernet - Industry standard EMI-suppressed RJ-45 connector.

### GPI Interface

Support not yet available as of software version 2.2.
Connector: EMI suppressed DB-9 female connector.

### Power Requirements

Voltage: 100-250 VAC, 47-63 Hz., Less than 40 VA.
Power Connector: EMI suppressed IEC male.
Detachable 3-wire power cords supplied for US and European use.
Power Supply: Internal. Overvoltage and short circuit protected. Meets EN55022, EN55011 Level B Conducted Emissions. EN61000-4-2, -3, -4, -5, -6 level 3 immunity compliant. Full international safety approval. CE marked.

Environmental
Operating Temperature: 32 to 122 deg. F / 0 to 50 deg. C for all operating voltage ranges. Humidity: 0-95% RH, non-condensing.

## Omnia ONE Multicast/DAB & Studio Pro

### Note

All measurements made using "Bypass" mode, which is available in the Input/Output menu.

# SPECIFICATIONS

## General Audio Specifications

Frequency Response: ± 0.50 dB, 20 Hz to 20 kHz with high pass filter disabled.
Distortion: Less than 0.05% THD 20 Hz – 20 kHz bandwidth.
*Signal-Noise Ratio: Greater than -100 dB, 20 Hz –- 20 kHz
bandwidth, referenced to 0dBfs
*The measured noise floor will depend upon the settings of the Input and Output Gain controls and is primarily governed by dynamic range of the Crystal Semiconductor A/D Converter which is specified as >100 dB. The dynamic range of the internal digital signal processing chain is >144 dB.

## Stereo Separation

Greater than 80 dB, 20 Hz –- 20 kHz; 90 dB typical.

## Analog Audio Input

Left/Right Stereo.
Electronically balanced.
Input impedance 10k ohms resistive.
Maximum Input Level +24 dBu.
Nominal Input Level: +4dBu (A +12dBu input results in –12dBFS input meter reading with Input Gain set to 0.0dB. A 0dBu input signal results in a -12dBFS input level when Input Gain is +12dB.)

## A/D Conversion

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

## Analog Audio Output

Left/Right Stereo. Electronically balanced.
Output Impedance 20 ohms.
Minimum load Impedance 600 ohms.
Output Level adjustable from -2 dBu to +22dBu peak in 0.1dB steps.
D/A Conversion: Crystal Semiconductor CS4390 24 bit, 128x
oversampled.
Connectors: Two, EMI-suppressed XLR-male. Pin 1 chassis ground, Pin 2 "Hot".

# SPECIFICATIONS

## Digital Audio Input

Configuration: Two-channel stereo per AES3 standard via CS8420 Digital Audio Transceiver with 24-bit resolution. Software selection of stereo, mono from left, mono from right or mono from sum. Automatically accepts sample rates between 24 kHz and 96 kHz.
Connector: EMI-suppressed RJ-45 female pinned according to StudioHub+® standards. Transformer isolated, balanced, and floating according to AES3 standard.

## Digital Audio Output

Stereo per AES3 standard. Digital Output sample rate software selectable for internal 48kHz, synchronize to AES input, or synchronize to auxiliary AES sync input (per AES-11 / DARS).
Connector: EMI-suppressed RJ-45 female according to Studio- Hub+® standards. Transformer isolated, balanced, and floating according to AES3 standard.

## Digital Output Level

-22.0 to 0.0 dBFS peak, software adjustable in 0.1dB steps.

## Digital Sync Input

Output sample rate can be synchronized to the signal present on the AES/EBU input or to the AES3 signal applied to the Ext. Sync connector.

## External Sync Range

Accepts 32kHz to 96 kHz for synchronization of the Digital Output signal to an external reference. Automatically accepts sample rates between 32kHz and 96kHz.
Connector: EMI-suppressed RJ-45 female pinned according to StudioHub+® standards, Transformer isolated, balanced, and floating according to AES3 standard.

## Remote Control Methods

External Modem or 10/100BaseTX Ethernet.
Modem (support not yet available as of software version 2.2): Usersupplied Hayes command compatible external serial modem connected to rear-panel DB-9 male serial port.
Ethernet: TCP/IP control via web page interface and Java (TM) remote control program included in the web pages. All software is served from the built-in web server; there is nothing to install on the user's computer.
Ports Used: The defaults are TCP Ports 4545 and 4546 (for control and metering data, respectively).
Connectors: Modem port - EMI-suppressed DB-9 male connector.
Ethernet - Industry standard EMI-suppressed RJ-45 connector.

# SPECIFICATIONS

## GPI Interface

Support not yet available as of software version 2.2.
Connector: EMI suppressed DB-9 female connector.

## Power Requirements

Voltage: 100-250 VAC, 47-63 Hz. Less than 25 VA.

## Power Connector

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

## Power Supply

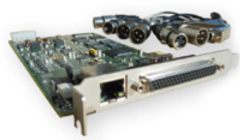
Internal. Overvoltage and short circuit protected. Meets EN55022, EN55011 Level B Conducted Emissions. EN61000-4-2, -3, -4, -5, -6 level 3 immunity compliant. Full international safety approval. CE marked.

## Environmental

Operating Temperature: 32 to 122 deg. F / 0 to 50 deg. C for all operating voltage ranges. Humidity: 0-95% RH, non-condensing.

# OMNIA.S4 CARD SOLUTIONS

## For FM and HD processing, webcasting, voice processing, and IP connectivity.



## OVERVIEW

Omnia.S4 cards represent a wide variety of options for the processing of FM, HD, voice, webcasting, and IP connectivity.  The Omnia.S4 card is the result of a joint project merging the technologies of Sound4 of Lyon, France and Omnia Audio to create a series of high performance, card-based solutions for any budget or application.

# FEATURES

## FM/HD Processing

**Omnia.S4 HD/FM2+ (two band)**
**Omnia.S4 HD/FM4+ (four band)**

## Internet Streaming Processing/Encoding

**Omnia.S4 x4 (four streams)**
**Omnia.S4 x8 (eight streams)**

## Voice Processing

**Omnia.S4 Voice ULA (four channel)**
**Omnia.S4 Voice ULA (eight channel)**
**Omnia.S4 Voice 1 A/D (single channel)**

## IP Audio Codec

**Omnia.S4 IP (able to encode and distribute up to 32 simultaneous audio links).**
**Omnia.S4 IP TX (encode card)**
**Omnia.S4 IP RX (decode card)**
**Omnia.S4 IP TX/RX (encodes and decodes)**

# SPECIFICATIONS

## Omnia.S4 for FM/HD Processing

**Omnia.S4 HD/FM – Powered by Sound4™**
**HD/FM 2+ (two band)**
**HD/FM 4+ (four band)**

Full-featured FM processor with wideband AGC, 4-band EQ and tone effects, stereo enhancer, 2-band AGC with Fidelity control and SIS™ (Sound Impact System) to manage spectral balance, 2-band limiter, stereo generator and ITU-R BS.412 MPX power limiter. Includes balanced analog, AES/EBU, and Livewire™ I/O, plus AES/EBU reference input. Composite MPX output with summing input for external SCAs. Configurable input fallback to backup sources, including file playback from host PC, in case of signal loss. Includes sophisticated Omnia.11 clipping routines. FM2+ version is easily upgradable to FM4+ version.

# SPECIFICATIONS

## Omnia.S4 for voice

### Omnia.S4 Voice ULA 4-Channel Card – Powered by Sound4™ with Livewire™ I/O

Voice processing card with four mono channels of high-quality 192kHz processing for microphone audio, including advanced de-esser, 3-band noise gate, 3-band dynamics processor, 4-band parametric EQ and brick-wall limiter. Unique "Preset Centralization" system automatically shares presets between all Omnia.S4 Voice cards on the network.

### Omnia.S4 Voice ULA 8-Channel Card – Powered by Sound4™ with Livewire™ I/O

Voice processing card with eight mono channels of high-quality 192kHz processing for microphone audio, including advanced de-esser, 3-band noise gate, 3-band dynamics processor, 4-band parametric EQ and brick-wall limiter. Unique "Preset Centralization" system automatically shares presets between all Omnia.S4 Voice cards on the network.

### Omnia.S4 Voice 1A/D Card – Powered by Sound4™

Single channel voice processor card with analog, AES/EBU and Livewire™ digital I/O. High quality 192kHz processing for microphone audio includes advanced de-esser, 3-band noise gate, 3-band dynamics processor, 4-band parametric EQ and brick-wall limiter. Unique "Preset Centralization" system automatically shares presets amongst all Omnia.S4 Voice cards on the network. Base card includes one channel of processing, but may be extended to support up to six channels of voice processing.

## Omnia.S4 IP Card Solutions

Omnia.S4 IP cards are a complete solution for sending programs to transmitter sites over IP connections. Much more than a simple "point to point codec", the Omnia.S4 IP codec is able to encode and distribute up to 32 simultaneous audio links.

### Omnia.S4 IP TX

This is the "encode card", typically installed in the studio to feed IP links to transmitter sites. The Omnia.S4 IP/TX encodes and streams only; requires an Omnia.S4 IP/RX or Omnia.S4 IP TX/RX card at the "far end" for audio decoding.

### Omnia.S4 IP RX

This is the "decode card", typically installed at the transmitter site to decode and distribute program

# SPECIFICATIONS

audio. The Omnia.S4 IP/RX decodes only; requires an Omnia.S4 IP/TX or Omnia.S4 IP TX/RX card at the studio origination site for audio encoding.

### Omnia.S4 IP TX/RX

This is the "encode + decode card". It is a true CODEC, capable of both sending and receiving encoded audio via IP links.

## Omnia.S4 for Webcasting

### Omnia.S4 x8 – Powered by Sound4™

Omnia.S4 x8 delivers eight channels of audio processing and encoding for Web streaming applications. On-card audio processing includes wideband AGC, 4-band EQ and tone effects, a stereo enhancer, 3-band AGC with Fidelity control and SIS™ (Sound Impact System) to manage spectral balance, a 4-band limiter and brick-wall final limiter. Bitreduced encoding is handled by host PC CPU. Supports MP3, AAC, and HEAAC v1/v2 encoding, plus 3GP-compatible encoding for mobile phones. Compatible with all standard streaming server platforms including Darwin, Flash, Helix, Icecast 2, Red5, Shoutcast and Wowza using HTTP/ICY, RTSP/RTP Unicast, and RTMP protocols. Accepts audio input via Axia Livewire™ connection or PCI (WDM driver). Advanced metadata management with scripting capability ensures compatibility with automation/playout systems.

### Omnia.S4 x4 – Powered by Sound4™

Omnia.S4 x4 delivers four channels of audio processing and encoding for Web streaming applications. On-card audio processing includes wideband AGC, 4-band EQ and tone effects, a stereo enhancer, 3-band AGC with Fidelity control and SIS™ (Sound Impact System) to manage spectral balance, a 4-band limiter and brick-wall final limiter. Bitreduced encoding is handled by host PC CPU. Supports MP3, AAC, and HEAAC v1/v2 encoding, plus 3GP-compatible encoding for mobile phones. Compatible with all standard streaming server platforms including Darwin, Flash, Helix, Icecast 2, Red5, Shoutcast and Wowza using HTTP/ICY, RTSP/RTP Unicast, and RTMP protocols. Accepts audio input via Axia Livewire™ connection or PCI (WDM driver). Advanced metadata management with scripting capability ensures compatibility with automation/playout systems.

All Omnia.S4 cards are PCI-Express x1-type cards, using the host PC for power and control — card continues to run even if host PC OS halts or reboots. Startup delay is < 2 seconds. Link and Share-ready for agnostic support of automation / playout systems.

# OMNIA A/XE

## Basic software for the encoding and processing of internet streams


## OVERVIEW

Omnia A/XE can process audio for a variety of applications, bitrate-reduced and linear. It runs in the background as a Windows service, can be fully-managed and configured remotely with a web browser, and can even process and encode multiple streams in various formats simultaneously.

# FEATURES

Genuine Omnia processing to improve audio levels, loudness and perceived quality.

Software only, no special cards required

Runs as a Windows service in the background. No need to log in.

Managed from anywhere through a web browser, locally or across the Internet

Each license = one stereo input. The user can add each license to the same PC or separate PCs.

Each program input can be processed and encoded in multiple ways, and sent to multiple servers simultaneously.

Processed audio can also be sent to a local sound device for monitoring.

## Additional Features

High-performance, low memory footprint, native application

Can operate with Virtual Audio Cable driver (downloadable at ftp://beta.zephyr.com/~tlscorp/pub/Omnia/AXE/VAC/). This
allows A/XE to accept audio from a playout system or other applications on the same PC. It can also be used to feed the processed audio from A/XE to another application on the same PC.

All configuration information is stored in a single XML file for simple configuration backup/restore.

# IN DEPTH

## Seamlessly integrates with other software

The new Virtual Patch Cable allows Omnia A/XE to receive, process, and send audio to other software on the PC. Internally encoded Shoutcast or Wowza server streams can be “tagged” with “now-playing” information received from automation systems or another application. We’ve even built-in a scheduler to allow streams to be started and stopped at specific times, as well as processing presets can be changed on a schedule, perhaps processing the morning show differently than the afternoon one.

# IN DEPTH

Included with A/XE is a license to the multi-channel version of the Axia IP-Audio driver. Customers with a Livewire installation can use the Axia IP-Audio driver to read or write audio directly from the network without the need for hardware audio cards.

## Metadata

Accepts metadata from a variety of sources and uses it to “tag” the audio stream. This information is then sent to the media server and (eventually) displayed to the user (in a player-specific manner). Metadata can be accepted over TCP/IP and UDP, or from text files. A/XE can accept just about any format, from simple, line-based messages to XML messages and anything in between. We in¬clude a set of metadata filters (small scripts, using the Lua scripting language) which can be further edited and customized.

## Processing by Omnia

Omnia A/XE features adjustable wide-band AGC with a three-band compressor/limiter, IIF EQ and low-pass filter, and a precision look-ahead final limiter to prevent clipping. Resulting streams are cleaner, clearer, and with more presence and detail.

# SPECIFICATIONS

## Hardware Requirements

32-bit Windows XP and later

Minimum 512MB RAM

20MB free hard-drive space

Network Interface Card

## Codecs

MP3, AAC, HE-AAC, HE-AAC v2.
The highest quality codecs from Fraunhofer

# SPECIFICATIONS

## Streaming Servers Supported

ShoutCAST-compatible servers, including ShoutCAST v2

Icecast

Adobe Flash Media server

Wowza server

Live365

Windows Media Server

# OMNIA F/XE

## Processing/encoding software for file-based material


## OVERVIEW

Combines Omnia audio processing with the Fraunhofer MP3 and AAC codecs for high quality processing preparation for podcasting or filebased streaming/encoding.

# FEATURES

Genuine Omnia processing to improve audio levels, loudness and perceived quality.

Software only, no special cards required

Able to read PCM WAV files, MPEG Layer-2 and MPEG Layer-3 source files.

Can automatically send the output file to an FTP server.

Can notify the user by email if problems are detected

Logs are kept during processing so you can find the source of a problem

## Additional Features

Read metadata from external files and embed the information as ID3 tags in the output files.

Encode the output audio using MP3 or AAC (including HE AAC and HE AAC v2), or save linear PCM WAV audio files.

Core processing and encoding uses high-performance, low memory footprint, native application

Drop files on FileProcessor for on-demand processing and encoding, or automate your work using FolderBot to watch folders for new files and automatically process them as they arrive.

You can define multiple configurations in FileProcessor. Each configuration can process and encode the files with a different set of parameters or send the output to different locations. This makes it easy to define and reuse project-specific configurations.

FolderBot watches one or more folders and automatically processes the files as they are added to the folder. Files can be handled differently based on the watched folder.

# IN DEPTH

## Metadata

Omnia F/XE will read metadata from external files and embed the information as ID3 tags in the output files. The core processing and encoding uses high-performance, low memory footprint, native application. You can use drop files on FileProcessor for on-demand processing and encoding, or automate your work using FolderBot to watch folders for new files and automatically process them as they arrive. Multiple configurations are able to be defined in FileProcessor. Each configuration can process and encode the files with a different set of parameters or send the output to different locations. This makes it easy to define and reuse project-specific configurations.

## Included with F/XE

F/XE includes a license to the multi-channel version of the Axia IP-Audio driver. Customers with a Livewire installation can use the Axia IP-Audio driver to read or write audio directly from the network without the need for hardware audio cards.

# SPECIFICATIONS

## System Requirements

Windows XP or later with 20MB of free disk space

Microsoft .NET client framework 4.0

Internet access

## Codecs

MP3, AAC, HE-AAC, HE-AAC v2.
The highest quality codecs from Fraunhofer

# OMNIA 9/XE

## The ultimate, high-quality processing/encoding software with proprietary audio correction and sonic management

## OVERVIEW

Based on the technology found in the popular Omnia.9 audio processor, 9/XE is not simply a streaming processor-encoder, but a complete audio management system which will actually improve the flaws found in most recorded source material – both music and voice – as well as address the specific technical challenges of internet distribution.

# FEATURES

Exclusive "Undo" technology with De-clipper prevents listener fatigue by removing distortion and selectively undoing the over-compression so common in mastering today.

Optimizes sound quality of low bit rates by literally removing distortion components so that they do not waste bits during encoding.

6-band Parametric EQ for your signature sound.

Downward Expansion (source noise reduction).

Multiband stereo enhancer

## Additional Features

Software only, no special cards required

Includes Virtual Audio Cable to receive audio from other programs on the same machine

Includes AXIA LiveWire driver

Runs as a Windows service in the background,
no need to log in

Manage from anywhere with NfRemote, locally or
across the internet

Up to 16 fully independent stereo processors in one instance, and up to 8 instances on one machine. Pay only for what you need. Upgrades available.

Local monitor output with patch-point selection and full speaker controller

Flexible remote control application with touch screen support, comprehensive instrumentation, and remote audio streaming of any patch-point, also includes full speaker controller

Separately adjustable sample rate (high quality conversion) and gain control per encoded stream.

Extremely high audio quality, efficient CPU usage and low memory footprint

# IN DEPTH

Omnia.9/XE comes with both a GUI application and a service which contain the exact same processing. During initial set-up (sound card configuration etc), use the
GUI application. Once initial configuration is done and tested, switch over to using the Service, which you can then control with NfRemote from any computer.

Everything can be controlled with NfRemote except for which sound cards to use. Omnia.9/xe and NfRemote are standard Windows 32-bit native applications and do not use Microsoft.NET or similar.

Omnia.9/XE is primarily designed for streaming and only has one local sound card output. However NfRemote has built in dedicated PCM audio streaming for monitoring, so that you can monitor with low delay from any computer, for example while adjusting the processing.

9/XE can encode audio to MP3, AAC, HE-AAC v1/v2 h (aacPlus), MP2 and WMA. Low complexity AAC (AAC-LC) and high efficiency AAC (HE-AAC) are both supported. AAC has been standardized under both MPEG-2 and MPEG-4. The format most commonly used is MPEG-4 AAC-LC. Often this is called just 'AAC'. HE-AAC adds Spectral Band Replication to AAC and it is sometimes called AAC+ (sometimes seen as 'aacPlus' or 'AACplus'). There is also an HE-AAC v2 format which adds parametric stereo optimizations to HE-AAC. Sometimes this is called AAC+ v2 or Enhanced AAC+. 9/xe can also use Windows Media codecs installed on the system, 48kbps or higher.

9/XE can directly feed SHOUTcast-style servers (SHOUTcast, h Icecast, Steamcast, etc.). The Wowza server is also supported for streaming to Flash clients. Windows Media streams can be sent to Windows Media server.

## A few words about Undo

Undo is two stages:

First, the de-clipper removes distortion by detecting clipped edges of the waveform and resynthesizing the missing part. Unlike simpler algorithms, no distortion is
ever created as the resynthesizing is performed entirely in frequency domain.

Second, the amount of short-term dynamics is detected for each of 5 frequency bands, and automatically controls the threshold and expansion ratio of 5 upwards expanders, to undo excessive compression and peak limiting.

# IN DEPTH

Both techniques together result in an incredible "is that really the same recording" level of improvement. Audio quality of low bit rate codecs is also vastly improved, as a less distorted waveform is less complicated for the codec to encode (thus using fewer bits) and more dynamic, punchy sound gives the codec a place to hide the bit rate reduction artifacts.

# SPECIFICATIONS

## System requirements:

9/XE will run on Windows XP or newer. Minimum requirements are Core 2 Duo, 512 MB RAM.

## General

A Core i7 2600 and 4 GB RAM comfortably runs 16 stereo processors with several encoders each.

Supports multiple ASIO and WDM (Wave/Direct-Sound/Kernel Streaming) audio interfaces simultaneously. Input selection can be done on the fly.

Simultaneous MP3/AAC/aacPlus/MP2/WMA encoding, compatible with Shoutcast, Icecast, Wowza and Windows Media server.

# OMNIA VOCO 8

## Up to eight individually processed mics networkable through an entire facility

## OVERVIEW

Voco 8 is the world's first voice processor with:

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

Omnia Voco 8 is adaptable to all different voice characteristics. From "natural tone" to "big", everything is possible in just a few clicks. Plus, in advanced mode, the Omnia Voco 8 is also the perfect tool for production studios.

# FEATURES

## Processing

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

## Processing Chain Extra Features

- Low Pass / High filters
- Phase scrambler
- Dominate-It (when main talent speaks, it reduces the other participants to keep intelligibility)
- 2 Bus Mix
- Presets centralization and sharing
- Multi-Studio Ready
- Session Recall
- Link & Share ready
- Main sampling process frequency 192 kHz.
- Ultra low delay ~3 milliseconds

## Inputs

Omnia Voco 8 is powered with a first class mic-preamplifier, adaptable to any voice. It is also possible to use Voco in Analog line level, AES EBU, Livewire and AES67.

## Outputs

Each output is available on AES EBU, Livewire and AES67.

## Bus Mix

Omnia Voco 8 offers two independent Bus-Mix to group Mics in a single output.  This is a great feature to simplify use.

# IN DEPTH

## A User Friendly Control Interface

- Control all Mics on one screen
- Settings from one single screen. No more opening and closing windows to go from one function to another.
- "Basic" mode and "Advanced" mode: "Basic", the mode where everything is simple and rapid. "Advanced" mode, to explore all of the processor functionalities. Ultra-rapid VU meters for true control over modulation.
- Unlimited "Undo/Redo" versioning function for presets with the possibility of recall.
- Innovative "compare" function with reference notion.
- Works on operating systems: Microsoft: XP SP3, Seven 32& 64 bits, 2008 R2. Linux (Debian)

## GUI #1: Studio Mode GUI

Easy sound setup for each talent.

## GUI #2: Live Mode GUI

The dedicated graphical user interface shows all 8 mics, status, affectations and user names. Recalling a mic to a user is done in two clicks. Another feature is "Session Recall". It is possible to save all mics' characteristics + user presets. Then recall them all in one click! Omnia Voco 8 is also externally automatable for dayparts and automatic session recall.

## HQSound 192 kHz

Omnia Voco 8 is powered with the HQSound 192 kHz algorithm. While it is running at 192 kHz for dynamics stages, HQSound provides the possibility to control important amounts of gain range without any pumping or smashed sound effect. The result: a strong and robust sound.

## Effective 3-band noise gate

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

# IN DEPTH

## S.I.S – Sound Impact System

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

## Preset Sharing

Another unique feature, preset sharing authorizes users to synchronize presets and all changes on an unlimited group of Voco 8s. No need to access to each processor to load or change presets, Preset Sharing will automatically update all your processors.  Moreover, when a new setup for a new talent is created in one studio, all other Voco will receive the new talent preset in it's memory.

## Multi-Studio Management

The Omnia Voco 8 can process separately up to 8 microphones. Thanks to Multi-Studio mode, an Omnia Voco 8 can distribute these resources over several studios. For example, you have two studios to equip with three microphones for each studio: with a single Omnia Voco 8  you can "split in two" to get two processors which operate separately on each studio. Moreover studios may save and recall their own sessions.

## Security

Omnia Voco 8 offers the possibility to chain a second processor as a backup unit.

# SPECIFICATIONS

## Mic Input

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

## Line Level Input

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

## Digital Input

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

## Livewire/AES67 Input

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

## AES/EBU Input Sync

- Quantity / Connectors: 1 BNC female connector
- Sync type: Word Clock - 32 to 192 kHz
- Level: 1 to 6 volt

# SPECIFICATIONS

## Digital Output

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

## Livewire/AES67 Output

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

## GPI Interface

- Connector: Standard DB-15

## Audio Performances

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

## Compatible Operating System for Remote Control Software

- Microsoft Windows: Windows XP SP3 - Windows 7 (32 & 64 bits) - Windows 8 (32 & 64 bits) - Windows Server 2008 R2 - Windows Server 2012, Linux: Linux (Debian)

## Omnia Voco to Client Communication Interface

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

# OMNIA MPX TOOL/ MODULATION ANALYZER

## Modulation analysis and monitoring made easy

## OVERVIEW

MPX Tool/Modulation Analyzer brings all of the modulation analysis tools from our acclaimed Omnia.9 processor to a stand-alone platform, including an over-sampled digital oscilloscope, RTA (real time analyzer), and FFT spectrum analyzer to provide you with everything you need to take a critical look at your station's modulation and transmission characteristics.

# FEATURES

## Stereo Baseband Audio Destinations

Stereo baseband audio can be sent to MPX Tool/Modulation Analyzer via analog, AES, Livewire/AES 67 inputs, or by two composite MPX inputs.

## FM Tuner is Standard

A built-in FM tuner demodulates and decodes FM, HD Radio, DAB, and DAB+ signals, providing a means by which to analyze and monitor your -- and other -- off air signals.

## Power Supply

Dual redundant power supplies are standard.

## Loudness Management

Helps to identify loudness-robbing peaks and overshoots.

## Modulation Level Management

Check your modulation or the level of your pilot or RDS signal.

## Proof of Performance

Compare the output of your processor to your off-air signal to verify the performance of your exciter and transmitter.

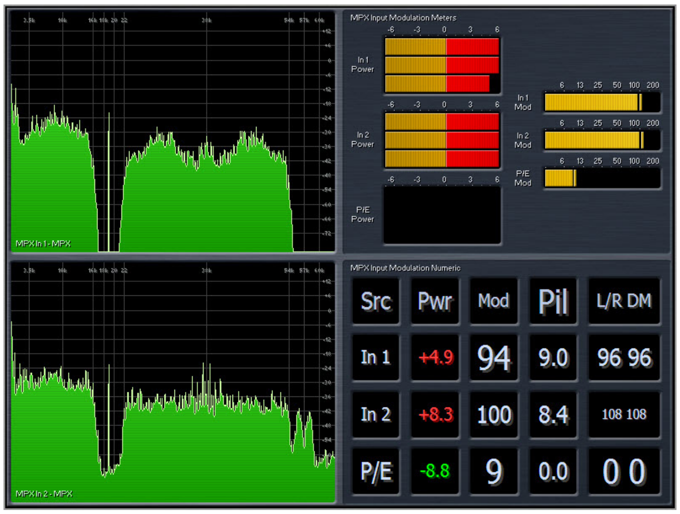
## Align Your Backup Processor

Plug in your backup processor and compare its performance to your main air chain.

# IN DEPTH

## Taking Control

A front panel LCD touchscreen display provides local control and monitoring. Comprehensive remote software provides full control from remote locations, including streaming audio back to your PC.



# SPECIFICATIONS

## General

- Standard 2RU 19" rack mount.
- PSU: Dual, redundant universal 100-250 VAC PSU

## Standard I/O Connectivity (Rear Panel)

| I/O | Connection Type | Function |
|---|---|---|
| Analog In | Neutrik XLR Female Connector | L/R Analog In |
| Analog Out | Neutrik XLR Male Connector | L/R Analog Out |
| O.D./AES In | Neutrik XLR Female Connector | Omnia.Direct/AES In |
| O.D./AES Out | Neutrik XLR Male Connector | Omnia.Direct/AES Out |

## Sync to internal (48k), AES In, or Livewire (AES 67)

| | | |
|---|---|---|
| RF In | 50-Ohm BNC Female | Tuner Input |
| LAN 1 | Ethernet Port | Control |
| LAN 2 | Ethernet Port (1G) | Livewire |
| IEC | Power Connector (x2) | AC Power |

# SPECIFICATIONS

## Livewire Features

Livewire Control Protocol (LWCP)
Livewire GPIO
Livewire low-latency
AES67 Fully Compliant
Sync master capability

## Remote Control User Interface

Uses existing NFRemote dedicated application.

## Processing

Dual FM/HD Radio/DAB/DAB+ tuner

## GPIO

8 in, 8 out

## Compliance

CE, RoHS, FCC

# PROGRAM DELAY MANAGER

## Profanity Delay Reinvented


# OVERVIEW

- PD Alert™ instantly emails time-stamped audio files whenever Dump is pressed
- Files capture what took place both on- and off-air
- Seamlessly builds and exits delay
- Configurable delay time, build and dump options
- Delayed IP data, serial streams and GPIO sync'd to audio

# FEATURES

## It's About Time

Leave it to 25-Seven Systems to re-invent the profanity delay. Program Delay Manager (PDM) brings the possibilities of the Internet age to a "stand-alone box" technology that hasn't advanced much since the 1980's. Ease of use, transparent audio quality and program director friendly features converge in PDM to take an old process to a new level.

## The Air Check is in the Email

Program Directors have more on their plates today than ever before. There's no way anyone can monitor every broadcast hour of every day, but PDs need to be the first to know what happened when that "dump" button got pressed.

With Program Delay Manager's patented PD-Alert™ feature, two time-stamped audio files capturing what took place both on air and off air get internally archived and emailed to the PD (or GM, or CE, or the legal team) every time questionable material is "dumped".

For stations serious about protecting their license, PDM provides an instant log record establishing your station's action and intent to keep the airwaves clean.

## 99 Seconds Of Delay Your Way

PDM comes standard with 99 seconds of stereo audio delay, and a dump button that can be set to remove any number of seconds you choose.

Build a delay through pre-rolling, time expansion or audio file play-out capabilities built right into PDM. Exit a delay through time compression or use the Cough button to simply wait and exit.

Dump audio through the standard "cut and rebuild" method, or use PDM's Overkill™ feature to play a "fill" file. Overkill allows you to select a show specific file from a list and play it over the dump buffer instead of collapsing the delay.

## How PDM Does It

### Superior Audio Algorithm Quality

25-Seven has a well-deserved reputation for offering the industry's most transparent time compression and expansion algorithms. Your listeners probably won't appreciate our superior, artifact-free audio because they won't perceive it's in use!

# FEATURES

### Flawless Expansion/Compression

25-Seven Systems' imperceptible audio time compression algorithms serve up smooth, crisp, stutter-free audio in PDM, even on stereo music. Unlike other products, we never splice at level thresholds or alter pitch. Clean audio is what we do best... now you can be sure the content is "clean" as well! Better algorithms mean delays can be rebuilt faster, so you can safely get back to callers. Build or Exit rates can be adjusted in real time, so you can be more or less aggressive, depending on audio content.

### Audio, RDS, Data Streams and GPI/O Stay Synced

PAD or "now playing" data streams are delayed in precise synchronization with the audio as it grows, shrinks or whenever the dump button is pressed. PDM's data-follow-audio capabilities allow flexible synchronization from any data input to any data output. For example, serial data entering the RS-232 input can be routed to an IP output while remaining synchronized to the audio. 2 independent data delays are supported, and GPI/O closures stay in sync, too.

### Future-Proof Audio Quality

Superior balanced analog I/O, with AES digital standard. 85dB s/n, response 25Hz-18kHz (+0/-0.2dB) and 0.02% THD+N... even during compression/expansion. Audio is always linear, so no lossy data reduction enters your signal path.

### AES Digital, Balanced Analog or Livewire AoIP

The first program delay to provide Audio over IP (AoIP) and control over Axia Livewire audio networks, PDM comes in two models: one with balanced analog and AES digital I/O and the other with AoIP for Livewire. Whether you already have a Livewire network or you want to keep your plant AoIP-capable, PDM has you covered with Ethernet connectivity.

### Superior Control

Choices, choices! PDM presents you with easy-to-use front panel controls, designed for the rigors of radio. Contact closure commands can be synced to the audio delay by the smart, programmable 8x8 GP I/O. Full bi-directional serial control over both RS232 and IP include advanced real-time status monitoring of parameters such as current delay depth and audio levels. A comprehensive web interface allows your PDM to be managed from nearly anywhere. Our Multi-View web feature permits networks and big facilities to monitor and manage up to 20 PDM's from a single browser screen.

### Web Configurable

Say goodbye to hieroglyphs. Navigating through "set and forget" parameters is a breeze with our built-in web server. Change your settings, upload audio files and manage PDM's dump archives using simple browser screens, so you don't waste time trying to enter data though an ill suited LCD interface. Talk to PDM over your LAN or WAN. What could be easier?

# IN DEPTH

## Web Interface

In addition to controlling Program Delay Manager using GPIO (contract closures), PDM comes with a built-in, password-protected web server, allowing you to remote control your unit across a local or wide area network.

The server gives you five separate pages for complete and convenient control over your PDM.

## Front Panel

An Adobe Flash-based application replicates PDM's front panel on your web browser, so every button and display is present and functions just like the real front panel. Through careful client-server communications management, round-trip latency is almost imperceptible, creating a seamless user experience. You can even control PDM from multiple computers. Just open a web browser interface on each, and anything you do on one computer will be reflected on the others, as well as on PDM's physical front panel.

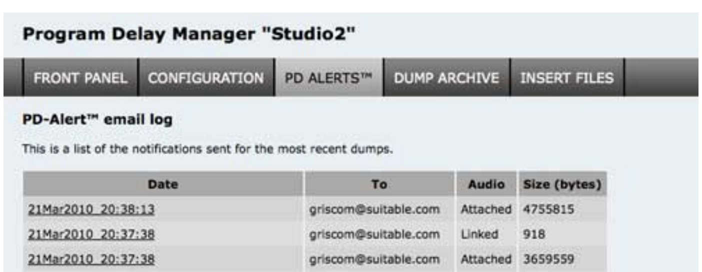
## Configuration

Tired of learning hieroglyphics just to configure a profanity delay? Navigating though "set and forget" parameters is a breeze with the PDM's Configuration page. You'll find obvious control with all your settings on one simple screen, so you don't waste time entering data though an ill-suited LCD interface.

## PD Alerts™

A dedicated page lists all of the PD Alert emails the PDM has sent to your chosen staff.

## Dump Archive



# IN DEPTH

A Dump Archive shows all your saved dumped audio files. Easily review just what's been cut out of your air stream.

## Insert Files

Easy management of the files you can use for quickly building your buffer at the beginning of the show. No more flash drives and cryptic file names!

# SPECIFICATIONS

## Power Input

PDM comes with a standard IEC C14 power connector.

## Network

PDM connects to a standard 10/100Base-T network connection. This port is used for Axia Livewire™, synchronization to a network time server, and secure remote control via a web browser. If connected to the Internet, it should be behind a hardware firewall.

## Analog Inputs and Outputs

Stereo inputs are electronically balanced XLR females, pin 2 hot, with a load of 20kΩ: this makes it compatible with all modern electronically-balanced outputs. If fed from a transformer-balanced output, we recommend bridging a 680Ω resistor between pins 2 and 3. Outputs are electronically balanced XLR males, pin 2 hot, designed to feed a load of 600Ω or greater. Input and output sensitivity default levels can be set from the front panel, and can range between +20dBu and -10dBu.

## Digital Inputs and Outputs

When set to AES/EBU via the configuration menu, this input conforms to IEC 958 Professional (5v p-p, 110Ω balanced) on XLR connectors. When set to s/pdif, the voltage and impedance switches to IEC 958 Consumer (.5v p-p, 75Ω unbalanced): connect signal to pin 2 and shield to pins 1 and 3. Digital output (selectable AES or s/pdif) is always active, regardless of whether you are using analog or digital inputs. PDM will lock to any valid 32 kHz, 44.1 kHz, or 48 kHz signal at the digital input connector, even if you have selected analog for the input. In that case, the digital input controls PDM's internal sample rate. If PDM is not connected to a digital input, it uses its own high-reliability 44.1 kHz sample clock

# SPECIFICATIONS

## Axia Livewire Version Inputs and Outputs

On the Livewire version of PDM, audio connections are exclusively via the network. PDM-Axia also supports Livewire-based GPIO.

## GPIO

Eight parallel control inputs and eight parallel control outputs appear on a DB-25 connector. Input and output functions are assigned through a configuration menu on the front panel. Inputs and outputs are opto-isolated for easy interface to other equipment. A +5v supply and ground are also brought out to the DB-25 for simple remote controls using pushbuttons and LED status readouts. The +5v supply can carry 200 mA, more than adequate for 8 LEDs and 8 logic inputs. It is protected by an internal, self-resetting thermal circuit breaker.

## Detailed Specifications

### Audio

S/N ≥ 84 dBA with 10 dB headroom (≥94 dB dynamic range); THD @1 kHz < .01%; IMD (IHF) < .01%; Frequency response ± 0.5 dB, 20 Hz – 20 kHz, measured analog input to analog output.

### Dimensions

1RU (rack unit); 19" W (with rack ears) x 12" D x 1.75" H (483 x 305 x 44mm)

### Power

100-240 VAC, 50/60 Hz; typical consumption 32 VA.

# AUDIO TIME MANAGER

## Create Time... Shift Time... Control Time


## OVERVIEW

- Compress and time-shift programs in real time
- Insert audio into live programs with no loss of content
- Smoothly introduce random-starting events like press conferences and concerts
- Generate extra ad and promotional inventory and revenue

# FEATURES

## Imagine A Longer Hour

You could run more local news, promos, ad-libs, whatever you need! 10% speed-up without pitch change, glitches or artifacts, so smooth listeners won't even know it's in use, even on stereo music. Audio Time Manager's proprietary algorithms are designed for today's radio needs.

Depending on program material, you will be able to add several minutes an hour with complete intelligibility.

## Imagine Doing This On-The-Fly with No Math and No Operator Training

Our simple two-button interface lets you automatically create local inserts anywhere in the hour, and still join the network perfectly. Simple tape-like operation lets you "pause" a live feed, insert content, and rejoin on cue, without missing a beat.

## Now, Imagine What ATM Can Do for You

### Eliminate Back-Timing Hassles

Make the network start when you're ready, even if you don't know how long a local break will be. Just press Record to mark the beginning of the feed. Make the break as long as you want, and push Play. Audio Time Manager will play the feed from the top, with just the right amount of time compression, and then seamlessly join real-time. You can even have your automation system control Audio Time Manager... and do it all automatically!

### Smoothly Introduce Random-Starting Events Like Press Conferences or Concerts

When the event starts, press Record, or use our Cue feature to rewind or skip forward to a pickup point. Ad-lib any length intro and then press Play. It's as if they were waiting for your cue! No more awkward, time-wasting talk-ups.

### Drop A Custom Break in the Middle of a Continuous Program

Insert ID's, traffic updates, breaking news, commentary on an event... without missing anything. Just listen for any kind of pause—a beat between news stories or a breath—and press Record. Take your break and then press Play. Your listeners will hear the new content, and the entire original program.

### Create Extra Ad, Promo or Underwriting Inventory When Needed

ATM can protect revenue and create programming options you've only dreamed about.

# FEATURES

## How ATM Does It

### Virtually Imperceptible Time Compression

At 10% faster — an additional 6 minutes per hour — most listeners won't even be aware ATM is in use. Our technology makes this possible without choppiness, stutters or fuzziness. Even a smooth 20% speed-up is possible on some program material.

### No Complicated Programming

Two-button operation means you can react on-the-fly to severe weather alerts, late-starting press conferences, concert intermissions or breaks in a sporting event.

In this example, a three-minute per hour (5%) rate will catch real-time at 10:57:14pm, in 5:08 minutes. Adjustments always sound smooth and glitch free.

### Continuous Recording

Think of ATM as a continuously recording one-hour stereo buffer, where audio is always recording, no matter what you are playing. You can always cue back to the correct audio, even if you pressed Record a few seconds late.

### Keeps Programs Sounding Natural

Some technologies can drastically cut into pauses, destroying a personality's careful delivery. Our proprietary algorithms intelligently process speech: pauses stay in proportion; pacing and inflection are maintained. Even stereo music stays clean and steady.

### "Future-Proof" Audio Quality

Superior balanced analog I/O, with AES digital standard. 85 dB s/n, response 25 Hz-18 kHz (+0/-0.2dB) and 0.02% THD+N... even during time compression. Audio is always linear; no lossy data reduction to mess up your sound

### Axia Livewire Compatible

ATM operates directly on Livewire AoIP networks.

### Works with Automation Systems and Network Receivers

Expand local breaks and have perfect joins automatically. Keeps network cues in sync with audio, even during compression or delay.

# FEATURES

### Flexible Remote Control

Run ATM from its front panel. Add remote push-button and LEDs, or connect automation systems to the built-in 8x8 GPI/O. Control using serial RS-232 or via your station's computer network.

# IN DEPTH

## Virtually imperceptible time compression

At 10% faster—an additional 6 minutes per hour—most listeners won't even be aware ATM is in use. New technology makes it possible to do this with no choppiness, no stutters, no fuzziness, and absolutely no change in pitch or timbre. Even a 20% speed-up is glitch- and artifact- free... though at 20% with most sources, listeners will notice things being much faster than normal. Obviously, sources that speak slowly or music with a relaxed tempo can be sped up more than those already at breakneck speed.

## No complicated programming needed

Simple two-button operation means you can react on-the-fly to unplanned events like weather alerts, late-starting press conferences, or holds in a concert or sporting event. Our Time/Rate Management Calculator™ (TRMC) adjusts to what you want to do.

## Total flexibility with compression ratios

Change the upcoming playback ratio during record. Change it while playing. Or during playback, tell ATM when it should smoothly join real-time and the proper ratio is applied automatically. Change these ratios again and again, as many times as you want.

## Even advanced operations are simple

Most settings need just a few key presses. Softkeys mean that users don't have to memorize menu structures.

## Keeps programs sounding natural

Some technologies can drastically cut into pauses, destroying a personality's delivery. Our proprietary algorithms intelligently compress tiny sections of soundwaves: pauses stay in proportion; pacing and inflection are maintained.

# IN DEPTH

## Automatic ramp-down at end of compressed playback

No abrupt shifts: transitions back to real-time are undetectable.

## You're in complete control, all the time

Change vital parameters, even during compressed playback. Adjustments are smooth and glitch-free.

## Flexible remoting

Run ATM from its front panel. Wire pushbuttons and LEDs to the built-in 8x8 GPIO. Connect using serial RS-232. Securely control ATM from a browser over your network!

## Works with your satellite and automation system

Advanced GPIO features let ATM lock incoming contact closures to incoming audio, and then fire outgoing closures when that specific audio plays out. Local liners, ID's and spots happen at the right point in the network program, whether you're using time compression or not.

## Automatic creation of extra availabilities during network breaks

Special commands let your traffic department put more spots in a break than would normally fit. ATM works with your satellite or network cue receiver and automation system to automatically fit those spots in, without an operator, and without losing any network content.

## Time-Aware

ATM synchronizes itself to network clocks. Hook ATM up to any local data network that has Internet access, give it the address of a Network Time Server, and it'll always know the exact time. Or if your station has its own time server, lock ATM to it.

## Clear, concise display

The large, backlit LCD display is simple to understand and easy to read, even from across the control room.

## Quiet operation

The only moving part is an ultra-quiet low-speed fan. Run ATM next to a live mic!

# IN DEPTH

## "Future-proof" audio quality

Excellent sound quality, both balanced analog and AES digital stereo I/O mean you can use ATM now and keep using it tomorrow. No data reduction to mess up your sound. Time-compression algorithms are good enough to use on critical music programs.

## ATM With Automation

Audio Time Manager's sophisticated GPIO features allow you to remote control the unit using your automation system and satellite / network receiver. Local breaks can be extended by simply entering more spots in your program automation's log than would normally fit. ATM will delay the return to network exactly long enough to accommodate the added spots, without losing any network content, and without operator intervention. Affiliates can use ATM to add more local content to network programming, or add availabilities when needed.

### Integration is simple

ATM is placed between the satellite receiver and automation system, and contact closures are connected through ATM's 8x8 GPIO. Your automation system just needs to tell ATM when it's playing a local break, and ATM does the rest. What's more, ATM let's you lock incoming contact closures to incoming audio, then fire outgoing closures when that audio plays out. Local liners and ID's stay synced to the right audio, whether you are delaying the network or not.

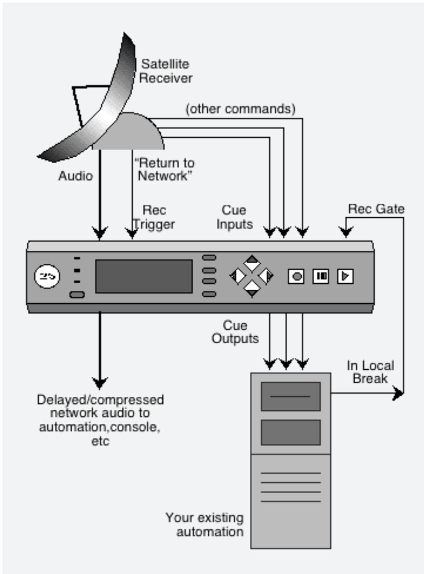
### Here's how it works

When the network sends a "Begin Local Break" command, your automation system sends a signal to close ATM's Record Gate relay, and begins playing local spots.
The automation system keeps the Record Gate relay closed as long as local spots are playing, even if there are more spots than would normally fit in the network break. When the network sends a "Return to Network" command, it activates the Record Trigger and ATM starts recording network programming.

When the local break is over, your automation system sends the command to release the Record Gate. This puts ATM into play mode, seamlessly joining the network and sounding as if they had given you a longer break.

Any additional network cues that would normally go to the automation system can be routed through ATM. Network cue output relays feed ATM's GPIO inputs, then get assigned to numbered Cue flags. ATM's GPIO outputs feed these matching Cue flags to the automation system's inputs. By running these relay closures through ATM, network cues are kept in perfect sync against the audio, even when ATM is delaying or time compressing the programming.

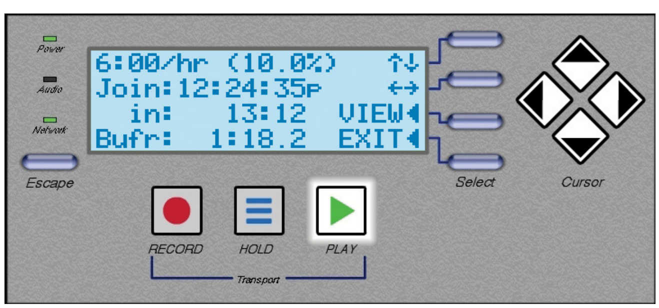
# IN DEPTH




## ATM Web Interface

In addition to controlling Audio Time Manager using RS-232 or GPIO (contract closures), ATM comes with a built-in, password-protected web server, allowing you to remote control your unit across a local or wide area network. An Adobe Flash-based application replicates ATM's front panel on your web browser, so every button and display is present and functions just like the real front panel. Through careful client-server communications management, round-trip latency is almost imperceptible, creating a seamless user experience.

By placing ATM on your facility's router, the browser interface allows you to use Audio Time Manager from any studio in your facility. You can even control ATM from multiple computers. Just open a web browser interface on each, and anything you do on one computer will be reflected on the others, as well as on ATM's physical front panel.



# SPECIFICATIONS

## Power Input

ATM comes with a standard IEC C14 power connector.

## Network

ATM connects to a standard 10/100Base-T network connection. This port is used for Axia Livewire™, synchronization to a network time server, and secure remote control via a web browser. If connected to the Internet, it should be behind a hardware firewall.

## RS-232

This DB9 jack is used for serial remote control, and configured as a computer DTE port similar to the Comm 1 jack on most PCs. ATM communicates at 9600 baud, no parity, 8 bits, 1 stop bit, no flow control (usually referred to as "9600N81"). If you want to control it from a computer or dumb terminal, you'll need a null modem or a cable that reverses pins 2 and 3, pins 4 and 6, and pins 7 and 8. A simple command language is documented in the ATM manual.

## Analog Inputs and Outputs

Stereo inputs are electronically balanced XLR females, pin 2 hot, with a load of 20kΩ: this makes it compatible with all modern electronically-balanced outputs. If fed from a transformer-balanced output, we recommend bridging a 680Ω resistor between pins 2 and 3 recommended. Outputs are electronically balanced XLR males, pin 2 hot, designed to feed a load of 600Ω or greater. Input and output sensitivity default levels can be set from the front panel, and can range between +20dBu and -10dBu.

## Digital Inputs and Outputs

When set to AES/EBU via the configuration menu, this input conforms to IEC 958 Professional (5v p-p, 110Ω balanced) on XLR connectors. When set to s/pdif, the voltage and impedance switches to IEC 958 Consumer (.5v p-p, 75Ω unbalanced): connect signal to pin 2 and shield to pins 1 and 3. Digital output (selectable AES or s/pdif) is always active, regardless of whether you are using analog or digital inputs. ATM will lock to any valid 32 kHz, 44.1 kHz, or 48 kHz signal at the digital input connector, even if you have selected analog for the input. In that case, the digital input controls ATM's internal sample rate. If ATM is not connected to a digital input, it uses its own high-reliability 44.1 kHz sample clock

# SPECIFICATIONS

## GPIO

Eight parallel control inputs and eight parallel control outputs appear on a DB-25 connector. Input and output functions are assigned through a configuration menu on the front panel. Inputs and outputs are opto-isolated for easy interface to other equipment. A +5v supply and ground are also brought out to the DB-25 for simple remote controls using pushbuttons and LED status readouts. The +5v supply can carry 200 mA, more than adequate for 8 LEDs and 8 logic inputs. It is protected by an internal, self-resetting thermal circuit breaker.

## Studio installation

ATM can be inserted anywhere in an analog or digital signal path where you want to control program timing and duration. But for maximum flexibility and efficiency, we recommend using a routing switcher (in many cases, this router already exists in your facility). Have every source you'd want to time-control (such as remotes and network feeds) connected to an input, and at least one output for ATM and one for monitoring. Connect ATM's output directly to a console input channel, or back through another router input.

## Detailed Specifications

### Audio

S/N ≥ 84 dBA with 10 dB headroom (≥94 dB dynamic range); THD @1 kHz < .01%; IMD (IHF) < .01%; Frequency response ± 0.5 dB, 20 Hz – 20 kHz, measured analog input to analog output.

### Dimensions

2RU (rack units); 19" x 12.5" x 3.5" (480 x 320 x 85mm)

### Weight

7.5 lbs (3.5kg)

### Power

100-240 VAC, 50/60 Hz; typical consumption 32 VA

# PRECISION DELAY

## Your Station... In Sync and On Time


## OVERVIEW

- Time-shift and replay programs across multiple time zones
- Keep HD Radio and analog signals in sync using mod monitor data
- Preserves PPM watermarks during delay builds and exits
- Sync transmission booster/repeater networks
- Delayed IP data, serial streams and GPIO sync'd to audio

# FEATURES

## Saving Time, Managing Time

For nearly a decade, 25-Seven Systems has been helping you solve your station's time management problems. Now we've got something for your toughest challenges. Precision Delay, our fourth specialized product, addresses applications such as drift between analog and HD Radio transmission signals and broadcast repeater synchronization.

Precision Delay offers sample-accurate delay times adjustable in fractions of a second; seamless PPM-protecting builds and exits; synchronized data streams; network accessible control; and 25-Seven quality and support.

## Watermark Friendly

Protecting the integrity of PPM codes during delay builds and exits presents special challenges. Precision Delay's unique Watermark Safe Mode helps accommodate the time-based structure of PPM encoding. Our algorithms never alter pitch, so unlike other time manipulation processes, they never undermine the critical frequencies upon which PPM depends.

## Small Delays: Keep Boosters In Sync

Proper time alignment is critical to keeping main signals in sync with boosters or other transmitters relaying on the same frequency. Precision Delay lets you adjust delays by increments as small as a single sample.

## Large Delays: Shift Across Time Zones

For facilities that need to delay content by several minutes to as much as four hours, Precision Delay provides a flexible solution with no spinning hard drives and no complicated programming. With "set and run" simplicity and solid-state reliability.

## Delay Data & GPIO

Precision delay supports up to 3 independent data delays. Serial data over IP or RS-232 such as "now playing" metadata can be delayed in sync with audio — even when delay time is in transition. Contact closures can also be delayed on input so that they trigger against the appropriate audio on output.

## AES, Balanced Analog and Livewire IP Audio

AES digital, balanced analog and Livewire IP audio are standard. Whether you already have an IP audio system or you want to keep your plant IP-capable, Precision Delay matches your signal path today and for years to come.

# FEATURES

## Control Precision Delay from Anywhere

Precision Delay offers complete configuration and control over a LAN or WAN using a common web browser. Navigating though parameters is a breeze with our internal password-protected web server. The network interface also lets you remotely install software updates. Whether your unit is located in your main equipment room or at the transmitter, control is probably right where you're sitting now.

# IN DEPTH

## HD Radio – Shift Happens

Precision Delay syncs HD Radio and analog using mod monitor data A highly placed New York City Radio engineer sums up the dilemna. "HD Radio delay is around eight seconds, but there is no precise match among devices. For any number of reasons, things drift and just a few frames is enough to be detectable, primarily on voice."

With more and more vehicles equipped with HD Radio receivers, stations can't afford confusing listener experiences due to blending out-of-sync analog and HD Radio signals.

Precision Delay lets you precisely set offset measurements by querying and retrieving them over IP from your BELAR FMHD1 or AUDEMAT Golden Eagle modulation monitor.

## Take Precision Delay Out to the Ballgame

When sports fans listen to radio play-by-play at the stadium, they may not know if they are in the right ballpark when HD Radio diversity delay is running. Getting your station into "ball-game mode" means switching the HD Radio signal on and off without annoying listeners or impacting ratings. Precision Delay lets you smoothly build in and out of delay.

## Only Precision Delay has

- Delay range from 10 ms to four hours
- 10 μs resolution
- Watermark Safe Mode to help preserve PPM data
- Inaudible ramping in and out of delay

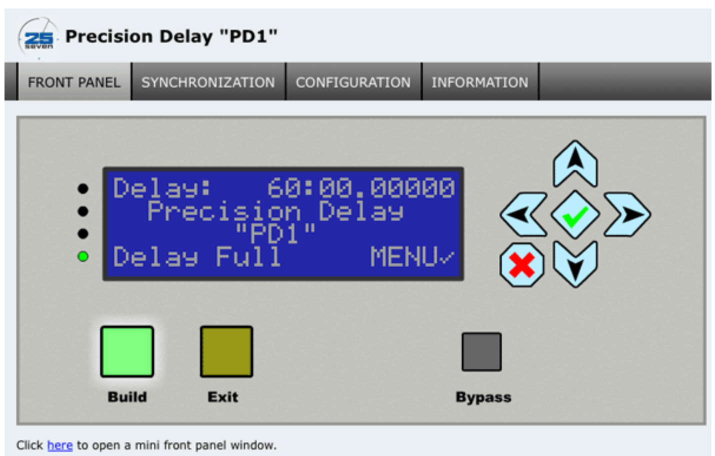
# IN DEPTH

## Precision Delay in your station

- Delay IP data, serial streams and GPIO sync'd to audio
- Connect to AES digital, balanced analog and Livewire IP audio
- Control from anywhere with simple web interface

## Web Interface

In addition to controlling Precision Delay using GPIO (contract closures), PD comes with a built-in, password-protected web server, allowing you to remote control your unit across a local or wide area network. The server gives you five separate pages for complete and convenient control over your PD.




## Front Panel

An Adobe Flash-based application replicates PD's front panel on your web browser, so every button and display is present and functions just like the real front panel. Through careful client-server communications management, round-trip latency is almost imperceptible, creating a seamless user experience. You can even control PD from multiple computers. Just open a web browser interface on each, and anything you do on one computer will be reflected on the others, as well as on PD's physical front panel.

# IN DEPTH

## Configuration

Tired of learning hieroglyphics just to configure a delay? Navigating though "set and forget" parameters is a breeze with the PD's Configuration page. You'll find obvious control with all your settings on one simple screen, so you don't waste time entering data though an ill-suited LCD interface.

# SPECIFICATIONS

## Power Input

Precision Delay comes with a standard IEC C14 power connector.

## Network

Precision Delay connects to a standard 10/100Base-T network connection. This port is used for Axia Livewire™, synchronization to a network time server, and secure remote control via a web browser. If connected to the Internet, it should be behind a hardware firewall.

## Analog Inputs and Outputs

Stereo inputs are electronically balanced XLR females, pin 2 hot, with a load of 20kΩ: this makes it compatible with all modern electronically-balanced outputs. If fed from a transformer-balanced output, we recommend bridging a 680Ω resistor between pins 2 and 3. Outputs are electronically balanced XLR males, pin 2 hot, designed to feed a load of 600Ω or greater. Input and output sensitivity default levels can be set from the front panel, and can range between +20dBu and -10dBu.

## Digital Inputs and Outputs

When set to AES/EBU via the configuration menu, this input conforms to IEC 958 Professional (5v p-p, 110Ω balanced) on XLR connectors. When set to s/pdif, the voltage and impedance switches to IEC 958 Consumer (.5v p-p, 75Ω unbalanced): connect signal to pin 2 and shield to pins 1 and 3. Digital output (selectable AES or s/pdif) is always active, regardless of whether you are using analog or digital inputs. PD will lock to any valid 32 kHz, 44.1 kHz, or 48 kHz signal at the digital input connector, even if you have selected analog for the input. In that case, the digital input controls Precision Delay s internal sample rate. If Precision Delay is not connected to a digital input, it uses its own high-reliability 44.1 kHz sample clock.

# SPECIFICATIONS

## Axia Livewire Inputs and Outputs

Precision Delay supports Livewire, with audio connections exclusively via the network. Precision Delay also supports Livewire-based GPIO.

## GPIO

Eight parallel control inputs and eight parallel control outputs appear on a DB-25 connector. Input and output functions are assigned through a configuration menu on the front panel. Inputs and outputs are opto-isolated for easy interface to other equipment. A +5v supply and ground are also brought out to the DB-25 for simple remote controls using pushbuttons and LED status readouts. The +5v supply can carry 200 mA, more than adequate for 8 LEDs and 8 logic inputs. It is protected by an internal, self-resetting thermal circuit breaker.

## Detailed Specifications

### Audio

S/N ≥ 84 dBA with 10 dB headroom (≥94 dB dynamic range); THD @1 kHz < .01%; IMD (IHF) < .01%; Frequency response ± 0.5 dB, 20 Hz – 20 kHz, measured analog input to analog output.

### Dimensions

1RU (rack unit); 19" W (with rack ears) x 12" D x 1.75" H (483 x 305 x 44mm)

### Delay range

10 ms to 4 hours; adjustable in 10 μs increments

### Power

100-240 VAC, 50/60 Hz; typical consumption 32 VA.

# ELEMENT

## World's Most Popular IP-Audio Console

## OVERVIEW

The Axia Element is the world's most popular IP-Audio mixing console, in use at thousands of broadcast facilities every single day. Element is a modular console. Frames are available in sizes from 4 to 28 positions, with support for up to 40 faders in multiple linked frames. The Element control surface works with the Axia PowerStation and StudioEngine DSP mixing engines, and connects to the Axia network with a single CAT-6 Ethernet cable. The networked nature of Element (and all Axia mixing consoles) allows sharing of local audio resources and associated GPIO control across multiple studios.

Element features four stereo Program buses, four Send buses, two Return buses and a number of VMix (Virtual Mixer) channels which allow combining up to 5 audio sources for presentation on a single console fader. A variety of module types provide control of mic/line inputs, telephones and other devices.

# OVERVIEW

Enhanced, integrated features for phones and codecs include auto-assigned, auto-generated mix-minus on each channel, easy individual or group talkback for remote talent cueing, one-button off-air phone record mode, and optional integrated Telco line switching. Show Profiles allow console "snapshots" with different preferences, layouts and defaults to be loaded instantly, customizing the board to each show or talent if desired.

Additionally, Element includes digital EQ which may be applied individually to all audio sources, dynamic microphone processing from Omnia Audio for all mic sources, and many other advanced features.

# FEATURES

- From 4 to 40 fader channels, each with instant, unlimited access to any source. You can assign any type of source to any channel.
- Four main stereo outputs (Program-1 through Program- 4), plus four stereo Aux sends and two Aux returns.
- 10-character alpha-numeric displays above each fader channel always show the current selected source.
- Each channel is equipped with a Status Symbol™ display which provides talkback, mix-minus, and other source-related communication information.
- Every channel has a stereo Preview ("cue") function, with a unique latching interlock system for fast, intuitive operation. Multiple channels may be assigned to Preview simultaneously.
- Reconfigurable monitor section with reassignable controls let operators instantly change monitored sources "on-the-fly."
- Flexible, intuitive talkback system lets board ops talk to hosts, studio guests, external feeds — any source with an associated backfeed.
- Software control of options such as EQ, mic dynamics, aux sends and returns, pan and balance and other features delivers maximum flexibility without panel clutter or intimidating controls.
- Built-in Omnia dynamics processing lets operators combine compression, de-essing and expansion with EQ to "sweeten" microphone sources.
- A unique Record Mode enables one-button setup of record mixes for phone bits or off-air interviews.

# FEATURES

- Consolidated user display conveys meter, clock, timer and monitor source information at a single glance. Use any external VGA monitor you choose, from a 12” LCD to a DLP wall projector!
- Precision timer and clock functions, including an event timer that can be triggered by pre-defined sources, a countdown timer with last-minute alerting and a time-of-day clock that can be synchronized to network time using NTP.
- Show Profiles set-save-recall feature allows users to instantly recall a customized personal profile, or a profile tailored to specific show types. Up to 99 Show Profiles can be saved for interview shows, music-intensive programming, call-in talk shows, etc.
- Console functions can be accessed remotely for configuration, management and diagnostic purposes using any standard Web browser.
- Built-in 5.1 discrete mixing capabilities for production use features synchronized stereo upmix/ downmix capabilities.
- Optional Telos phone control module provides direct, on-the-console line switching control of any Telos multi-line broadcast phone system.
- Numeric keypad (with # and * keys) lets operators quickly place calls with phone systems or codecs attached to the Axia network.
- Completely automatic mix-minus generation for every Phone caller or remote Codec source.
- Built-in control keys for external profanity delay unit integrate via Livewire with 25-Seven Program Delay Manager, or can be slaved to any other PDU using standard GPIO closures.
- No audio passes directly through Element — all mixing and processing is performed by the StudioEngine or PowerStation mixing engines. Console connects to the Axia network using just one cable.
- Long-life conductive-plastic faders with side-loading actuation defy dirt, grit and dust.
- Aircraft-grade switches with LED lighting have been tested to withstand millions of operations.
- Modules are available in choice of Bronze or Silver, with high-impact Lexan overlays. Custom-designed fader and switch surrounds prevent cracking, chipping or peeling; markings can’t fade or rub off — ever.
- Can be directly remote-controlled using Axia SoftSurface software for Windows.

# IN DEPTH

## The Choice of Connected Broadcasters Everywhere.

Axia was launched by Telos in 2003 to make digital mixing consoles. But we had a unique vision: Axia consoles would be integrated with the routing switcher, and networked to share resources and capabilities throughout the studio complex. Using this intelligent network of studio devices, talent would benefit from consoles more powerful and easier to use than ever. 10 years and more than 5,000 studios later, broadcasters have made Axia consoles the most popular networked consoles in the world, powering studios around the globe for the world's most demanding broadcasters.

So, why have broadcasters made Element the world's most popular IP-Audio console? Simple: when our team of obsessive console engineers first began designing Element, they asked broadcast professionals to describe their ideal mixing console. "Powerful," they said, "but easy to use, with the capabilities of a full-up production board."

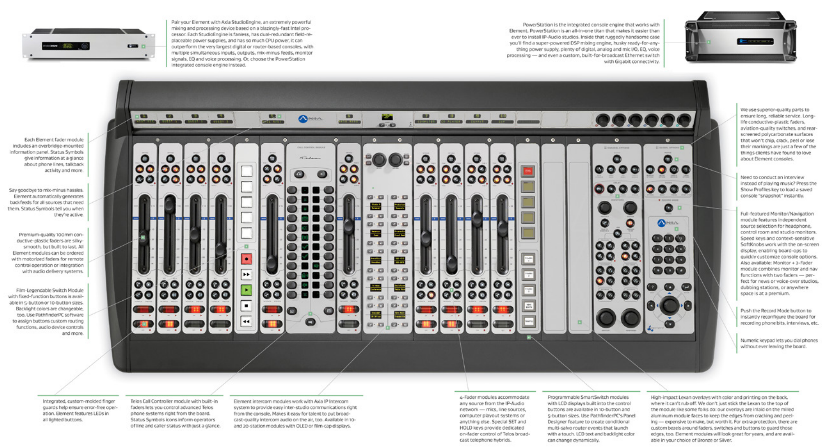
So our engineers went to the lab and blended the best ideas from old-school analog consoles with innovative new technology to produce bullet-proof boards that can actually make shows run smoother and sound better. The result: Axia's modular Element 2.0 mixing console. Scalable from 4 to 40 faders, Element is the ultra-reliable power source behind more than 4,000 Axia-equipped broadcast studios around the world.
Like all Axia broadcast equipment, Element consoles connect using standards-based Livewire IP-Audio networking technology, invented by Telos. Using Livewire, broadcasters can easily network studios, consoles and audio equipment using standard Ethernet. Livewire can carry hundreds of channels of real-time, uncompressed audio plus synchronized control logic and program-associated data on a single CAT-6 cable, reducing cost, complexity and studio construction time.

Because Axia networks are intelligent Ethernet-based routing systems, machine logic always follows source audio. When your operator loads a source to any fader, in any studio, that fader's controls are immediately communicating with the source device. Thanks to this scalable network technology, integrated router control is a standard feature of every Element.

Talent raves about Element Show Profiles. Each console contains up to 99 storage locations that operators can use to set, save and recall their favorite settings with the push of a button — audio sources, fader assignments, monitor settings and more. Show Profiles can also contain talent's personalized mic processing and voice EQ settings that load every time they're on the air — and, in case a jock gets himself into unfamiliar territory, Element provides a convenient one-key "panic button" that returns a Show Profile to its default state instantly.

# IN DEPTH

There's plenty of power under the surface, too. To make sure you have plenty of mixing capacity, Element features 4 Program buses, plus 4 Aux sends and 2 Aux returns, along with 16 five-channel "Virtual Mixers" that let you mix multiple audio inputs using virtual faders. More built-in convenience: Every voice channel has studio-grade Omnia audio processing, including mic compression, de-essing and gating, plus three-band parametric EQ, which can be set and saved with each Show Profile. Need a headphone processor for your talent? Element provides that, too, with built-in headphone processing to save the cost of a separate side-chain.



You'll also find fully-automatic mix-minuses; one for each fader if needed. Mix-minus settings are saved for each audio source, so that sources, backfeed and machine logic all load at once. And every fader has a "Talkback" key to communicate with phone callers, remote talent or other studios using the console mic; use them singly, or in multiple to communicate with entire groups of locations at once.

Axia's Livewire Ethernet backbone makes it easy to integrate and control all kinds of different devices on the same network. And Element puts those controls right on the console, where they're most useful. For instance: phone hybrid modules with dedicated faders control Telos talkshow systems. There's a dial pad, too, so talent can dial, answer, screen and drop calls without ever taking their eyes – or attention –off the console. Which translates into smoother, more error-free on-air phone segments. Axia's IP Intercom system connects to the Livewire network too, and drop-in Intercom modules for your Element

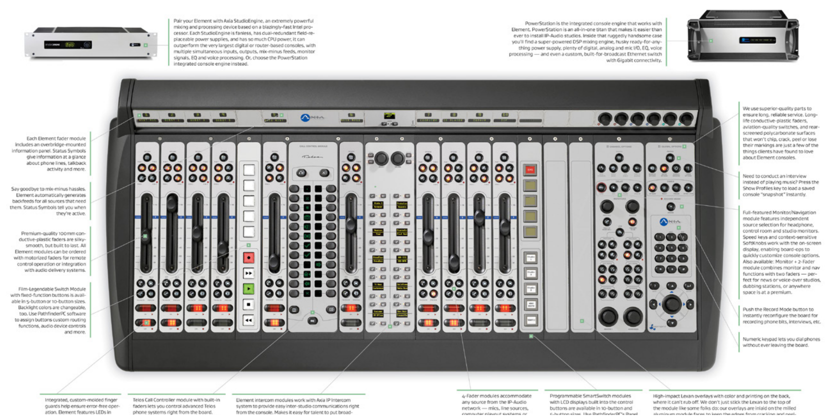
# IN DEPTH

place multi-station intercom controls right at jocks' fingertips. Which means that talent can now easily take broadcast-quality intercom audio directly to air, with only a button-press or two.

As with all Axia consoles, engineers can administer Element remotely. A password-protected Web server lets you examine the state of the console and make configuration changes. With our new SoftSurface companion software, you can even take direct remote control of Element from your office, home, or anywhere there's an Internet connection.

There's more to building a great board than just features, of course. Consoles have to perform flawlessly 24/7, 365 days-a year, for years at a time. So Element is fabricated from thick, machined aluminum extrusions — rigid and RF-immune. Power supplies are hardened for reliable, continuous uptime, and fanless for silent in-studio operation. Modules can be hot-swapped. Silky-smooth conductive-plastic faders actuate from the side, so dirt can't get in. High-end optical rotary encoders mean no wipers to get dirty or wear out. And our avionics-grade switches, with LED lighting, have been tested to withstand more than five million operations.

Some folks have said that Element consoles are over-engineered. To which we say, "thank you"! Not everyone appreciates this kind of attention to detail, but if you're one who seeks out and appreciates excellence wherever you may find it... Element just may be the answer you've been looking for.



Your station is customized to your listeners. Shouldn't your console be customized to your talent? Mix and match a variety of Element module types with enhanced features to suit your station's operational needs. Like integrated controls for phones, codecs and intercoms, EQ modules designed to speed off-air production, even motorized faders for remote control or integration with your delivery system. Choice is good!

# IN DEPTH

## 4-Fader Module

The 4-Fader module is the heart of any Element. Use it for any source: line, mic, hybrid, phone or codec source. Comes in standard and motorized-fader versions for use with automation systems or other moving-fader applications.

## Monitor Modules

The other basic module every console needs is the Monitor module. Element offers two types.

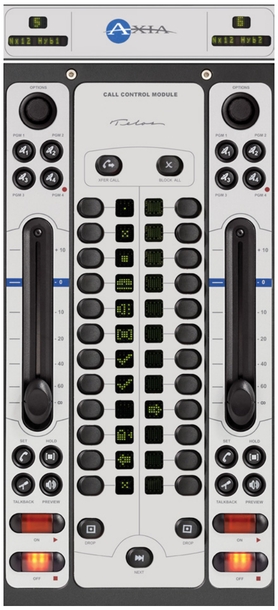
The Expert Monitor/Navigation module shown here has extended monitor, headphone and preview controls, plus a numeric entry/dialpad that can be used with Element phone modules, plus convenient profanity delay controls that can be linked to your delay unit.

# IN DEPTH

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

# IN DEPTH

## Call Controller Module



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

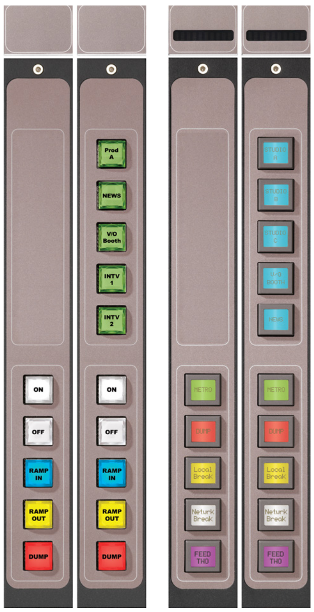
# IN DEPTH

## Production Module

Designed specifically for production professionals, the Production module gives your talent fast access to frequently-used options, such as Send bus levels, EQ settings, and panning tools for each channel. LED indications for all rotary controls give immediate information when a fader strip's Options knob is pressed.

# IN DEPTH

## Switch Modules




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

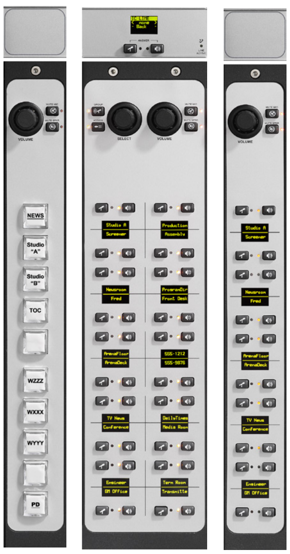
Economical Film-Cap switch modules (shown on left) with 5 or 10 buttons are perfect for giving talent access to often-used machine-control or GPIO-triggered routing commands. LED button backlights can be individually changed to any of 8 colors.

Need to give operators more complex control of routing functions? 5 and 10-button SmartSwitch modules (shown on right) feature dynamic, backlit LCD displays. Button functions, colors and even text can be programmed to change in response to user input using Axia PathfinderPC software. Construct custom routing salvos, cascading machine-logic command sets, or other complex routing operations.

# IN DEPTH

## IP Intercom Modules

Element consoles come equipped with a sophisticated Talkback system that allows board ops to communicate directly with remote talent via individual Talkback channels. But when larger facilities require even more powerful communication capabilities, these 10 and 20-station intercom modules, part of the Axia IP Intercom system, put broadcast intercom controls right in the console. Station presets and GPIO functions for both types of modules are programmed using any standard Web browser.



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

## Mixing Engines

Element consoles give you choices at every turn, and mixing engine platforms are no exception. You can build your Axia network using a la carte components – an Element control surface with a powerful Linux-based StudioEngine and separate xNode audio interfaces – or you can choose the PowerStation integrated console engine, an all-in-one powerhouse with audio I/O, DSP mixing engine and integrated zero-configuration network switch.

# IN DEPTH

## StudioEngine


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

## PowerStation

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

# SPECIFICATIONS

## Microphone Preamplifiers

- Source Impedance: 150 Ohms
- Input Impedance: 4 k Ohms minimum, balanced
- Nominal Level Range: Adjustable, -75 dBu to -20 dBu
- Input Headroom: >20 dB above nominal input
- Output Level: +4 dBu, nominal

## Analog Line Inputs

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

## Analog Line Outputs

- Output Source Impedance: <50 Ohms balanced
- Output Load Impedance: 600 Ohms, minimum
- Nominal Output Level: +4 dBu
- Maximum Output Level: +24 dBu

## Digital Audio Inputs and Outputs

- Reference Level: +4 dBu (-20 dB FSD)
- Impedance: 110 Ohms, balanced (XLR)
- Signal Format: AES-3 (AES/EBU)
- AES-3 Input Compliance: 24-bit with selectable sample rate conversion,
- 32 kHz to 96kHz input sample rate capable.
- AES-3 Output Compliance: 24-bit
- Digital Reference: Internal (network timebase) or external reference 48 kHz,
- +/- 2 ppm

# SPECIFICATIONS

- Internal Sampling Rate: 48 kHz
- Output Sample Rate: 44.1 kHz or 48 kHz
- A/D Conversions: 24-bit, Delta-Sigma, 256x oversampling
- D/A Conversions: 24-bit, Delta-Sigma, 256x oversampling
- Latency <3 ms, mic in to monitor out, including network and processor loop

## Frequency Response

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

## Dynamic Range

- Analog Input to Analog Output: 102 dB referenced to 0 dBFS,
- 105 dB "A" weighted to 0 dBFS
- Analog Input to Digital Output: 105 dB referenced to 0 dBFS
- Digital Input to Analog Output: 103 dB referenced to 0 dBFS, 106 dB "A" weighted
- Digital Input to Digital Output: 138 dB

## Equivalent Input Noise

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

## Total Harmonic Distortion + Noise

- Mic Pre Input to Analog Line Output: <0.005%, 1 kHz, -38 dBu input,
- +18 dBu output
- Analog Input to Analog Output: <0.008%, 1 kHz, +18 dBu input, +18 dBu output
- Digital Input to Digital Output: <0.0003%, 1 kHz, -20 dBFS
- Digital Input to Analog Output: <0.005%, 1 kHz, -6 dBFS input, +18 dBu output

# SPECIFICATIONS

## Crosstalk Isolation, Stereo Separation and CMRR

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

## Audio Processing

### Equalizer

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

### Compressor

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

### Expander/Noise Gate

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

### De-esser

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

## Power Supply AC Input, StudioEngine

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

# SPECIFICATIONS

## Power Supply AC Input, Element Power Supply/GPIO

- Auto-sensing supply, 90VAC to 240VAC, 50 Hz to 60 Hz, IEC receptacle, internal fuse
- Power consumption: 150 Watts

## Power Supply AC Input, PowerStation Aux & Main

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

## Operating Temperatures

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

# SOFTSURFACE

## Software for Axia Element Console

## OVERVIEW

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

# FEATURES

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

# IN DEPTH

## Control at the Click of a Mouse.

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

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

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

# IN DEPTH

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

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

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

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


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

# IN DEPTH

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

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

# IN DEPTH


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


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

# IN DEPTH


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


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

# IN DEPTH


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

# SPECIFICATIONS

## Hardware Requirements

- A PC with the Microsoft™ Windows XP, Windows Vista, or Windows 7 operating system installed.
- A display screen with a minimum resolution of 1280x1024 pixels.
- A mouse, touch-screen, or other suitable pointing device.
- A 100Base-T LAN connection with a static IP address.
- 20Mb free disk space.

## Audio Processing

### Equalizer

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

# SPECIFICATIONS

## Compressor

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

## Expander/Noise Gate

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

## De-esser

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

# iQ
## The Smarter IP Console

## OVERVIEW

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

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

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

# FEATURES

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

# FEATURES

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

# IN DEPTH

## Easy Installation. Fast Configuration. Intuitive Operation.

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

# IN DEPTH

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

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

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

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

iQ saves your studio furniture, too. Its desktop design lets you place it atop any solid surface — no templates to decipher or countertops to cut (unless you really want to). Since iQ only requires a single

# IN DEPTH

cable to connect control surface to mixing engine, even cable access holes can be small and unobtrusive. And iQ lets you choose between freestanding or contiguous console designs: you can easily join iQ expansion frames into one unit, or leave them separate to deploy a split-console design.

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

## iQ System Components

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

## iQ Main Frame

The heart of your iQ console; can be installed as a standalone console or connected to an Axia studio network. Has three dedicated stereo Program buses, plus a stereo

# IN DEPTH

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

## 8-Fader Expansion Frame

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

# IN DEPTH

## 6-Fader Expansion Frame with User Keys

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

## 6-Fader Telco Expansion Frame

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

# IN DEPTH

## iQ6 6-Line Telco Gateway

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

## QOR.32 Integrated Console Engine

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

# IN DEPTH

## QOR.32 Backup power supply

The QOR.32 Backup power supply is a hardened, auto-switching power supply that is perfect for facilities where redundant power backup is required. Connects to your QOR.32 console engine in less than a minute using a single cable that supplies failsafe backup power with automatic switchover (should the need ever arise).

# SPECIFICATIONS

## QOR.32 Connections

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

# SPECIFICATIONS

## Microphone Preamplifiers

- Source Impedance: 150 Ohms
- Input Impedance: 4 k Ohms minimum, balanced
- Nominal Level Range: Adjustable, -75 dBu to -20 dBu
- Input Headroom: >20 dB above nominal input
- Output Level: +4 dBu, nominal

## Analog Line Inputs

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

## Analog Line Outputs

- Output Source Impedance: <50 Ohms balanced
- Output Load Impedance: 600 Ohms, minimum
- Nominal Output Level: +4 dBu
- Maximum Output Level: +24 dBu

## Digital Audio Inputs And Outputs

- Reference Level: +4 dBu (-20 dB FSD)
- Impedance: 110 Ohm, balanced (XLR)
- Signal Format: AES-3 (AES/EBU)
- AES-3 Input Compliance: 24-bit with selectable sample rate conversion, 20 kHz to 216kHz input sample rate capable.
- AES-3 Output Compliance: 24-bit
- Digital Reference: Internal (network timebase) or external reference 48 kHz, +/- 2 ppm
- Internal Sampling Rate: 48 kHz
- Output Sample Rate: 48 kHz

# SPECIFICATIONS

- A/D Conversions: 24-bit, Delta-Sigma, 256x oversampling
- D/A Conversions: 24-bit, Delta-Sigma, 256x oversampling
- Latency <3 ms, mic in to monitor out, including network and processor loop

## Frequency Response

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

## Dynamic Range

- Analog Input to Analog Output: 102 dB referenced to 0 dBFS, 105 dB "A" weighted to 0 dBFS
- Analog Input to Digital Output: 105 dB referenced to 0 dBFS
- Digital Input to Analog Output: 103 dB referenced to 0 dBFS, 106 dB "A" weighted
- Digital Input to Digital Output: 125 dB

## Equivalent Input Noise

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

## Total Harmonic Distortion + Noise

- Mic Pre Input to Analog Line Output: <0.005%, 1 kHz, -38 dBu input, +18 dBu output
- Analog Input to Analog Output: <0.008%, 1 kHz, +18 dBu input, +18 dBu output
- Digital Input to Digital Output: <0.0003%, 1 kHz, -20 dBFS
- Digital Input to Analog Output: <0.005%, 1 kHz, -6 dBFS input, +18 dBu output

## Crosstalk Isolation, Stereo Separation And CMRR

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

# SPECIFICATIONS

## Audio Processing

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

## Power Supply AC Input, QOR.32 With iQ Console

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

## Operating Temperatures

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

## Dimensions

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

# RADIUS

## Throw Your Budget A Curve

## OVERVIEW

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

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

# FEATURES

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

# FEATURES

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

# IN DEPTH

## Spend Less. Get More.

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

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

# IN DEPTH

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

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

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


There are also OLED displays on every fader that provide source assignments, pan & balance settings, fader options and more — which means no additional computer monitors or mice to clutter up your

# IN DEPTH

studio. The display can also work with the Soft Keys just below to trigger GPIO events, step automation events, and adjust source input options.

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

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

# SPECIFICATIONS

## QOR.16 Connections

- Microphone Inputs: 2x balanced XLR-F, with selectable Phantom power
- Analog Inputs: 8x RJ-45, StudioHub+ standard.
- Analog Outputs: 4x RJ-45, StudioHub+ standard.
- AES/EBU Inputs: 1x RJ-45, StudioHub+ standard.
- AES/EBU Outputs: 1x RJ-45, StudioHub+ standard.

# SPECIFICATIONS

- GPIO: 4x DB-15
- Livewire:
    - 4x 100Base-T with PoE, RJ-45
    - 2x 100Base-T, RJ-45
    - 2x 1000Base-T, RJ-45
    - 2x Gigabit, SFP (Small Form Pluggable)
- Console Frame Connections: 1x, 6-pin "latch and lock" style
- Accessory Connections: 1x, 6-pin "latch and lock" style

## Microphone Preamplifiers

- Source Impedance: 150 Ohms
- Input Impedance: 4 k Ohms minimum, balanced
- Nominal Level Range: Adjustable, -75 dBu to -20 dBu
- Input Headroom: >20 dB above nominal input
- Output Level: +4 dBu, nominal

## Analog Line Inputs

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

## Analog Line Outputs

- Output Source Impedance: <50 Ohms balanced
- Output Load Impedance: 600 Ohms, minimum
- Nominal Output Level: +4 dBu
- Maximum Output Level: +24 dBu

# SPECIFICATIONS

## Digital Audio Inputs And Outputs

- Reference Level: +4 dBu (-20 dB FSD)
- Impedance: 110 Ohm, balanced (XLR)
- Signal Format: AES-3 (AES/EBU)
- AES-3 Input Compliance: 24-bit with selectable sample rate conversion, 20 kHz to 216kHz input sample rate capable.
- AES-3 Output Compliance: 24-bit
- Digital Reference: Internal (network timebase) or external reference 48 kHz, +/- 2 ppm
- Internal Sampling Rate: 48 kHz
- Output Sample Rate: 48 kHz
- A/D Conversions: 24-bit, Delta-Sigma, 256x oversampling
- D/A Conversions: 24-bit, Delta-Sigma, 256x oversampling
- Latency <3 ms, mic in to monitor out, including network and processor loop

## Frequency Response

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

## Dynamic Range

- Analog Input to Analog Output: 102 dB referenced to 0 dBFS, 105 dB “A” weighted to 0 dBFS
- Analog Input to Digital Output: 105 dB referenced to 0 dBFS
- Digital Input to Analog Output: 103 dB referenced to 0 dBFS, 106 dB “A” weighted
- Digital Input to Digital Output: 125 dB

## Equivalent Input Noise

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

# SPECIFICATIONS

## Total Harmonic Distortion + Noise

- Mic Pre Input to Analog Line Output: <0.005%, 1 kHz, -38 dBu input, +18 dBu output
- Analog Input to Analog Output: <0.008%, 1 kHz, +18 dBu input, +18 dBu output
- Digital Input to Digital Output: <0.0003%, 1 kHz, -20 dBFS
- Digital Input to Analog Output: <0.005%, 1 kHz, -6 dBFS input, +18 dBu output

## Crosstalk Isolation, Stereo Separation And CMRR

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

## Audio Processing

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

## Power Supply AC Input, QOR.16 With Radius Console

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

## Operating Temperatures

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

## Dimensions

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

# RAQ

## Rack-mount IP Console

## OVERVIEW

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

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

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

# FEATURES

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

# FEATURES

switch setup required. Supports Simple Networking, allowing up to 4 QOR engines to be daisy-chained without the need for a separate core switch.

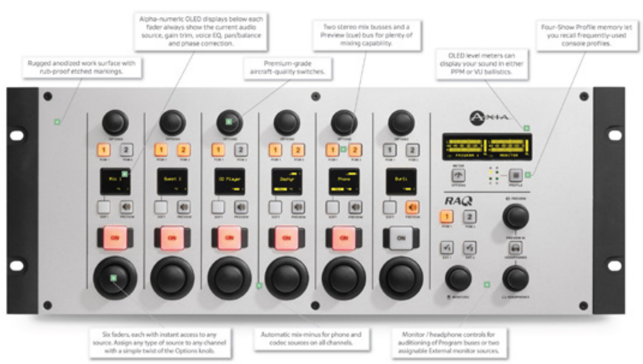
- Each QOR.16 can support two connected RAQ or DESQ consoles, or one of each.
- Fan-free, convection-cooled power supply for noiseless in-studio operation.

# IN DEPTH

## A big console for small spaces.

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

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




RAQ is a six-channel mixer over-engineered the Axia way, with super-duty rotary faders, aluminum front-panel, high-resolution OLED displays for channel assignment and metering, heavy-duty switches

# IN DEPTH

with LED lighting, and four Show Profile snapshot locations you can use to store and instantly recall favorite console configurations. One touch, and presto! Talent's favorite sources are loaded, monitor source configured, and bus assignments made.

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

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

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

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

# SPECIFICATIONS

## QOR.16 Connections

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

## Microphone Preamplifiers

- Source Impedance: 150 Ohms
- Input Impedance: 4 k Ohms minimum, balanced
- Nominal Level Range: Adjustable, -75 dBu to -20 dBu
- Input Headroom: >20 dB above nominal input
- Output Level: +4 dBu, nominal

## Analog Line Inputs

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

## Analog Line Outputs

- Output Source Impedance: <50 Ohms balanced

# SPECIFICATIONS

- Output Load Impedance: 600 Ohms, minimum
- Nominal Output Level: +4 dBu
- Maximum Output Level: +24 dBu

## Digital Audio Inputs And Outputs

- Reference Level: +4 dBu (-20 dB FSD)
- Impedance: 110 Ohm, balanced (XLR)
- Signal Format: AES-3 (AES/EBU)
- AES-3 Input Compliance: 24-bit with selectable sample rate conversion, 20 kHz to 216kHz input sample rate capable.
- AES-3 Output Compliance: 24-bit
- Digital Reference: Internal (network timebase) or external reference 48 kHz, +/- 2 ppm
- Internal Sampling Rate: 48 kHz
- Output Sample Rate: 48 kHz
- A/D Conversions: 24-bit, Delta-Sigma, 256x oversampling
- D/A Conversions: 24-bit, Delta-Sigma, 256x oversampling
- Latency <3 ms, mic in to monitor out, including network and processor loop

## Frequency Response

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

## Dynamic Range

- Analog Input to Analog Output: 102 dB referenced to 0 dBFS, 105 dB “A” weighted to 0 dBFS
- Analog Input to Digital Output: 105 dB referenced to 0 dBFS
- Digital Input to Analog Output: 103 dB referenced to 0 dBFS, 106 dB “A” weighted
- Digital Input to Digital Output: 125 dB

## Equivalent Input Noise

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

# SPECIFICATIONS

## Total Harmonic Distortion + Noise

- Mic Pre Input to Analog Line Output: <0.005%, 1 kHz, -38 dBu input, +18 dBu output
- Analog Input to Analog Output: <0.008%, 1 kHz, +18 dBu input, +18 dBu output
- Digital Input to Digital Output: <0.0003%, 1 kHz, -20 dBFS
- Digital Input to Analog Output: <0.005%, 1 kHz, -6 dBFS input, +18 dBu output

## Crosstalk Isolation, Stereo Separation And CMRR

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

## Audio Processing

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

## Power Supply AC Input, QOR.16 With RAQ Console

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

## Operating Temperatures

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

## Dimensions

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

# DESQ

## Compact Desktop IP Console

## OVERVIEW

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

There's also an OLED time-of-day clock + timer display, with auto / manual reset option. As with all Axia consoles, aircraft-quality switches feature all-LED lighting; the anodized work surface has rub-proof etched markings that can't rub off. Additional features include automatic mix-minus for phone callers and codec sources, EQ for voice sources, and Show Profile instant recall of up to 4 pre-defined console "snapshots".

# OVERVIEW

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

# FEATURES

- 6 faders, each with instant access to any source. Assign any type of source to any channel with a simple twist of the Options knob.
- Proven surface-and-core architecture separates control from mixing processes. No audio passes directly through DESQ; all mixing and processing is performed in the QOR.16 Integrated Console Engine – so studio "accidents" don't turn into off-air events.
- Two stereo mix buses and a Preview (cue) bus.
- Alpha-numeric OLED displays below each fader always show the current audio source with audio confidence meter, and, when the Options knob is pressed, offer fast adjustment of fader gain trim, EQ, pan and balance and other features without panel clutter or intimidating controls.
- Channel-input confidence meters assure talent of audio presence before taking sources to air.
- Built-in three-band per-source EQ.
- Each fader's context-sensitive Soft key can be used to activate talkback, start delivery system events, or perform other special functions.
- Every channel strip has a stereo Preview ("cue") function, with a unique interlock system for fast cuing of multiple sources.
- Reconfigurable CR monitor section with direct-selection of Program buses and reassignable buttons that allow instant monitoring of external sources.
- Four custom Show Profile "snapshots" can be saved to instantly recall frequently-used console setups – useful to quickly prepare for interview segments, music-intensive programming, call-in talk shows, etc.
- Automatic mix-minuses for phones, remote talent, etc.
- Bright OLED meter display provides responsive, readable VU or PPM metering styles. Switchable display allows metering either Program bus.

# FEATURES

- All functions can be accessed remotely for configuration, management and diagnostic purposes using any standard Web browser.
- Separate OLED clock/timer display features NTP-capable time-of-day clock and event timer that can be manually or automatically reset via source activation.
- Network gateway enables loading networked sources while simultaneously exporting outputs back to the network.
- Easy-to-deploy QOR.16 integrated console engine includes console CPU and power supply, DSP mixing engine, custom Ethernet switch with 6 Livewire ports and 2 Gigabit ports for studio networking, 8 analog inputs and 4 analog outputs, 1 AES input and 1 AES output, 2 Mic inputs with switchable Phantom power, and 4 GPIO ports for machine control. I/O can be expanded using Axia xNodes.
- QOR.16's integrated zero-configuration network switch is custom-designed for broadcasting — no switch setup required. Supports Simple Networking, allowing up to 4 QOR engines to be daisy-chained without the need for a separate core switch.
- Each QOR.16 can support two connected RAQ or DESQ consoles, or one of each.
- Fan-free, convection-cooled power supply for noiseless in-studio operation.

# IN DEPTH

## A big console for small spaces.

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

DESQ is a compact, special-purpose IP console from Axia. It may be compact in stature, but it's big on features and performance. DESQ has "big board" capabilities you won't find in other consoles of this size

# IN DEPTH

— automatic per-fader mix-minus, built-in EQ for voice and codec sources, and the ability to instantly load new local or networked sources to any fader with just the turn of a knob. Which means DESQ easily out-classes mixers with similar form factors — and even ones that take up much more room.


It's built Axia-tough, with a machined-aluminum work surface that takes the rough stuff jocks can dish out. Our familiar 100 mm. conductive-plastic faders feel like silk under the fingertips, and you'll also find the avionics-grade switches with LED lighting that have become an Axia hallmark.

Other features include OLED channel and meter displays, four-source monitor section with two external locations that can be reassigned "on the fly", and an OLED time-of-day clock and event timer. Like its rackmount cousin, RAQ, DESQ also has four Show Profile console snapshot locations, and push-and-turn Options knobs at the top of each fader that give instant access to fader source assignments, pan/balance, and input gain trim.

Despite all these features, DESQ is so cost-effective, broadcasters are coming up with creative, new uses for them. Its big features and small footprint make DESQ the perfect console for interview studios, live performance spaces for on-air broadcast, news and feature production — whatever. Take it on road trip remotes, or to sporting events where multiple mics are required. Or put it in mobile units or ENG kits. Perfect for personal production studios, too.

# IN DEPTH

Audio I/O, GPIO, console CPU, super-duty power supply, and even a network switch are all built into the QOR.16. Just plug in your mics, CD players, codecs, profanity delays, whatever. There are 16 audio I/O ports: two Mic inputs with switchable Phantom power, eight analog inputs and four analog outputs, and one AES/EBU input and output. QOR.16 also has four GPIO logic ports for machine control of studio peripherals, six 100Base-T ports for Livewire devices, and two Gigabit ports with SFP for connection to the outside world. For more I/O, just add Axia xNode interfaces. And you can daisy chain as many as four QOR engines without the need for an external Ethernet switch, making installation even more economical. And here's the kicker: one QOR.16 can power two DESQ mixers — or a DESQ and a RAQ (DESQ's six-fader, rackmount cousin)!

# SPECIFICATIONS

## QOR.16 Connections

- Microphone Inputs: 2x balanced XLR-F, with selectable Phantom power
- Analog Inputs: 8x RJ-45, StudioHub+ standard.
- Analog Outputs: 4x RJ-45, StudioHub+ standard.
- AES/EBU Inputs: 1x RJ-45, StudioHub+ standard.
- AES/EBU Outputs: 1x RJ-45, StudioHub+ standard.
- GPIO: 4x DB-15
- Livewire:
    - 4x 100Base-T with PoE, RJ-45
    - 2x 100Base-T, RJ-45
    - 2x 1000Base-T, RJ-45
    - 2x Gigabit, SFP (Small Form Pluggable)

# SPECIFICATIONS

- Console Frame Connections: 1x, 6-pin "latch and lock" style
- Accessory Connections: 1x, 6-pin "latch and lock" style

## Microphone Preamplifiers

- Source Impedance: 150 Ohms
- Input Impedance: 4 k Ohms minimum, balanced
- Nominal Level Range: Adjustable, -75 dBu to -20 dBu
- Input Headroom: >20 dB above nominal input
- Output Level: +4 dBu, nominal

## Analog Line Inputs

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

## Analog Line Outputs

- Output Source Impedance: <50 Ohms balanced
- Output Load Impedance: 600 Ohms, minimum
- Nominal Output Level: +4 dBu
- Maximum Output Level: +24 dBu

## Digital Audio Inputs And Outputs

- Reference Level: +4 dBu (-20 dB FSD)
- Impedance: 110 Ohm, balanced (XLR)
- Signal Format: AES-3 (AES/EBU)
- AES-3 Input Compliance: 24-bit with selectable sample rate conversion, 20 kHz to 216kHz input sample rate capable.
- AES-3 Output Compliance: 24-bit

# SPECIFICATIONS

- Digital Reference: Internal (network timebase) or external reference 48 kHz, +/- 2 ppm
- Internal Sampling Rate: 48 kHz
- Output Sample Rate: 48 kHz
- A/D Conversions: 24-bit, Delta-Sigma, 256x oversampling
- D/A Conversions: 24-bit, Delta-Sigma, 256x oversampling
- Latency <3 ms, mic in to monitor out, including network and processor loop

## Frequency Response

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

## Dynamic Range

- Analog Input to Analog Output: 102 dB referenced to 0 dBFS, 105 dB "A" weighted to 0 dBFS
- Analog Input to Digital Output: 105 dB referenced to 0 dBFS
- Digital Input to Analog Output: 103 dB referenced to 0 dBFS, 106 dB "A" weighted
- Digital Input to Digital Output: 125 dB

## Equivalent Input Noise

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

## Total Harmonic Distortion + Noise

- Mic Pre Input to Analog Line Output: <0.005%, 1 kHz, -38 dBu input, +18 dBu output
- Analog Input to Analog Output: <0.008%, 1 kHz, +18 dBu input, +18 dBu output
- Digital Input to Digital Output: <0.0003%, 1 kHz, -20 dBFS
- Digital Input to Analog Output: <0.005%, 1 kHz, -6 dBFS input, +18 dBu output

## Crosstalk Isolation, Stereo Separation And CMRR

- Analog Line channel to channel isolation: 90 dB isolation minimum, 20 Hz to 20 kH
- Microphone channel to channel isolation: 80 dB isolation minimum, 20 Hz to 20 kHz

# SPECIFICATIONS

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

## Audio Processing

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

## Power Supply AC Input, QOR.16 With DESQ Console

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

## Operating Temperatures

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

## Dimensions

- W 15.31in (388.85 cm), H 15.30 in (388.72 cm), D 2.79 in (57.15 cm)

# AXIA STUDIOENGINE

## Bulletproof Power for 24/7 Operation.


## OVERVIEW

The networked Axia StudioEngine provides bulletproof mixing console signal processing for Element mixing consoles. Each StudioEngine is equipped with multiple simultaneous inputs, outputs, mix-minus feeds, monitor signals, etc. and can provide EQ for multiple channels, voice dynamics, studio headphone processing and multiple VMix (Virtual Mixer) channels.

The StudioEngine is fanless for cool, silent in-studio deployment, and is equipped with Gigabit Ethernet ports for network connection, and dual-redundant, field-replaceable internal power supplies with automatic switching for complete peace of mind.

In addition to providing mixing for physical console surfaces, StudioEngine can be used in conjunction with Axia SoftSurface Virtual Console software to create a powerful "virtual console" of up to 48 faders that's controlled with a standard Windows-based laptop — perfect for places where space constraints do not permit a physical mixing device.

# FEATURES

- Fanless design with heavy machined heat-sinks is completely silent in-studio.
- Front-panel OLED display delivers monitors power status, operating temperature, and a fast manual setup option.
- Telecom grade dual-redundant power supplies are designed for maximum uptime under harsh conditions. Automatic, seamless switching.
- Field-replaceable, internally fused fault-protected power modules change out in under 1 minute.
- Separate Gigabit LAN and WAN connections gives maximum network security while allowing administrative remote access.

# IN DEPTH

## Mixing Power, a la carte.

Although Axia mixing consoles resemble traditional broadcast consoles, no audio is actually mixed by or even passes through their faders. Instead, think of Axia consoles as a "remote control" for Axia DSP-based mixing engines.

The rugged StudioEngine is a standalone mixing engine for use with Axia Element control surfaces; it has no audio I/O of its own, instead allowing you maximum flexibility in designing your audio network with a la carte I/O using Axia xNode IP-Audio interfaces. In this way, you can construct bespoke systems to suit your specific needs.

# IN DEPTH

The StudioEngine itself is an extremely powerful mixing device, based on a blazingly-fast Intel processor that can out-perform even the largest dedicated-DSP embedded designs. The StudioEngine accesses audio streams, modifies them, and then presents the resulting streams back to the network as program output (or monitor output, or mix-minus output, et cetera). This approach is ideally suited to a network-based audio architecture since all input and output streams are routed through a Gigabit Ethernet port.

To deliver the reliability and ultra-low latency required, we equipped the StudioEngine with a fast, robust version of the Linux real-time operating system. Then we optimized our engine processing program so that total input to output latency is just a few hundred microseconds.

In fact, each StudioEngine has so much CPU power, it can outperform the very largest digital or router-based consoles, with multiple simultaneous inputs, outputs, mix-minus feeds, monitor signals, etc. for consoles as large as 40 faders. It can even provide EQ for and voice dynamics for multiple audio channels, as well as multiple VMix (virtual mixer) channels that allow combination of multiple audio channels on "virtual faders" that can then be mapped to a single physical fader. One StudioEngine supplies mixing power for even the largest Fusion or Element console.

Additionally, StudioEngine can be paired with Axia SoftSurface software to create a "virtual console" controlled by any computer with the Windows operating system — an ideal way of putting big mixing power into very small spaces.

# SPECIFICATIONS

## Power Supply AC Input

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

## Operating Temperatures

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

## Dimensions (HxWxD) and Weight

- 3.5 x 19 x 15 inches, 15 pounds

## Network Interface

- 2x 1000Base-T ports, standard RJ-45 connectors.

## Audio Processing

### Equalizer

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

### Compressor

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

### Expander/Noise Gate

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

### De-esser

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

# AXIA POWERSTATION

## Integrated Console Engine

## OVERVIEW

PowerStation is an all-in-one studio solution that combines audio I/O, a console power supply, mixing engine and built-for-broadcast network switch into one easy-to-deploy package. Each PowerStation Main provides 4 Analog inputs and 6 Analog outputs, 2 AES/EBU inputs and 2 AES/EBU outputs, 2 Microphone inputs with selectable Phantom power, 4 GPIO machine-control logic ports, each with 5 inputs and 5 outputs, an integrated network switch with 14 100Base-T Ethernet ports and 2 1000Base-T (Gigabit) ports with SFP, a heavy-duty Telecom-grade power supply with fanless convection cooling, and an industrial-grade CPU designed for harsh-environment reliability.

Use PowerStation Main with an Element mixing console as a standalone studio solution, or connect to other Axia equipment as part of a larger IP-Audio network. Simple Networking allows daisy-chain connection of up to 4 PowerStation-based studios without the use of an external network switch. Connecting a PowerStation Aux adds auto-switching redundant backup power and doubles audio I/O and GPIO capacity. I/O can also be easily expanded using Axia Audio Nodes.

# FEATURES

- Fanless design with heavy machined heat-sinks is completely silent in-studio.
- Front-panel status display monitors power and network status.
- Telecom grade power supplies are designed for maximum uptime under harsh conditions.
- Add a PowerStation Aux to PowerStation Main for dual-redundant power supply with automatic, seamless switching.
- Built-in, zero-configuration network switch with Gigabit and SFP for long-distance fiber connection.
- Large variety of built-in audio I/O boasts studio-grade audio performance specs.
- Add more I/O with PowerStation Aux, or a la carte using Axia Audio xNodes.

# IN DEPTH

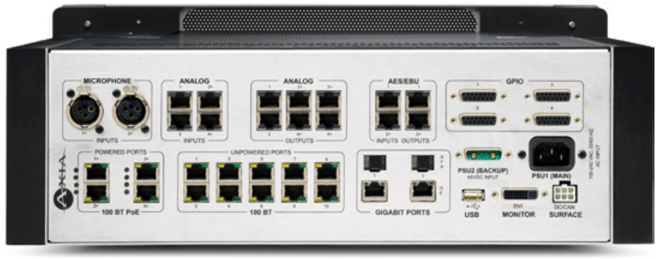
## There's no such thing as too much uptime.

If you set out to build a console engine designed to power your studio 24 hours a day, 7 days a week, 52 weeks a year, you probably wouldn't skimp. You'd equip it with the most bulletproof, telecom-grade power supply you could find. You'd give it a redundant-power option, for even more peace of mind. You'd make it convection-cooled — no noisy cooling fans to assault your quiet studio. You'd give it plenty of I/O — analog, digital, Mic-level and GPIO logic. And then, the pièce de résistance: you'd equip it with a zero-configuration, built-for-broadcast Ethernet switch.

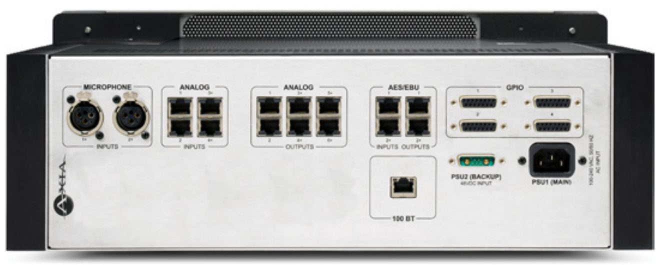
That's what we did when we designed PowerStation, the muscle behind our industry-leading Element mixing consoles. PowerStation is over-engineered to Axia standards, every part chosen for its ability to give constant, uninterrupted service. PowerStation combines four separate devices – a DSP mixing engine, a console CPU and power supply, audio I/O, GPIO and a custom, Axia-designed Ethernet switch – into a self-contained console engine that's engineered to ensure years of reliable, trouble-free service.

There are no compromises: PowerStation uses only best-of-the-best components, like studio-grade mic preamps and 24-bit, 256x oversampling A/D converters, a rigid, EM-tight chassis, an ultra-reliable DSP platform (not a common PC motherboard) and a hardened power supply designed for unfailing service, even in the harshest environments.

# IN DEPTH




PowerStation Main is where you start. Inside is a bulletproof mixing engine capable of powering consoles of up to 40 faders. There's a massive fanless, convection-cooled power supply. There are two Mic inputs, four Analog inputs and six outputs, two AES/EBU inputs and two outputs, and four GPIO ports, each with five opto-isolated inputs and five opto-isolated outputs. There are 14 100Base-T Ethernet ports with Livewire for single-cable connection of Telos phone systems, Omnia audio processors and other Axia equipment, as well as gear from our huge list of Livewire partners. Two Gigabit ports with SFP enable connection to other studios via copper or fiber. Just connect it to your Element console (it only takes a single cable), plug in your audio devices, and perform some fast web-based configuration. Add power and you're on the air. It's that simple!




To beef up your PowerStation studio even further, there's PowerStation Aux. Connect it to the PowerStation main to instantly double mic, analog, AES and GPIO ports, and add a redundant backup power supply with auto-switchover. Most redundant supplies protect only the console, but with PowerStation, the mixing engine, audio I/O and network switch are protected as well.

Best of all, there's that zero-configuration Ethernet switch that's built specifically to handle IP-Audio. No settings to tweak, no configuration code to upload – just plug it in and go. There are even two Gigabit ports with SFP, to connect to other studios via fiber or copper. You can even daisy-chain up to four PowerStation studios directly, for a self-contained network that doesn't require an external Ethernet switch. No other console company makes AoIP this easy.

# SPECIFICATIONS

## Microphone Preamplifiers

- Source Impedance: 150 Ohms
- Input Impedance: 4 k Ohms minimum, balanced
- Nominal Level Range: Adjustable, -75 dBu to -20 dBu
- Input Headroom: >20 dB above nominal input
- Output Level: +4 dBu, nominal

## Analog Line Inputs

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

## Analog Line Outputs

- Output Source Impedance: <50 Ohms balanced
- Output Load Impedance: 600 Ohms, minimum
- Nominal Output Level: +4 dBu
- Maximum Output Level: +24 dBu

## Digital Audio Inputs and Outputs

- Reference Level: +4 dBu (-20 dB FSD)
- Impedance: 110 Ohms, balanced (XLR)
- Signal Format: AES-3 (AES/EBU)
- AES-3 Input Compliance: 24-bit with selectable sample rate conversion, 32 kHz to 96kHz input sample rate capable.
- AES-3 Output Compliance: 24-bit
- Digital Reference: Internal (network timebase) or external reference 48 kHz, +/- 2 ppm
- Internal Sampling Rate: 48 kHz
- Output Sample Rate: 44.1 kHz or 48 kHz

# SPECIFICATIONS

- A/D Conversions: 24-bit, Delta-Sigma, 256x oversampling
- D/A Conversions: 24-bit, Delta-Sigma, 256x oversampling
- Latency <3 ms, mic in to monitor out, including network and processor loop

## Frequency Response

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

## Dynamic Range

- Analog Input to Analog Output: 102 dB referenced to 0 dBFS, 105 dB "A" weighted to 0 dBFS
- Analog Input to Digital Output: 105 dB referenced to 0 dBFS
- Digital Input to Analog Output: 103 dB referenced to 0 dBFS, 106 dB "A" weighted
- Digital Input to Digital Output: 138 dB

## Equivalent Input Noise

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

## Total Harmonic Distortion + Noise

- Mic Pre Input to Analog Line Output: <0.005%, 1 kHz, -38 dBu input, +18 dBu output
- Analog Input to Analog Output: <0.008%, 1 kHz, +18 dBu input, +18 dBu output
- Digital Input to Digital Output: <0.0003%, 1 kHz, -20 dBFS
- Digital Input to Analog Output: <0.005%, 1 kHz, -6 dBFS input, +18 dBu output

## Crosstalk Isolation, Stereo Separation and CMRR

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

# SPECIFICATIONS

## Audio Processing

### Equalizer

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

### Compressor

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

### Expander/Noise Gate

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

### De-esser

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

## Power Supply AC Input, PowerStation Aux & Main

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

## Operating Temperatures

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

## Dimensions (HxWxD) and Weight

- PowerStation Main/Aux: 7 x 19 x 15.5 inches (behind rail)
- Front panel extends 2.25 inches in front of rack rail
- PowerStation Main: 45 pounds
- PowerStation Aux: 40 pounds

# AXIA QOR.32

## Integrated Console Engine

## OVERVIEW

QOR.32 is an Axia integrated console engine for iQ mixing consoles that combines audio I/O, a console power supply, mixing engine and built-for-broadcast network switch into one easy-to-deploy package. Each QOR.32 provides 16 Analog inputs and 8 Analog outputs, 2 AES/EBU inputs and 2 AES/EBU outputs, 4 Microphone inputs with selectable Phantom power, 8 GPIO machine-control logic ports, each with 5 inputs and 5 outputs, an integrated network switch with 6 Livewire 100Base-T Ethernet ports and 2 1000Base-T (Gigabit) ports with SFP, a heavy-duty Telecom-grade power supply with fanless convection cooling, and an industrial-grade CPU designed for harsh-environment reliability.

Use QOR.32 with an Axia iQ mixing console as a standalone studio solution, or connect to other Axia equipment as part of a larger IP-Audio network. Simple Networking allows daisy-chain connection of up to 4 QOR-based studios without the use of an external network switch. Connecting a QOR Backup adds auto-switching redundant backup power. I/O can easily be expanded using Axia Audio Nodes.

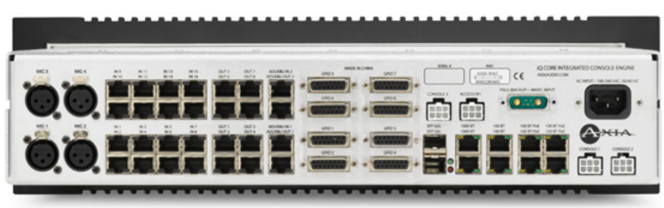
# FEATURES

- Fanless design with heavy machined heat-sinks is completely silent in-studio.
- Front-panel LED display monitors power and network status.
- Telecom grade power supplies are designed for maximum uptime under harsh conditions.
- Add a QOR Backup to QOR.32 for dual-redundant power supply with automatic, seamless switching.
- Built-in, zero-configuration network switch with Gigabit and SFP for long-distance fiber connection.
- Large variety of built-in audio I/O boasts studio-grade audio performance specs.
- Add more I/O a la carte using Axia Audio xNodes.

# IN DEPTH

## QOR.32 Integrated Console Engine

The QOR.32 integrated console engine is a DSP-based mixing engine with onboard I/O, GPIO, console power supply and custom-built, configuration-free Ethernet switch. You'll find plenty of I/O, including mic inputs with selectable Phantom power, analog and AES/EBU inputs and outputs, plenty of GPIO machine-control logic ports, and that powerful integrated Ethernet switch with Livewire ports to add local sources, and Gigabit ports for networking with the rest of your plant. That's a lot of I/O, but if you need more you can instantly add it just by plugging in Axia xNode audio interfaces. And QOR.32 is convection-cooled for utterly silent, fan-free operation.



Let's take a look around back, shall we? You'll find everything you need for an average, medium-sized studio: 4 mic inputs with selectable Phantom power, 16 stereo analog inputs and 8 stereo analog outputs, 2 AES/EBU inputs and 2 AES/EBU outputs, 8 GPIO machine-control logic ports (each with 5 opto-isolated inputs and 5 outputs).

# IN DEPTH

There's Livewire I/O as well: the QOR.32 has an integrated Ethernet switch with 6 Livewire 100Base-T ports. 4 of those ports have PoE (Power over Ethernet) that you can use to connect and power networked devices compatible with the IEEE 802.1af PoE standard (like our xNode audio interfaces, or Telos VSet phones). You'll also find 2 1000Base-T Gigabit ports (RJ-45 & SFP) that you can use to connect to other studios. 4 CANBus ports provide for connection of up to 3 Axia iQ console frames, allowing construction of consoles up to 24 faders in size.

By the way, that zero-configuration Ethernet switch is built specifically to handle IP-Audio. No settings to tweak, no configuration code to upload – just plug it in and go. The built-in configurable network gateway allows loading sources from other studios, while simultaneously exporting audio streams for use elsewhere; the gateway can be configured for 12-in, 4-out or 8-in, 8-out modes. You can even daisy-chain up to four QOR-based studios directly, for a self-contained network that doesn't require an external Ethernet switch. No other console company makes AoIP this easy.

For installations that require redundant backup power, there's the QOR Backup. Connect it to the QOR.32 and you've added a redundant backup power supply with auto-switchover. Most redundant supplies protect only the console, but with Axia's integrated console engine concept, the mixing engine, local audio I/O and network switch are protected as well.

# SPECIFICATIONS

## QOR.32 Connections

- Microphone Inputs: 4x balanced XLR-F, with selectable Phantom power
- Analog Inputs: 16x RJ-45, StudioHub+ standard.
- Analog Outputs: 8x RJ-45, StudioHub+ standard.

# SPECIFICATIONS

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

## Microphone Preamplifiers

- Source Impedance: 150 ohms
- Input Impedance: 4 k ohms minimum, balanced
- Nominal Level Range: Adjustable, -75 dBu to -20 dBu
- Input Headroom: >20 dB above nominal input
- Output Level: +4 dBu, nominal

## Analog Line Inputs

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

## Analog Line Outputs

- Output Source Impedance: <50 ohms balanced
- Output Load Impedance: 600 ohms, minimum
- Nominal Output Level: +4 dBu
- Maximum Output Level: +24 dBu

# SPECIFICATIONS

## Digital Audio Inputs And Outputs

- Reference Level: +4 dBu (-20 dB FSD)
- Impedance: 110 Ohm, balanced (XLR)
- Signal Format: AES-3 (AES/EBU)
- AES-3 Input Compliance: 24-bit with selectable sample rate conversion, 20 kHz to 216kHz input sample rate capable.
- AES-3 Output Compliance: 24-bit
- Digital Reference: Internal (network timebase) or external reference 48 kHz, +/- 2 ppm
- Internal Sampling Rate: 48 kHz
- Output Sample Rate: 48 kHz
- A/D Conversions: 24-bit, Delta-Sigma, 256x oversampling
- D/A Conversions: 24-bit, Delta-Sigma, 256x oversampling
- Latency <3 ms, mic in to monitor out, including network and processor loop

## Frequency Response

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

## Dynamic Range

- Analog Input to Analog Output: 102 dB referenced to 0 dBFS, 105 dB "A" weighted to 0 dBFS
- Analog Input to Digital Output: 105 dB referenced to 0 dBFS
- Digital Input to Analog Output: 103 dB referenced to 0 dBFS, 106 dB "A" weighted
- Digital Input to Digital Output: 125 dB

## Equivalent Input Noise

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

## Total Harmonic Distortion + Noise

- Mic Pre Input to Analog Line Output: <0.005%, 1 kHz, -38 dBu input, +18 dBu output
- Analog Input to Analog Output: <0.008%, 1 kHz, +18 dBu input, +18 dBu output

# SPECIFICATIONS

- Digital Input to Digital Output: <0.0003%, 1 kHz, -20 dBFS
- Digital Input to Analog Output: <0.005%, 1 kHz, -6 dBFS input, +18 dBu output

## Crosstalk Isolation, Stereo Separation And CMRR

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

## Audio Processing

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

## Power Supply AC Input, QOR.32 / QOR Backup

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

## Operating Temperatures

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

# AXIA QOR.16

## Integrated Console Engine

## OVERVIEW

QOR.16 is an Axia integrated console engine for Radius, DESQ and RAQ mixing consoles. QOR.16 combines audio I/O, a console power supply, mixing engine and built-for-broadcast network switch into one easy-to-deploy package. Each QOR.16 provides 8 Analog inputs and 4 Analog outputs, 1 AES/EBU input and 1 AES/EBU output, 2 Microphone inputs with selectable Phantom power, 4 GPIO machine-control logic ports, each with 5 inputs and 5 outputs, an integrated network switch with 6 Livewire 100Base-T Ethernet ports and 2 1000Base-T (Gigabit) ports with SFP, a heavy-duty Telecom-grade power supply with fanless convection cooling, and an industrial-grade CPU designed for harsh-environment reliability.

Use QOR.16 with a Radius, DESQ or RAQ mixing console as a standalone studio solution, or connect to other Axia equipment as part of a larger IP-Audio network. Simple Networking allows daisy-chain connection of up to 4 QOR-based studios without the use of an external network switch. I/O can easily be expanded using Axia Audio Nodes.

# FEATURES

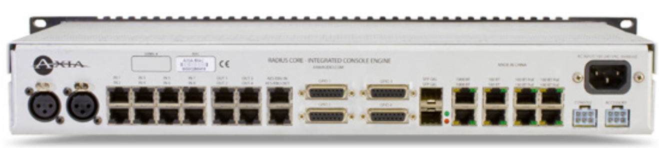
- Fanless design with heavy machined heat-sinks is completely silent in-studio.
- Front-panel LED display monitors power and network status.
- Telecom grade power supply is designed for maximum uptime under harsh conditions.
- PoE (Power over Ethernet) capability can supply power for PoE-compliant studio devices.
- Built-in, zero-configuration network switch with Gigabit and SFP for long-distance fiber connection.
- Large variety of built-in audio I/O boasts studio-grade audio performance specs.
- Add more I/O a la carte using Axia Audio xNodes.

# IN DEPTH

## QOR.16 Integrated Console Engine

The QOR.16 integrated console engine is a DSP-based mixing engine with onboard I/O, GPIO, console power supply and custom-built, configuration-free Ethernet switch. It's the smaller brother of our QOR.32 integrated console engine, designed and built with the same high-grade components for deployment with Radius, DESQ and RAQ consoles in smaller studios where large amounts of I/O are not required.

QOR.16 comes with a wide variety of I/O, including mic inputs with selectable Phantom power, analog and AES/EBU inputs and outputs, plenty of GPIO machine-control logic ports, and that powerful integrated Ethernet switch with Livewire ports to add local sources, and Gigabit ports for networking with the rest of your plant. If more you I/O is needed, you can instantly add it just by plugging in Axia xNode audio interfaces. And QOR.16 is convection-cooled for utterly silent, fan-free operation.



QOR.16 has all the analog and digital inputs and outputs an average small studio requires: 2 mic inputs with selectable Phantom power, 8 stereo analog inputs and 4 stereo analog outputs, 1 AES/EBU input and 1 AES/EBU output, and 4 GPIO machine-control logic ports (each with 5 opto-isolated inputs and 5 outputs).

# IN DEPTH

Of course, Livewire connections are built in. The QOR.16 has an integrated Ethernet switch with 6 Livewire 100Base-T ports. 4 of those ports have PoE (Power over Ethernet) that you can use to connect and power networked devices compatible with the IEEE 802.1af PoE standard (like our xNode audio interfaces, or Telos VSet phones). You'll also find 2 1000Base-T Gigabit ports (RJ-45 & SFP) that you can use to connect to other studios.

By the way, that zero-configuration Ethernet switch is built specifically to handle IP-Audio. No settings to tweak, no configuration code to upload – just plug it in and go. The built-in configurable network gateway allows loading sources from other studios, while simultaneously exporting audio streams for use elsewhere; the gateway can be configured for 12-in, 4-out or 8-in, 8-out modes. You can even daisy-chain up to four QOR-based studios directly, for a self-contained network that doesn't require an external Ethernet switch. No other console company makes AoIP this easy.

And here's a neat trick: if you're building audio workstations, news bullpens or ingest facilities, where small consoles like Axia DESQ or RAQ shine, a single QOR.16 can provide mixing power for two DESQ or RAQ mixers — or one of each! Just another way choosing Axia helps stretch your equipment budget.

# SPECIFICATIONS

## QOR.16 Connections

- Microphone Inputs: 2x balanced XLR-F, with selectable Phantom power
- Analog Inputs: 8x RJ-45, StudioHub+ standard.
- Analog Outputs: 4x RJ-45, StudioHub+ standard.
- AES/EBU Inputs: 1x RJ-45, StudioHub+ standard.
- AES/EBU Outputs: 1x RJ-45, StudioHub+ standard.
- GPIO: 4x DB-15
- Livewire:
    - 4x 100Base-T with PoE, RJ-45
    - 2x 100Base-T, RJ-45
    - 2x 1000Base-T, RJ-45
    - 2x Gigabit, SFP (Small Form Pluggable)

# SPECIFICATIONS

- Console Frame Connections: 1x, 6-pin "latch and lock" style
- Accessory Connections: 1x, 6-pin "latch and lock" style

## Microphone Preamplifiers

- Source Impedance: 150 ohms
- Input Impedance: 4 k ohms minimum, balanced
- Nominal Level Range: Adjustable, -75 dBu to -20 dBu
- Input Headroom: >20 dB above nominal input
- Output Level: +4 dBu, nominal

## Analog Line Inputs

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

## Analog Line Outputs

- Output Source Impedance: <50 ohms balanced
- Output Load Impedance: 600 ohms, minimum
- Nominal Output Level: +4 dBu
- Maximum Output Level: +24 dBu

## Digital Audio Inputs And Outputs

- Reference Level: +4 dBu (-20 dB FSD)
- Impedance: 110 Ohm, balanced (XLR)
- Signal Format: AES-3 (AES/EBU)
- AES-3 Input Compliance: 24-bit with selectable sample rate conversion, 20 kHz to 216kHz input sample rate capable.
- AES-3 Output Compliance: 24-bit
- Digital Reference: Internal (network timebase) or external reference 48 kHz, +/- 2 ppm
- Internal Sampling Rate: 48 kHz

# SPECIFICATIONS

- Output Sample Rate: 48 kHz
- A/D Conversions: 24-bit, Delta-Sigma, 256x oversampling
- D/A Conversions: 24-bit, Delta-Sigma, 256x oversampling
- Latency <3 ms, mic in to monitor out, including network and processor loop

## Frequency Response

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

## Dynamic Range

- Analog Input to Analog Output: 102 dB referenced to 0 dBFS, 105 dB "A" weighted to 0 dBFS
- Analog Input to Digital Output: 105 dB referenced to 0 dBFS
- Digital Input to Analog Output: 103 dB referenced to 0 dBFS, 106 dB "A" weighted
- Digital Input to Digital Output: 125 dB

## Equivalent Input Noise

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

## Total Harmonic Distortion + Noise

- Mic Pre Input to Analog Line Output: <0.005%, 1 kHz, -38 dBu input, +18 dBu output
- Analog Input to Analog Output: <0.008%, 1 kHz, +18 dBu input, +18 dBu output
- Digital Input to Digital Output: <0.0003%, 1 kHz, -20 dBFS
- Digital Input to Analog Output: <0.005%, 1 kHz, -6 dBFS input, +18 dBu output

## Crosstalk Isolation, Stereo Separation And CMRR

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

# SPECIFICATIONS

## Audio Processing

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

## Power Supply AC Input, QOR.16 With Radius Console

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

## Operating Temperatures

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

# XNODES IP-AUDIO INTERFACES

## The most advanced AoIP interfaces on the planet.

## OVERVIEW

xNodes are the new generation of light, half-rack, high-performance IP-Audio interfaces from Axia. They're loaded with advanced features and capabilities. One-button configuration takes a new xNode from out-of-the-box to on-the-air in under one minute. They're fanless, which means they're noiseless too. They have a versatile mounting arrangement that lets you deploy two xNodes in just 1RU of rack space, as well as by themselves on ceilings, walls or under counters with the available wall-mount kit. They've got studio-grade audio performance specs. And they even have managed power options (via AC mains or Ethernet PoE) and dual network interfaces for redundant operation.

xNodes are available in Analog, AES/EBU, Microphone-level, Mixed-Signal and GPIO versions to handle virtually any signal encountered in today's broadcast studio.

# FEATURES

- Fanless design with heavy cast-aluminum heat-sinks is completely silent in-studio. Front-panel heat sinks are cooled by ambient air, not "rack air," eliminating overheating worries.
- High-resolution front-panel multi-function OLED display meters inputs and outputs or GPIO status, gives software and other status information.
- xNodes feature redundant power with auto-switching. Multi-color LED indicator glows green when AC mains power is present, red when running on PoE, and orange when both AC and PoE are connected.
- Redundant power plan lets xNodes run on AC or Power over Ethernet (IEEE 802.3af) supplied by compliant Ethernet switches (such as Axia xSwitch and QOR integrated console engines). xNode will run on either power supply; connect both for redundant power with automatic switchover.
- Power-efficient: xNodes use just 14 Watts each.
- Network connection has backup, too: dual NICs allow you to connect xNode to separate network branches for full audio pathway redundancy. Automatic failover activates backup connection should the primary be interrupted.
- Built-in Syslog server with configurable event filter and SNMP (Simple Network Management Protocol) support help you stay fully informed, should an xNode's power or connection status change.
- Synchronize your AES master clock to a designated xNode AES/EBU input to keep all of your AES streams synchronized to the house clock.
- xNodes use premium components: rugged cast aluminum faceplates and heat sinks, high-resolution OLED displays, bulletproof power supplies designed for high-availability telecom applications, studio-quality SRCs with recording-studio specs.
- I/O connections via industry-standard RJ-45 audio connectors or high-density DB-25 connections, both available prefabricated and ready to attach in seconds.
- Versatile mounting options: use freestanding, rack singly or side-by-side, or wall-mount using an accessory surface-mount kit.
- Analog xNodes can be configured to create 4 stereo audio channels for your network to use, or 8 true mono channels. They can also receive 4 stereo or 8 mono channels, and send them to its outputs. This option is easily selectable in software via the built-in Web interface.

# IN DEPTH

## The Power of the Network.

One day, all audio equipment will be networked. Until then, there are xNodes, the world's first self-configuring AoIP interfaces.

xNodes give you an easy way to add non-networked audio devices to your studio network. And they pack a lot of I/O into a very small space. xNodes nearly configure themselves. Just plug them in and they go to work, configuring channel numbers and even signal names (editable by you) all by themselves. In just moments, you're ready to start sending and receiving network audio.

All xNodes feature a high-resolution OLED front panel display and two "soft" buttons to provide status information and assist with initial setup, and a multi-color LED that gives at-a-glance information about the xNode's power configuration. To ensure ultra-reliable network operations and extremely low delay, xNodes run Linux on an embedded processor, and a built-in web server in each node gives you remote configuration and control – in an intuitive, easy-to-understand manner – using any standard Web browser.

xNodes are loaded with features designed to ensure the uptime of your network. Dual Ethernet ports can provide redundant connections to separate network segments. Redundant power capability with automatic switchover enables xNodes to run on house mains or PoE (Power over Ethernet), letting the network switch itself supply power, and enabling easy single-cable setup in places where AC power isn't practical. Built-in Syslog servers with a configurable event filter and SNMP (Simple Network Management Protocol) support let you stay fully informed, should an xNode's power or connection status change.

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

xNodes are easy to deploy — easier than any other AoIP interface. When you connect an xNode to your network, it automatically prompts you to give it an ID via the front-panel controls. Then, it derives a unique static IP address, and even gives names to its sources and outputs (which you can edit later, from the comfort of your computer). All you have to do is connect devices to the inputs, and it advertises that its audio sources are available for use, allowing any users access to them.

# IN DEPTH

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

xNodes are easy to deploy, too. They're fanless, so you can tuck one anywhere you need I/O. They're compact: two xNodes fit side-byside in a single rack space using a simple rack-mount kit. Or, mount them to walls, ceilings, under countertops, using the optional surface-mount kit.

5 different xNodes provide analog and AES ins and outs, microphone inputs and GPIO logic ports, wherever you need them. No need for "home runs" to a central rack – one CAT-5 cable connection is all an xNode needs to interface multiple channels of bi-directional audio to your network.

## Microphone xNode

The Microphone xNode has four professional-grade microphone preamps with selectable Phantom power and software-adjustable gain. There are also four balanced analog line outputs to conveniently deliver headphone and studio monitor feeds back to your talent. Inputs and outputs are presented both on easy-to-install RJ-45s and high-density DB-25s, both of which connect to easily available 3rd-party breakout cables, to suit your connection preference.

## Analog xNode

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

# IN DEPTH

## AES/EBU xNode

Our AES/EBU xNode has 4 AES/EBU inputs and 4 AES/EBU outputs. Left and right input signals may be split and routed independently as mono signals. Stunning performance specs include 48 kHz sampling rate, 126dB of dynamic range, and <0.0003% THD. Sample rate conversion is available on all inputs; the unit can also be synchronized to a house clock to provide sync to your entire Axia network.

## Mixed-Signal xNode

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

## GPIO xNode

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

# SPECIFICATIONS

## Microphone Preamplifiers:

- Source Impedance: 150 Ohms
- Input Impedance: 4 k Ohms minimum, balanced
- Nominal Level Range: Adjustable, -75 dBu to -20 dBu
- Input Headroom: >20 dB above nominal input
- Phantom power: +48VDC, switchable

## Analog Line Inputs:

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

## Analog Line Outputs:

- Output Source Impedance: <50 Ohms balanced
- Output Load Impedance: 600 Ohms, minimum
- Nominal Output Level: +4 dBu
- Maximum Output Level: +24 dBu

## Digital Audio Inputs And Outputs

- Reference Level: +4 dBu (-20 dB FSD)
- Impedance: 110 Ohm, balanced
- Signal Format: AES3 (AES/EBU)
- AES3 Input Compliance: 24-bit with sample rate conversion
- AES3 Output Compliance: 24-bit
- Digital Reference: Internal (network timebase) or external reference 48 kHz, +/- 2 ppm
- Internal Sampling Rate: 48 kHz
- Input Sample Rate: 32 kHz to 192 kHz
- Output Sample Rate: 44.1 kHz or 48 kHz

# SPECIFICATIONS

- A/D Conversions: 24-bit, Delta-Sigma, 256x oversampling
- D/A Conversions: 24-bit, Delta-Sigma, 256x oversampling

## Frequency Response

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

## Latency

- Analog Input to Analog Output, 2.75ms including network, converters, and mixing process
- Digital Input to Digital Output, 1.75ms including network mixing engine (ASRC off)

## Dynamic Range

- Analog Inputs to Analog Outputs 108dB referenced to 0dBFs, 111dB A-weighted
- Analog Inputs to Digital Outputs 110dB referenced to 0dBFs, 113dB A-weighted
- Digital Inputs to Analog Outputs 112dB referenced to 0dBFs, 115dB A-weighted
- Digital Inputs to Digital Outputs 126dB

## Equivalent Input Noise

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

## Total Harmonic Distortion + Noise

- Mic Pre Input to Analog Output: < 0.005%, 1 kHz, -36dBu input, +18dBu output
- Analog Input to Analog Output: < 0.005%, 1 kHz, +18dBu input, +18dBu output
- Analog Input to Digital Output: < 0.004%, 1 kHz, +18dBu input, -6dBFs output
- Digital Input to Analog Output: < 0.004%, 1 kHz, -6dBFs input, +18dBu output
- Digital Input to Digital Output: < 0.0003%, 1 kHz, -20dBFs

## Crosstalk Isolation, Stereo Separation And Cmrr

- Analog Line channel to channel isolation: 90dB minimum, 20Hz to 20kHz
- Analog Line stereo separation: 85dB minimum, 20Hz to 20kHz

# SPECIFICATIONS

- Analog Line Input CMRR: 80dB minimum, 20Hz to 20kHz
- Microphone Input CMRR: >60 dB, 20 Hz to 20 kHz

## Power Supply Ac Input

- Auto-ranging supply, 95VAC to 240VAC, 50 Hz to 60 Hz, IEC receptacle, internal fuse
- Power consumption: 14 Watts

## Operating Temperatures

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

## Dimensions

- 8.5" (22 cm) wide; two may be mounted side-by-side in a standard 1RU rack space; 1.72" (4.4 cm) height, 11.75" (30 cm) deptht

# XSWITCH

## The Network Switch Built for IP-Audio

## OVERVIEW

xSwitch is the world's only zero-configuration Ethernet switch optimized for Livewire IP-Audio applications. Fast setup requires only IP address assignment via front-panel OLED display or Axia iProbe software. Features 8 10/100MBit Ethernet ports — 4 with Power-over-Ethernet to power Axia xNodes, Telos VSet phones, and other networked devices compatible with the IEEE 802.1af PoE standard. Two Gigabit ports are provided for trunking, both with RJ-45 (copper) and SFP (fiber) connections; supports redundant copper/SFP Gigabit connections with auto-switching. Supports 2,000 Multicast groups and 2,000 ARP table entries (8x more than other small-form Ethernet switches). Web-based management interface uses built-in HTTP server. 9.5" x 11" half-rack form factor allows two xSwitches to be racked side-by-side, or placed in a rackmount with Axia xNode IP-Audio interfaces. Noiseless and fan-free; can be conveniently placed adjacent to your audio devices, rack-mounted using included hardware, or wall-mounted (with an accessory kit available separately).

# FEATURES

- Fanless design with heavy cast-aluminum heat-sinks is completely silent in-studio. Front-panel heat sinks are cooled by ambient air, not "rack air," eliminating overheating worries.
- Friendly OLED front-panel display with port status, IP address, PoE status and operating temperature readouts - features not available in other switches in this class.
- One-button setup eliminates programming and saves hours of setup time.
- Functions as a core switch for standalone studios, or as an edge switch in larger facilities, or at your Ethernet-connected transmitter site.
- Allows Axia network admins to add network ports economically, a la carte, instead of 24 or 48 at a time.
- xSwitch supports IGMP (Internet Group Management Protocol) Version 2, used to manage Multicast group traffic (an essential part of Livewire's intelligent audio routing system).
- xSwitch can handle up to 2,000 Multicast groups, and 2,000 ARP table entries, meaning it can't run out of bandwidth. (Other 8-port switches support only 250 groups.)
- Superior support for low-latency media streams, using four-level hardware strict priority QoS — other switches have only one strict priority queue.
- Works with Axia iProbe network management software, allowing easy administration from your office PC or remotely via WAN connection.
- Part of Axia's xNode family, xSwitch can be used as a freestanding device or racked singly or side-by-side with other xSwitch or xNode devices.
- Premium components include rugged cast faceplate and heat sinks, high-resolution OLED displays, and bulletproof power supplies designed for high-availability telecom applications.

# IN DEPTH

## Your radio station needs programming. Your network switch shouldn't.

We invented Livewire in 2003, with the idea of saving money in broadcast studio construction by using off-the-shelf Ethernet switches to power networks that distribute broadcast-quality audio nearly anywhere — across the hall, across the building, or across town. Some said it would never work! But

# IN DEPTH

10 years later, Axia is the #1 brand of IP consoles, networks and routing equipment to broadcasters worldwide. Maybe it's because Livewire IP-Audio is so flexible and easy to use that clients regularly tell us of days – even weeks – shaved off of studio installation time with components that simply click together using Cat-5 cables. Not to mention the money they've saved with Axia, compared to old-fashioned hard-wired studio builds.

But Axia fans told us there was one thing that could make Livewire even easier to install: A network switch that doesn't require setup or programming. So our engineers went to work. The result: xSwitch, the world's only zero-configuration network switch designed specifically for the needs of IP-Audio broadcasting.

xSwitch is different from any other Ethernet switch, because it's custom-tailored to the needs of Axia Livewire users. You see, third-party switches – even those certified for use with Axia – require programming to correctly configure them with the QoS settings Axia networks demand. Which generally means connecting a PC to the switch with a special cable, downloading a terminal emulation program, and entering lines of parameters and instructions.

Perhaps you've already got an Axia network installed (thank you!). Will an xSwitch work with the Axia gear you already have? Naturally! xNodes speak Livewire, the AoIP protocol that powers more than 50,000 networked pro audio devices at radio and TV stations around the world. One click to hook up, and they're ready to go.

xSwitch does away with switch programming. Our experts have already pre-configured xSwitch with all the instructions needed to run Livewire perfectly, flawlessly, out of the box. All you have to do is plug it in, perform a quick one-button setup, and start connecting Livewire devices. Easy, yes?


On the xSwitch's connection panel, you'll find two SFP (Small Form-Factor Pluggable) Gigabit ports, in addition to dual 1000Base-T copper ports. Use the SFP ports for copper or fiber connections to your Livewire network. The adjacent 1000BT copper ports provide a dual-redundant network interface; if the primary network link is interrupted, the secondary backup connection is automatically activated. You'll also find 8 100-BaseT Livewire ports, 4 with PoE (Power over Ethernet) to power xNodes audio adapters, Telos VSet telephones, or any other network device that uses the IEEE 802.3af standard.

# IN DEPTH

Speaking of power, note the internal, auto-ranging power supply with professional IEC connector: you'll never find wall-warts powering Axia gear.

xSwitch is built using the chassis developed for our award-winning xNode family of AoIP audio adapters, the latest generation of half-rack, high-performance IP-Audio interfaces. They're fanless, which means they're noiseless too; you can put them in any studio. They have a versatile mounting arrangement that lets you deploy two xSwitches into just 1RU of rack space (or rack an xSwitch alongside an xNode). This allows you the flexibility to do things impossible before — like combine an xSwitch with xNodes to create a "Supernode". An xSwitch connecting 8 analog xNodes creates a 32x32 stereo router - or a 64x64 mono router - in the space of just 4RU. Great for making an audio snake, for adding I/O to that add-on studio on the next floor, or even as the heart of a standalone studio.

# SPECIFICATIONS

## Power Supply AC Input

- Auto-ranging supply, 95VAC to 240VAC, 1.0 A, 50 Hz to 60 Hz
- IEC receptacle, internal fuse
- Power consumption: 75 Watts (all PoE ports under load)

## Power over Ethernet

- 15.4 W-per-port maximum, 61.6-W switch maximum

## Environmental Ranges

- Operating temperature: 32° F to 104° F (0°C to 40°C),
- Relative humidity: <90% (noncondensing)

# SPECIFICATIONS

## Physical Dimensions

- 8.5" (22 cm) wide; two may be mounted side-by-side in a standard 1RU rack space (with included mounting kit)
- 1.72" (4.4 cm) height, 11.75" (30 cm) depth
- Shipping Weight: 7 lbs. (3.2 kg.)
- Shipping Dimensions: 17" (43.2 cm) length, 13" (33 cm) width, 7" (17.8 cm) height

## Ethernet Switch Specifications

- 4 QoS levels
- VLANs supported: 1
- Hardware filter capacity: 8,000 (this is the total limit of MAC addresses + multicast group count supported).
- Supported protocols:
    - IPv4 hardware switching
    - IGMP version 2 snooping
    - IGMP snooping querier
    - DSCP (IP Type Of Service based priority)
    - 802.1p (Ethernet 802.1Q tag priority)
    - HTTP (WEB based management)
- Ports 100BT 1, 2, 3, 4: Fast Ethernet (10/100MBit/s), Power-over-Ethernet source
- Ports 100BT 5, 6, 7, 8: Fast Ethernet (10/100MBit/s)
- Ports GIG 1, 2: Copper or SFP (Small Factor Pluggable Transceiver) module

## IGMP Snooping Parameters

- Router present time out: 400s
- Query Response Interval: 10s

# SPECIFICATIONS

## Connector Specifications

### 10/100/1000 Ports

The 10/100/1000 Ethernet ports use standard RJ-45 connectors.

### Connecting to 100BASE-T-Compatible Devices

When connecting the ports to 100BASE-TX-compatible devices, you can use a two or four twisted-pair, Category 5e, straight-through cable.

### Connecting to 1000BASE-T Devices

When connecting the ports to 1000BASE-T devices, you must use a four twisted-pair, Category 6, straight-through cable.

### SFP Module Ports

The SFP module slot on a dual-purpose port uses SFP modules for fiber-optic and copper uplink ports. xSwitch works with the following supported SFP modules:

- Cisco Copper SFP Model:GLC-T=
- Cisco Copper SFP Model: SFP-GE-T=
- Cisco Multimode fiber model: GLC-SX-MMD=
- Cisco Multimode fiber model: GLC-SX-MM-RGD

# XSELECTOR ROUTER PANEL

## The Network Switch Built for IP-Audio


## OVERVIEW

The Axia xSelector combines the routing functions of an XY router control panel with the audio outputs of an Axia xNode. In addition to analog, AES3 and headphone outputs, the Router Selector Node also features an analog and an AES3 input — ideal for production or news studios where operators both create and play audio streams. Six convenient "radio buttons" can be quickly programmed for instant access to favorite sources.

# FEATURES

- Fanless design for silent in-studio operation.
- High-resolution front-panel multi-function OLED display meters inputs and outputs and provides audio source selection controls.
- Local I/O connections via industry-standard RJ-45 or XLR audio connectors,.
- In addition to being able to select audio streams from the Livewire network for use locally, the xSelector features one stereo input and one stereo output, allowing fast network distribution of locally created streams from audio workstations or portable audio devices. Each xSelector can create 1 stereo Livewire stream, which becomes available to other devices on the Livewire network.
- Local I/O is presented on both AES digital and analog balanced inputs and outputs. The user can feed audio into either a balanced analog input or an AES input.
- Both the AES and analog outputs are active simultaneously; both outputs have the same audio present.
- Includes two GPIO closures presented on standard DB-15 connectors for machine control of associated devices.
- xSelector's stereo outputs can be assigned to output either the locally-created audio stream, or a single stereo Livewire stream acquired from the network and easily selected from a list of available streams using the front panel OLED display.
- Six frequently-used streams can be assigned to the front panel "radio buttons" for instant access. Filmcap buttons can be labeled with names of assigned channels if desired.
- Dual Livewire 100Base-T Ethernet ports for redundant connection to your Livewire audio network.
- Front panel headphone jack and volume control make xSelector a valuable addition to dubbing and ingest stations where minimal infrastructure is desired.
- Built-in HTTP server for easy remote control using any PC with a Web browser.

# IN DEPTH

## The production-room powerhouse.

The Axia xSelector looks a lot like a traditional XY router control panel, but it's much, much more. So much more, in fact, that you'll make xSelector a staple in your TOC, production rooms, news stations — anywhere your talent needs to both create and consume networked audio streams.

# IN DEPTH

xSelector lets talent select from all available audio streams on the Livewire network, and route them to its local output (conveniently presented in both balanced analog and AES/EBU format). xSelector is easy to use: The front-panel LCD screen lists available network sources; talent uses the adjacent selector knob to browse sources and then pushes the knob to "take" the selected source, instantly routing that source to the local outputs for use with an audio workstation, a specific console input, recording device, etc.

Also on the front panel, six film-cap "radio buttons" provide instant access to frequently-used sources. There's also a stereo ¼" TRS jack with a volume control which supplies an internally-amplified audio output directly to talent headphones, making xSelector a perfect choice for small workstation environments by eliminating the need for an external headphone amp.

Around back, you'll find separate left and right balanced XLR and RJ-45 connections for the analog input and output, another set of XLR and RJ-45 connectors for the AES/EBU input and output, DB-15 connectors for the two GPIO machine-logic controls, and RJ-45s for the two redundant Livewire 100Base-T Ethernet connections.

All this functionality makes xSelector the perfect choice for news booths or dubbing stations where only one active feed is required, or for intake stations that allow non-technical folks to easily move audio from external sources (like field recorders) into the Axia network.

# SPECIFICATIONS

## Connections:

### AES/EBU

- 1x Stereo Input, presented on one XLR-F connection and one RJ-45 connection
- 1x Stereo Output, presented on one XLR-M connection and one RJ-45 connection

# SPECIFICATIONS

## Analog

- 1x Stereo, presented on two XLR-F connections and one RJ-45 connection
- 1x Stereo, presented on two XLR-M connections and one RJ-45 connection

## GPIO

- 2x DB-15, each with 5 opto-isolated inputs and 5 outputs

## Network

- 2x 100Base-T connections, presented on RJ-45

# Audio:

## Analog Line Inputs:

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

## Analog Line Outputs:

- Output Source Impedance: <50 Ohms balanced
- Output Load Impedance: 600 Ohms, minimum
- Nominal Output Level: +4 dBu
- Maximum Output Level: +24 dBu

## Digital Audio Inputs And Outputs

- Reference Level: +4 dBu (-20 dB FSD)
- Impedance: 110 Ohm, balanced
- Signal Format: AES3 (AES/EBU)
- AES3 Input Compliance: 24-bit with sample rate conversion
- AES3 Output Compliance: 24-bit
- Digital Reference: Internal (network timebase) or external reference 48 kHz, +/- 2 ppm
- Internal Sampling Rate: 48 kHz
- Input Sample Rate: 32 kHz to 192 kHz

# SPECIFICATIONS

- Output Sample Rate: 44.1 kHz or 48 kHz
- A/D Conversions: 24-bit, Delta-Sigma, 256x oversampling
- D/A Conversions: 24-bit, Delta-Sigma, 256x oversampling

## Frequency Response

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

## Dynamic Range

- Analog Inputs to Analog Outputs 108dB referenced to 0dBFs, 111dB A-weighted
- Analog Inputs to Digital Outputs 110dB referenced to 0dBFs, 113dB A-weighted
- Digital Inputs to Analog Outputs 112dB referenced to 0dBFs, 115dB A-weighted
- Digital Inputs to Digital Outputs 126dB

## Total Harmonic Distortion + Noise

- Analog Input to Analog Output: < 0.005%, 1 kHz, +18dBu input, +18dBu output
- Analog Input to Digital Output: < 0.004%, 1 kHz, +18dBu input, -6dBFs output
- Digital Input to Analog Output: < 0.004%, 1 kHz, -6dBFs input, +18dBu output
- Digital Input to Digital Output: < 0.0003%, 1 kHz, -20dBFs

## Crosstalk Isolation, Stereo Separation And Cmrr

- Analog Line channel to channel isolation: 90dB minimum, 20Hz to 20kHz
- Analog Line stereo separation: 85dB minimum, 20Hz to 20kHz
- Analog Line Input CMRR: 80dB minimum, 20Hz to 20kHz

## Power Supply Ac Input

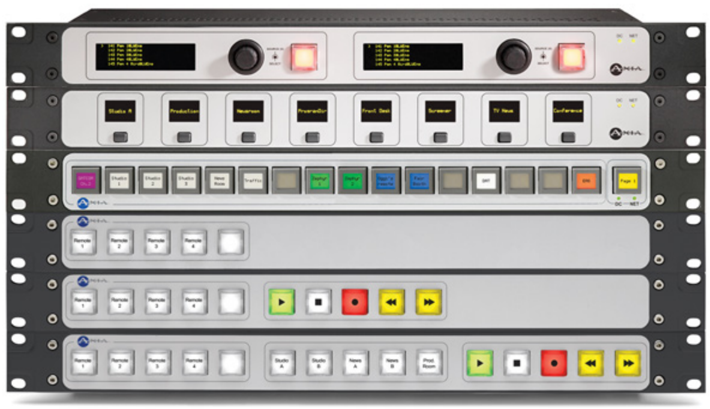
- Auto-ranging supply, 90VAC to 240VAC, 30 Hz to 60 Hz, IEC receptacle, internal fuse
- Power consumption: 35 Watts

## Operating Temperatures

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

# AXIA ROUTING CONTROL PANELS

## Fingertip control, just where you need it.



## OVERVIEW

Axia makes it easy to build large, IP-based routing networks of up to 10,000 audio streams. We also make hardware tools to help your operators control all that networking power. Along with PathfinderPC and PathfinderPRO routing control software, Axia Routing Control Panels put fingertip control of inputs, outputs and routing scene changes anywhere you've got a rack space.

# FEATURES

- Six accessory control panels for convenient talent / guest control of a variety of studio operations: routing scene changes, GPIO closures, XY control of inputs and outputs, etc.
- Slim panel design mounts in any 1RU rack space.
- Fanless, convection-cooled.

# IN DEPTH

## Take Control.

Axia control panels let you place routing power anywhere — in a studio turret, a TOC control panel or an equipment rack. These accessory control panels work with Axia's PathfinderPC and PathfinderPRO routing control hardware, allowing you to map routing commands – from simple contact closures to complex logic-driven events – to any button for fast execution.

Film-cap controllers with LED-backlit keys can be illuminated with a choice of colors; keycaps are film-legendable for quick function identification. SmartSwitch panels have dynamic, backlit LCD buttons that can change color and text with user activation. And the rack-mount 8-button SoftSwitch panel has high-resolution OLED buttons that can be loaded with user-created bitmaps for instant function identification. And the XY Routing Control Panel allows convenient on-the-fly routing of networked sources from anywhere in your facility; route any source to any output of your choosing with just a couple of knob twists.

### 17-Button LCD SmartSwitch Panel

The 17-button SmartSwitch Router Control Panel features backlit LCD buttons with dynamic text and color to provide 1-touch remote access to often-used machine-control or software functions. Multiple pages of button assignments can be programmed and recalled with just a touch; use PathfinderPC's Stacking Events Editor to map single commands or complex routing salvos to any button. Easy-to-use Web-based configuration pages can be accessed from any PC on the Livewire network.

# IN DEPTH

## Film-Cap Switch Panels

Use these Film-Cap Router Control Panels when dynamic-text capabilities are not required; lighted aircraft-grade switches provide fast execution of router salvos, machine-control or software functions programmed using PathfinderPC Router Control software. 5-, 10- and 15-button rackmount models are perfect for use in a studio turret, TOC control panel or equipment rack. Place film labels under the clear button caps; set the LED backlights to any of 8 different colors.

## OLED SoftSwitch Router Control Panel

The 8-Button OLED SoftSwitch provides high-visibility router control from any studio turret or equipment rack. Its eight bright, sharp OLED (Organic Light-Emitting Diode) readouts can display simple text or user-supplied monochrome bitmaps, and can be seen from nearly any angle — across the table, or across the room. Use PathfinderPC to program custom routing commands you can invoke instantly with the touch of a button.

## XY Router Control Panel

XY Router Control Panel lets you route any source to any destination (any-to-any routing) with the click of a button. Choose your desired audio stream, select your network output and press "Take" to route audio. Perfect for TOC program stream selection, ingest stations where a multitude of incoming feeds need routing to air, or production rooms — anywhere you need many-to-one control of networked audio streams.

# SPECIFICATIONS

## General

- Rackmount package requires 1RU of free rack space.
- All Router Control Panels require 1 free 100Base-T Ethernet port on a network switch for connection to the Axia network.
- 17-button SmartSwitch Panel requires PathfinderPC or PathfinderPRO software to program and execute conditional routing commands.
- FilmCap Button Panels require 1 free Axia GPIO port per each 5 buttons. PathfinderPC or PathfinderPRO software is not required for GPIO command of networked devices, but is required to program and execute routing commands.
- 8-Button OLED SoftSwitch requires PathfinderPC or PathfinderPRO software to program and execute conditional routing commands.

# AXIA STUDIO CONTROL PANELS

## Give Your Talent The Power


## OVERVIEW

Axia Studio Control Panels are a family of options panels designed for flush-mounting in desktop or turret cabinetry. They allow you to place control of headphone source selection, mic off/on control and even GPIO machine control at talent and guest desk positions, where they're most convenient.

# FEATURES

- Six accessory control panels for convenient talent / guest control of frequently-changed options, including headphone and mic control, GPIO closures, routing scene changes tied to Axia PathfinderPC software, Talkback to CR board op or Guest postions.
- Easy RJ-45 connection to console CANBus control network.
- May be flush-mounted in any flat or vertical solid surface.
- All panels measure 6" x 2"; require 2" mounting depth.

# IN DEPTH

## Options are just a touch away.

Axia consoles are nearly synonymous with "flexibility." You can save show settings and recall them in an instant... customize backfeeds and routing salvos... share audio sources and control throughout your facility... and that's just the beginning. Axia helps you customize your studio too, with accessory control panels that work seamlessly with your consoles to give talent fast access to headphone, mic and select switching controls.

### Mic Control Panel


The Mic Control panel gives talent or guests remote control of their mic channel. Press the Talkback key, and you open a comm channel to the board operator. There's a handy Mute key for those "frog-in-the-throat" moments, too. Works with all Axia consoles.

# IN DEPTH

## Producer's Mic Control Panel


Designed especially to suit the needs of busy talk show producers, the Producer's Mic Control panel provides control of microphone On/Off/Mute functions, and includes two special Talkback keys so producers can easily converse with studio remote talent. Works with all Axia consoles.

## Headphone Selector Panel


The Headphone Selector panel lets talent control their own headphone feeds. Turn the knob and control the volume. Push the knob, scroll through the list of available sources, and push again to "take." Preset buttons are provided for instant access to two programmed sources. Works with Element consoles.

# IN DEPTH

## Mic Control / Headphone Selector Panel


Why choose when you can have it all? Combination Mic Control/Headphone Selector panel gives talent remote control of headphone source and volume, mic channel on/off, and includes Mute and Talkback functions. Works with Element consoles.

## Five-Key Filmcap Button Panel


Five-key Button Panel can be placed wherever remote control of contact closures or routing commands

# IN DEPTH

is desired. Film-legendable keys contain LED backlights with individual color settings, and work with Pathfinder routing control software to put fingertip control right where it's needed. Works with Element consoles.

### Four-Key SmartSwitch Button Panel


Four-key SmartSwitch has illuminated, dynamic LCD keys that can change text and backlight color based on conditional logic macros you construct in Pathfinder routing control software, using simple drop-down tools. Works with Element consoles.

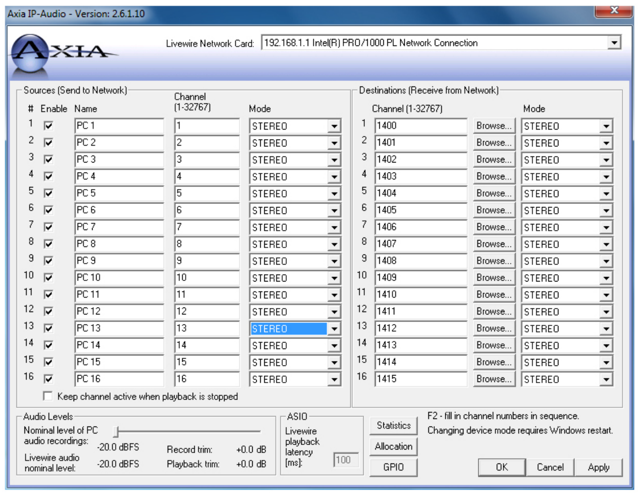
# SPECIFICATIONS

## General

- Desktop panels require access to Axia CANBus control network via CAT-5 connection.
- Flush-desktop mounting style requires routed 6″ x 2″ cutout in countertop or work surface. 2″ of space required behind each panel for adequate connector/cable clearance.
- Not all panels work with all Axia consoles. Consult Axia or your Axia representative for specific applications.

# AXIA IP-AUDIO DRIVER

## Pure Digital Audio from Networked PCs




## OVERVIEW

The Axia IP-Audio Driver lets you send and record single or multiple channels of stereo PC audio directly to and from Axia networks via Ethernet — no sound cards needed. Up to 24 channels of stereo audio can be sent simultaneously over a single CAT-5 Ethernet connection.

# FEATURES

- Sends audio sources to the Livewire network from PC/Windows audio applications such as multichannel delivery systems and other audio players.
- Receives audio from the Livewire network to destinations on the PC/Windows system, such as audio recording applications.
- GPIO function conveys “button-press” data from the Livewire network to destination applications; i.e., a console fader start button can command a PC/Windows-based audio player to start playback.
- The Axia IP-Audio Driver single-stream version emulates a standard sound card, with one stereo audio output device and one stereo audio input device. This version is suitable for typical two-channel (stereo) playback or recording applications.
- Axia IP-Audio Multichannel OEM versions emulate 4, 8 or 24 stereo sound cards (depending upon installed version), with one stereo audio output device and one stereo audio input device per “sound card”. These versions are intended for more complex professional applications.
- Supports 5.1 Surround audio streams as well as stereo, configurable on a per-stream basis.
- Windows version includes WDM and ASIO versions for maximum system flexibility.

# IN DEPTH

## Pristine PC Audio: No Sound Card Required

Way back when enormous cart machines still roamed the earth freely, we used XLR connectors to get recorded audio into the console. But when PCs replaced the cart machine, we — continued to connect to their sound cards with plain-Jane XLRs and a thick bundle of discrete wires that can’t carry logic, PAD or any of the useful information that PC playout systems provide. Why? With the Axia IP-Audio Driver, there’s a better way.

The PC is the heart of the modern radio studio. And Axia makes it easy to connect and exchange pristine digital audio with it. The Axia IP-Audio Driver for Windows is a special Windows driver that feeds your digital audio directly from your PC’s Ethernet port, through the Livewire network, to the Axia network, where the WAV-to-IP Audio conversion is performed. Up to 24 stereo playback channels and 24 stereo record channels can be accessed using our multi-stream driver that’s provided by your favorite digital delivery system provider; a single-play/single-record version is available for audio workstations.

# IN DEPTH

The IP-Audio Driver also provides GPIO-like start/stop and other control functions over the same network. It's available with the latest versions of high-end Windows audio delivery and editing software applications such as those from BSI, Burli, DAVID Systems, Dalet, ENCO, iMediaTouch, Netia, RCS, WideOrbit, and Zenon Media (to name just a few) and for Linux-based Rivendell through Paravel Systems — more than 20 systems and counting.

The Windows version of the IP-Audio Driver is available to broadcasters directly from Axia in 1-Stream and 4-Stream versions, and from Axia Delivery System Partners in 8-Stream and 24-Stream versions. Linux versions are available from our partner, Paravel Systems. For a full listing of Axia Delivery System Partners, visit www.AxiaAudio.com/partners/ .

# SPECIFICATIONS

## Microsoft Windows™ Operating System Requirements

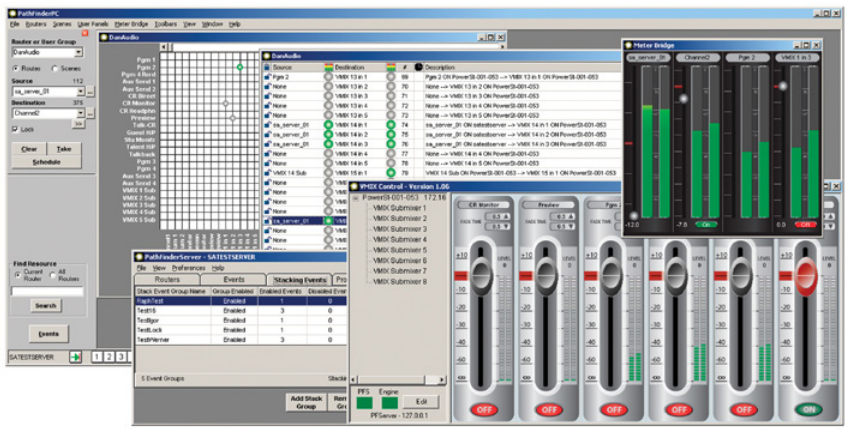
- Windows 2000 Professional
- Windows XP Pro (32-bit, standard)
- Windows XP Home Editions
- Windows 2003 Server R2
- Windows Vista (32- and 64-bit editions)
- Windows 7 and Windows 7 Pro (32- and 64-bit editions)
- Windows 8
- Minimum hardware requirements specified for your Windows operating system are sufficient to run the Axia IP-Audio Driver.

## Linux Operating System Requirements

- The Axia IP-Audio Driver for Linux is sold exclusively through Paravel Systems. Please contact them at www.paravelsystems.com/contact.html

# PATHFINDER ROUTING SOFTWARE

## Routing Automation for Axia Networks



## OVERVIEW

Axia's PathfinderPC and PathfinderPRO router control software for Windows is an amazingly rich set of tools you can use to customize and command your entire Axia network, allowing you to craft extremely sophisticated routing functions. Define automated switching events, construct custom software control panels, change between presets manually, on a daypart schedule, or via an external trigger. Pathfinder's advanced features include the ability to sense silence at a particular audio port and patch around it automatically — and even send the engineer an e-mail notification. And that's just the start.

Designed for automated routing control in small to medium-sized facilities, PathfinderPC provides a central point of control, via IP, of up to 25 Axia devices in your plant. Capabilities include route or scene changes based on scheduled events, GPIO closure or Silence Detect trigger events.

PathfinderPRO is the enterprise version of PathfinderPC. It contains all features found in PathfinderPC plus additional capabilities tailored to facilities with large physical plants or complex operational requirements. PathfinderPRO Controls an unlimited number of Axia devices and supports unlimited PathfinderPC client or PathfinderPC Mini connections plus direct Pathfinder control of motorized console faders, VMix Virtual Mixers and more.  Includes two server licenses for backup server or server clustering.

# FEATURES

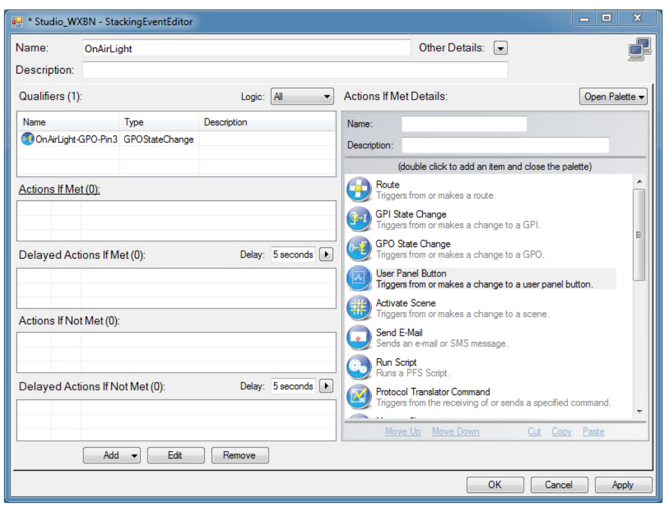
- Provides central routing interface for entire Axia networks. Pathfinder presents all Axia nodes and equipment and as a traditional single router, so you don't need to jump from place to place to see and manipulate your facility's routing infrastructure.
- Provides scheduling for one-time or regularly occurring routing events. These events can trigger route changes and/or GPIO events, from small-scale changes to system-wide "scene changes".
- Provides silence detection events with email and GPIO warnings, and automatically switches to an alternate pre-programmed route in the event of an audio failure. Includes audio presence meters in the onscreen routing matrix for instant visual audio confirmation.
- Pathfinder Stack Events allow you to design logic as complicated or as simple as you need, using simple or complex Boolean based logic events. So if you need to route a specific satellite feed to air only on Mondays, when a certain audio route exists, and the operator is holding down the blue button, you can.
- Panel Designer allows you to drop controls into your own software routing panel, then deploy it on studio PCs for users to select routing changes, monitor silence detection, and a whole host of other functions.
- Interfaces routing control with Axia console User Keys. Map custom-designed features to buttons mounted right in the console; depending upon console equipment, these buttons can even be programmed to dynamically change color and text to display status and engage actions.
- Built-in Protocol Translator, a part of PathfinderPRO software, allows your legacy systems to think they are talking to a router they understand — while behind the scenes, it's really Axia. Currently supported third party protocols include Pro-Bel General Router and General Switcher Protocols, Sine Systems ACU-1 Protocol and BTools protocols.
- Clustering support with Pathfinder PRO allows deployment of multiple PathfinderPRO servers which automatically monitor each other for backup and redundancy. If the primary server is unavailable, the clustered backup takes over, and all the client screens in the studios follow.
- Comprehensive logging capabilities include every route change, GPIO change, user button press, and more. Logging for each of these features may be enabled or disabled individually.

# IN DEPTH

## Power and Flexibility, At Your Fingertips

Power is good — but only if you can control it. Axia's Pathfinder family of router control tools let you customize and command your entire Axia network. Using your choice of graphical software or networked appliance, you can easily build extremely sophisticated routing functions like automated events, custom on-screen control panels — even change the entire network on a timed schedule if you like. Pathfinder can even give you peace of mind, by sensing silence on critical paths and patching around it automatically — then sending you an e-mail to let you know what happened. And that's just the start. Pathfinder can keep automatic logs of your studio network's routing operations — route changes, GPIO changes, user button presses, and more. Create sophisticated routing "scenes" with Boolean logic that automatically watch for and react to specified events, using a unique graphical editor that eliminates tedious script writing. Pathfinder Panel Designer even lets you construct custom on-screen controls that can be deployed on PCs across your network. Or, map custom features to rack-mounted button panels and user keys mounted right in the console.

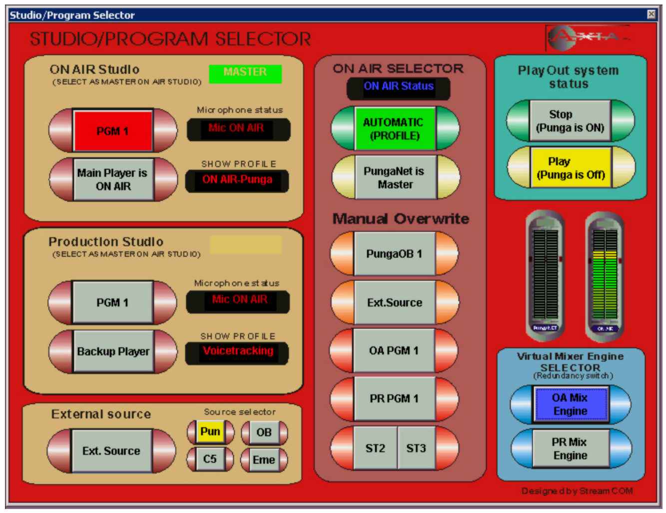
In today's broadcast environment, information is key. So Pathfinder allows you to keep logs of your studio network's routing operations — route changes, GPIO changes, user button presses, and much, much more.




You want to design that perfect automated system? Pathfinder gives you the tools. Pathfinder Stack Events allow you to design logic as simple (or sophisticated) as you need. An enhanced, graphical editor

# IN DEPTH

eliminates tedious script writing, allowing you to create sophisticated routing "scenes" with Boolean logic that automatically watch for and react to specified events.

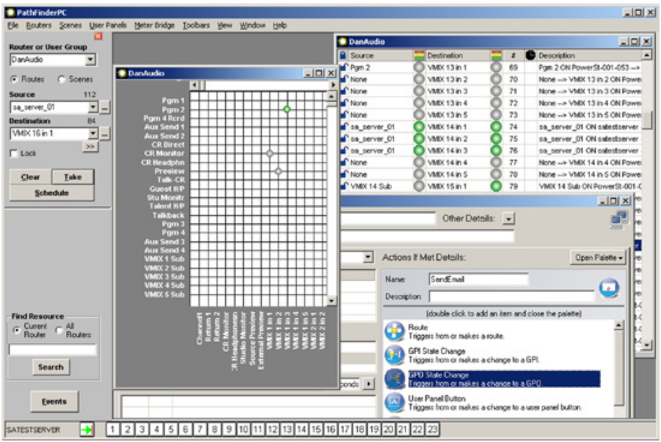



Pathfinder's Panel Designer applet lets you construct custom on-screen controls that can be deployed on PCs across your network. Or, map your custom designed features to rack-mounted button panels and user keys mounted right in the console. Some of these buttons can even dynamically change color and text to display status and engage actions.

There are two different Pathfinder software offerings tailored to your specific needs. Read on to find out which is right for you.

# IN DEPTH

## PathfinderPC Software



Designed for automated routing in small to medium-sized facilities, PathfinderPC gives you networked control of up to 25 Axia devices. This full-featured system runs on Windows PCs and allows you to construct and execute route or scene changes based on scheduled events, GPIO closures or Silence Detect trigger events. Using the client application, you can log in and change routing from anywhere you have network or Internet access. Use PathfinderPC to attach events to Axia SmartSwitch, SoftSwitch and Film-Cap button panels, or construct on-screen "virtual" controls that can run simultaneously on up to 10 PCs.

## PathfinderPRO Software

PathfinderPRO, the enterprise version of Pathfinder, contains all of the features found in PathfinderPC plus additional capabilities tailored to facilities with large physical plants or complex operational

# IN DEPTH

requirements. PathfinderPRO supports server "clustering" – running simultaneously on two connected, yet independent computers – for the ultimate in redundancy and security.

PathfinderPRO catalogs all of your Axia devices and supports as many end-user connections as your CPU can handle. PathfinderPRO can directly control console VMix virtual mixers, motorized faders on consoles so equipped, Show Profile changes, and more.

But PathfinderPRO doesn't stop at just controlling your Axia equipment. Complete delivery system integration is at your fingertips with Sine Systems ACU-1, Pro-Bel and BTools protocol emulators, plus support for routing and translating of serial, TCP and UDP ports. Snap-in real-time metering and Web browser controls provide added options for user-designed software panels. Browser controls even support multimedia audio and video, allowing embedded A/V streaming displays in software mini-panels.

# SPECIFICATIONS

## PathfinderPC Server

### Hardware

- Minimum hardware requirements specified for Windows XP, 2003 Server, 2008 Server, and Windows 7 are also acceptable to run PathfinderPC Client, PathfinderPC Mini, Panel Designer, SAPortRouter, VMIXControl, and the bridge application programs.

### Software

- Windows XP, 2003 Server, 2008 Server, or Windows 7. Microsoft .NET 3.5 SP1 is also required. Additionally, Windows 7 and 2008 require that the startup links be set to "Run as Administrator" in the compatibility frame.

## PathfinderPRO Server

### Hardware

- Minimum 512 Mb RAM, If the clustering option is used, minimum two NIC cards should be used, four are recommended.

# SPECIFICATIONS

### Software

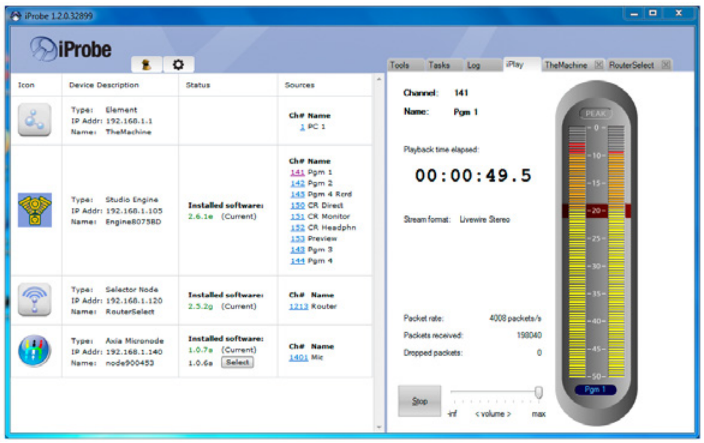
- Windows XP, 2003 Server, 2008 Server, and Windows 7. However, installations using more than 10 clients (PathfinderPC Client, PathfinderPC Mini) will require a server operating system such as Windows 2003 server or Windows 2008 Server.

## PC Client Applications

- Windows XP or any workstation operating system.

# AXIA IPROBE

## IP-Audio Network Management Software



## OVERVIEW

iProbe software helps with management, updating, and remote control of any Axia system. It has a powerful auto-documentation feature that generates configuration docs for every device, an Organizer that allows grouping networked audio devices into logical groups for easy management, facilitates uploading software to single or multiple devices, makes device configuration backups and more.

# FEATURES

- Discovery: the ability to scan your Axia Livewire network for Control Surfaces, Nodes (AES, Analog, GPIO), and Mixing Engines, as well as any Livewire devices from Axia Hardware Partners.
- Displays current firmware versions running on the all connected devices and gives you the ability to update firmware remotely, one device at a time or in logical groups of similar devices.
- Displays all the devices in your Livewire system and allows you to browse directly to a selected device's Web-based remote control interface. There is no need to type the device IP address into your browser.
- Complete configuration backup capabilities of individual devices, or all devices within your Livewire system.
- Integrated Syslog server for automated event logging.
- Auto-Documentation feature exports complete system data to a format of your choice for secure backup.
- Built-in iPlay module allows you to listen instantly to any channel on your network, and verify the levels of a given source.

# IN DEPTH

## Easy Network-Wide Backup, Update and Documentation

Axia iProbe is an intelligent network maintenance and diagnostics suite that consolidates managing, updating, and remotely-controlling your Axia system into one easy-to-use software application.

Axia networked audio devices are managed using a standard Web browser to view, configure, and administer each device. iProbe helps simplify this process by scanning and collecting all the information and presenting it a graphical interface. Along with this convenient central point of control, iProbe gives you powerful system tools like the Organizer, which performs advanced tasks such as gathering Livewire-enabled devices into logical groups for easy management and single-point administration of group settings. iProbe also helps with software version control, making it simple to upload software to single or multiple devices, back up device configuration, and more.

There's a powerful Auto-Documentation feature that queries and documents configuration settings for every networked Axia device — essential for administering large networks. Auto-Doc gives you the ability to export your Axia system data into an HTML format or text for printing a hard copy of your

# IN DEPTH

system configuration, or constructing a web page for future reference. You can also export to a tab-delimited text format for importing into other documents or spreadsheets. Exporting in XML format to other applications is also available.

And of course, there's a System Backup / Restore function that generates full-system backups which can be used to restore your Axia network from bare metal, should the need arise.

Even with all this power, iProbe is simple to use. An intuitive graphical interface lets you browse a list of similar devices, or click one-at-a-time on individual devices to make inspections or adjustments. You can even listen to individual sources with the integrated iPlay functions, and check on audio levels of any audio source.

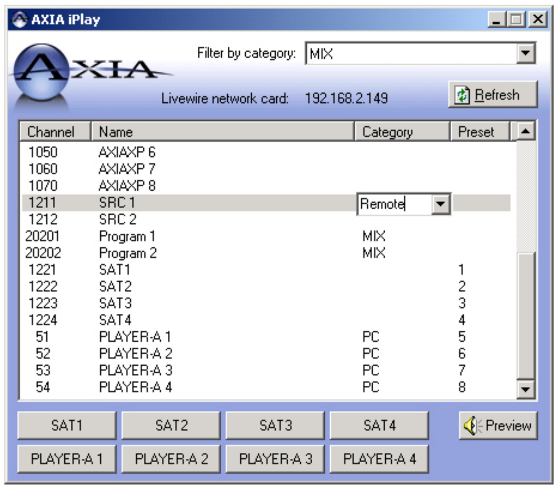
# SPECIFICATIONS

## System Requirements

- Windows 2000 Professional, Windows XP Professional, Windows Vista (32- and 64-bit editions), Windows 7 and Windows 7 Pro (32- and 64-bit editions), or Windows 8 operating system.
- 100Base-T or higher wired network adapter.
- Internet access to enable device firmware downloads.

# AXIA IPLAY

## Network Stream Player for Windows




# OVERVIEW

Software-based IP-Audio monitoring program lets Windows PC users select and listen to any audio source available to their Axia network. Choose from a complete list of available streams; eight user-programmable preset buttons provide quick access to frequently-accessed channels. On-screen level display meters auditioned audio.

# FEATURES

- Allows listening to any Axia network audio stream using standard PC sound card/speaker combo or headphones.
- Automatic detection of Livewire audio sources from connected PCs.
- User friendly interface allows filtering and sorting Livewire channels, making it easy to navigate in big systems containing hundreds or thousands of Livewire channels.
- Preset buttons that allow quick access to eight pre-selected channels.
- Administrator can restrict access to a set list of audio channels using built-in Access Control Lists.

# IN DEPTH

## Turn Any PC Into a Listening Station.

Remember the days when giving your Sales Manager a listening station meant running cable through the ceiling, installing a selector panel in the office wall, and mounting speakers in the drop tiles? And then, he could only hear a limited number of the audio channels your plant produced.

Axia iPlay PC software does away with old-fashioned speaker wire and rotary selectors. It allows any Windows PC to listen to streamed audio directly from your Axia network — any streamed audio. Not just Program feeds, but satellite downlinks, remote hosts, news production studios, interview rooms, etc. iPlay lets you give PC monitoring capabilities to PDs, GMs and sales staff using their existing computers, with no special wiring required. Just connect their PC's NIC to your Axia network, install iPlay, and presto! Every PC is a listening station. There's even an on-screen level display that meters the audio you're listening to — great for use as confidence meters for PDs or production personnel.

Users can choose from a list of all available audio, but in big plants that can be a lot to sift through. Not to worry: they can pre-set their favorite channels on the eight user-programmable preset buttons to get quick access to the streams they listen to most. And of course, you can filter out raw mic channels or other selected audio streams to prevent unauthorized listening.

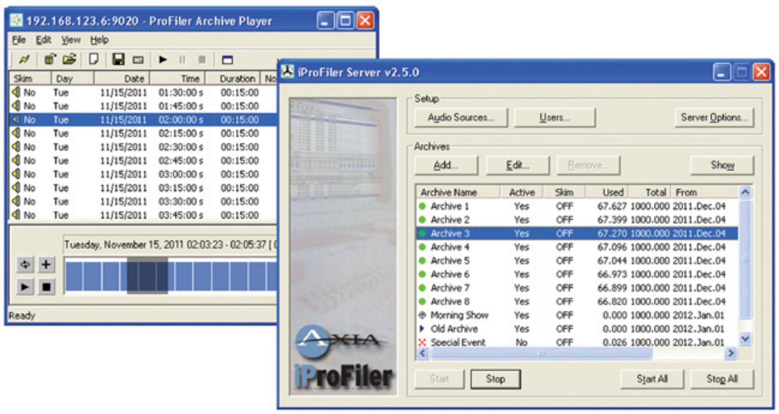
# SPECIFICATIONS

## System Requirements

- Windows 2000 Professional, Windows XP Professional, Windows Vista (32- and 64-bit editions), Windows 7 and Windows 7 Pro (32- and 64-bit editions), or Windows 8 operating system.
- 100Base-T or higher wired network adapter.

# IPROFILER

## Automated Program Archiving




## OVERVIEW

Axia's popular iProFiler logging software lets you simultaneously capture up to 24 stereo audio channels to time-stamped MP3 audio logs directly from your Axia IP-Audio network — no audio cards required. Included software records, manages and plays back archived audio files. Recording software runs under Windows XP and later; playback software runs under Windows NT, Windows 2000, Windows 98 or Windows XP and later. Record mode can be set for logging, skimming, or combination of both. Logged audio may be auditioned remotely via LAN, WAN, or Internet.

# FEATURES

- Simultaneously captures up to 24 channels of stereo audio.
- Directly records Axia digital audio streams — no sound card needed.
- Archived audio can be auditioned remotely via LAN, WAN or the Internet.
- iProFiler Live Player streams audio over any IP connection as it's being encoded. Great for consultants or group PDs listening remotely.
- NTP Time Sync synchronizes log file timestamps with your house NTP server (if equipped).
- Choose your skimming mode: Logging (continuous archival storage of program material), Skimming (records only when talent mic is open), or SmartSkimming (low-bitrate logging switches to a user-specified higher bitrate for quality captures when talent is on-mic).
- No "spool-up" time: iProFiler buffers incoming audio so that you never lose a word - no matter how late talent opens the mic.
- Recorded audio is time-stamped and stored in easy-to-search 15 minute blocks for fast retrieval.
- Standard MP3 file format allows logged audio to be played back on any media player application. Play files in iProFiler Archive Player to view detailed time-of-day data and user annotations.
- Easily select & export audio segments to WAV files for external editing.
- Choose any standard MP3 bit rate - from 16kbps - 320kbps - for the quality/drive space ratio that best suits your needs.
- Encoded program segments can also be set to upload automatically to an external drive, network share or FTP site.
- Remote monitoring application lets you "check up" on iProFiler remotely using a LAN or Internet connection; monitors disk space & audio presence.

# IN DEPTH

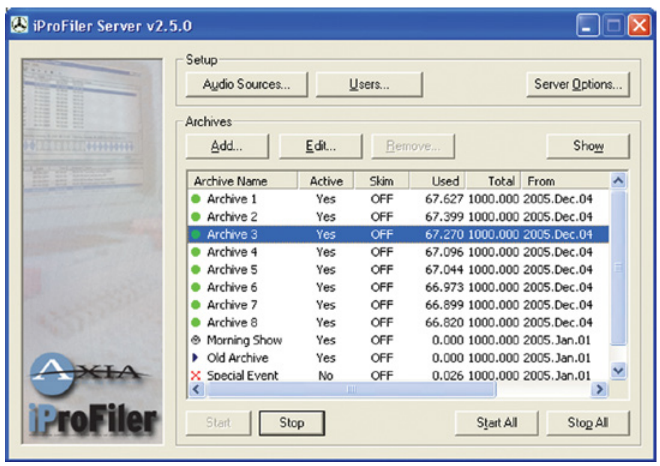
## Multi-channel, Multi-stream audio archiving for Axia Audio networks.

Sooner or later, someone's going to ask for a hard copy of a specific broadcast. Whether it's a client looking for proof of play, a Group PD that wants airchecks, or a listener claiming your morning show did something naughty, you'll need a record of your broadcast programming.

# IN DEPTH

Be prepared with Axia iProFiler, the award-winning audio archiving software that integrates with Axia IP-Audio networks to capture up to 24 simultaneous channels of Livewire Standard-stream stereo audio without sound cards. iProFiler uses the Axia IP-Audio Driver to exchange audio directly with Axia networks; just install iProFiler on a PC, connect the computer's NIC to the network with CAT-5, select the program streams you want to capture. It's as simple as that.

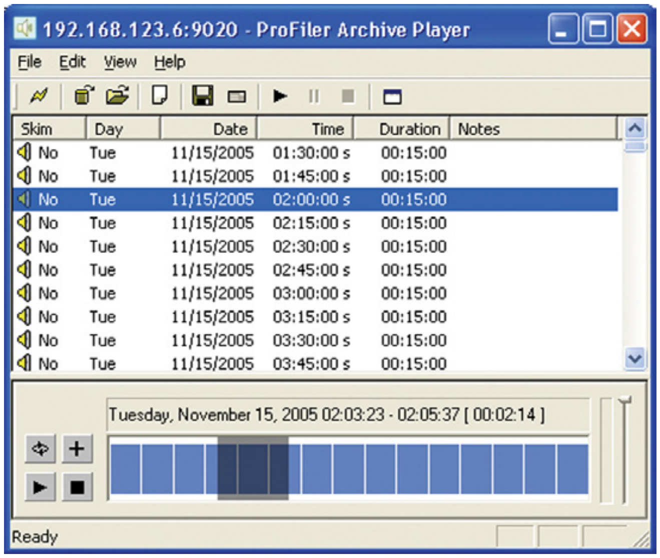
iProFiler's networked connectivity makes it the easiest logger to set up and operate, bar none. Just browse the audio streams available on your Axia network, select the ones you want to record, choose a bit rate for storage and off it goes.




And iProFiler is extremely flexible; you can continuously log program audio, automatically record telescoped talent airchecks, or record only what's broadcast when the mic isn't open. And iProFiler has "listen line" capability that lets you hear audio over your network (or the Internet) as it's being encoded - perfect for group PDs or consultants.

iProFiler's stored audio is networked, too. Any workstation or computer connected to your IP-Audio network can find and listen to time-stamped audio using a simple web-browser interface.

# IN DEPTH




iProFiler gives you a choice of operating modes for each archived audio stream:

- Choose "Logging" for continuous archival storage of program material, indefinitely (dependent upon storage space) or on a timed-record basis.
- Choose "Skimming" to record audio only when talent's mic is open, to capture live shows, call-in segments, talk shows or DJ bits. Program audio is pre-buffered so that there are no "up-cuts" upon record activation.
- Choose "SmartSkim" for a unique combination of skimming and logging. When talent mics are closed, ProFiler records audio in a low-bit rate logging mode, then switches to a higher bit rate for quality captures when talent is on-mic. All bit rates are user-selectable.

iProFiler is ideal for stations required by law to log program content, and since you can also listen to "live" audio over IP as it's being logged, it's great for Production Directors and morning show producers, program consultants or group PDs. Perfect for competitive monitoring, too — log other stations along with your own to fine-tune your formatics. An integrated audio browser lets your production crew tag segments and export them as WAV files for further editing, and logged shows can be automatically uploaded to FTP servers for storage or distribution.

# SPECIFICATIONS

## PC Hardware Minimum Requirements

- Pentium-IV, 2.4GHz processor or better with 512Mb RAM, 300 Gb free hard drive space, 100Base-T NIC.

## Operating System

- iProFiler Server: requires Windows XP or later. WAN/Internet connection required for remote monitoring.
- iProFiler Client: Requires Windows XP, Windows Vista, or Windows 7.

## Operating modes:

- Logging (continuous archival storage of program material)
- Skimming (records only when talent mic is open)
- SmartSkimming (low-bitrate logging switches to a user-specified higher bitrate for quality captures when talent is on-mic)
- Scheduled recording (date and time + length of program)

## Audio Interface

- 100Base-T or better Ethernet NIC with connection to Axia IP-Audio Network.
- Supports up to 24 stereo streams simultaneously.

## Audio Specifications

- Storage Format: MP3.
- Comproession Algorithm: Genuine Fraunhofer IIS
- Bit Rates Available: 8 kbps to 320 kbps, in standard increments
- Pre-roll and Post-roll Skim delay: up to 10 seconds, user-definable

# IP INTERCOM

## Go ahead: talk amongst yourselves.

## OVERVIEW

Axia IP Intercom is the only broadcast intercom system that takes advantage of the ease and efficiency of proven IP-Audio technology. Using a standard Ethernet backbone, IP Intercom saves on cost, space, and installation time, and eliminates special plug-in cards altogether.

The advantages of IP and Ethernet – low cost, easy installation and maintenance, efficient infrastructure – are well known. Installing IP Intercom is as simple as clicking together Ethernet gear! And of course it's easily scalable: plug as many stations into your switch as you want and add on from there. There's no expensive, hard-wired, custom-cable multi-pair infrastructure to deal with.

If you don't have an Axia studio network, IP Intercom can still help you save money, increase efficiency, and decrease the hard-wired infrastructure hassle It's a stand-alone system with I/O that will accommodate multiple mixing consoles. But if you do have an Axia system, you'll get seamless console integration that gives your operators benefits other systems can't, like the ability to take broadcast quality intercom audio directly to air, and feed IFB audio directly to intercom callers.

The IP Intercom system includes a variety of desktop and rackmount stations, a software Intercom application that turns any PC into an intercom station, and drop-in modules for popular Axia mixing consoles.

# FEATURES

- 100% digital system, end to end.
- Seamless integration between broadcast audio and communications channels. Full 20 Hz – 20 kHz audio response allows intercom channels to be taken to air with no degradation of sound quality
- Stand-alone rack-mount, desktop and integrated Axia console modules are available for a turnkey intercom installation.
- Program station presets and GPIO functions using any standard Web browser.
- Ethernet-based system has no central matrix or card-cage; is naturally scalable. Easily expand the number of intercom stations as your facility grows by simply plugging in new stations.
- Intercom keypad can also dial outside phone lines (using an optional telephone hybrid).
- Analog I/O presented on both XLR and StudioHub-compatible RJ-45 connectors.
- Front-panel locking connections accommodate popular mini-mics and headsets.
- Add PCs to the system with SoftCom Intercom Station for Windows.

# IN DEPTH

## Imagine a digital intercom system with no central matrix.

Actually, don't bother — we've built one. Axia IP Intercom saves on cost, space, and installation time, and eliminates special plug-in cards altogether. It's real plug and play that works every time — even when you need to add a station, or reconfigure the ones you've got.

Everybody knows the advantages of IP and Ethernet – low cost, easy installation and maintenance, efficient infrastructure. Thanks to its efficient Ethernet backbone, installing IP Intercom is a simple single-click connection. Of course it's easily scalable: plug as many stations into your switch as you want and add on from there. Then start talking! And if you move to a new location, you can just pick up the gear and take it with you — there's no expensive, hard-wired, custom-cable multi-pair infrastructure mess to
deal with.

Don't have an Axia studio network? That's OK. You'll still save money and increase efficiency by choosing IP-Intercom; it's a stand-alone system with I/O that will accommodate multiple consoles. But if you do

# IN DEPTH

have an Axia system, you'll get seamless console integration that gives your operators benefits other systems can't. For instance, you can take broadcast-quality intercom audio directly to air. And you can feed IFB audio directly to intercom callers.

IP Intercom gives you unlimited full-bandwidth access to any studio, news or sports venue, office, hallway, broom closet or wherever. Talk and listen to individuals or groups hands-free, with no echo or feedback — IP Intercom features exclusive AEC advanced echo cancellation from Fraunhofer Labs (the inventors of MP3), so there's never any open-mic feedback during conversations. Ever.

IP Intercom system is completely digital. Other intercom systems try to make you think they're digital by piping their analog signals over CAT-5 cables, but the last thing you need during a breaking story or transmitter failure is hum and buzz getting between you and the guy you need to talk to. With IP Intercom, there isn't any.

So you've gotta be a genius to use it, right? Actually, anyone with an index finger can operate this system with ease. The web interface makes setup simple. Sharp, high-contrast OLED displays are easy to read from anywhere in the room. And our clever callback feature makes sure you'll never miss a call, no matter what you're doing. There are also functions that allow talent to mute calls from other stations, to make sure there's never an interruption on-air.

IP Intercom comes in several rack-mount and desktop styles, plus drop-in modules for Axia Element consoles. And our unique SoftCom software lets you turn any connected PC into an intercom station! Just mix and match to build a system customized to your needs.

## Rackmount Stations

### IC.20 Rackmount Station

The IC.20 intercom panel features 20 station presets for quick contact with frequently-called stations. Perfect for Master Control or TOC, the IC.20 includes a keypad and associated display for fast access to stations system-wide, plus group talk and auto-answer functions. Keypad can also dial outside phone lines (using an optional telephone hybrid). 2RU rackmount package features high-visibility 10-character

# IN DEPTH

OLED (organic LED) displays, built-in speaker, front- and rear-panel mic connections, 4-pin locking headset jack, analog I/O presented on both XLR and StudioHub-compatible RJ-45 connectors, GPIO connection for speaker mute/dim and external line-status tallies, and an Ethernet jack for single-cable network connection.

## IC.10 10-Station Intercom Panel

The IC.10 is a 10-station version of the IC.20 we talked about earlier. It has 10 station presets with high-resolution OLED displays, a built-in speaker, front- and rear-panel mic connections, 4-pin locking headset jack, analog I/O on XLR and StudioHub-compatible RJ-45 connectors, GPIO connection for speaker mute/dim and external line-status tallies, and an Ethernet connection.

## IC.1 10-Station Intercom Panel

The IC.1 is a cost-effective way to add intercom capabilities to any studio. It features 10 LED-backlit film-cap buttons that are easily labeled with station names; like other IP Intercom station, programming is via Web interface. IC.1 has a built-in speaker and front-panel 4-pin locking headset jack, front- and rear-panel mic inputs, analog I/O with XLR and RJ-45 connectors, GPIO speaker mute/ dim control. An Ethernet jack completes the connection complement.

# IN DEPTH

## Desktop Stations

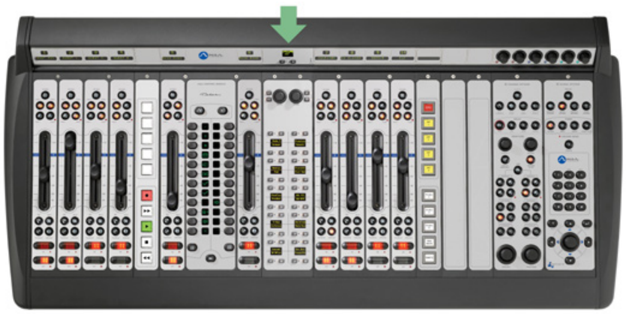
The IC.1D 20-station desktop intercom is perfect for producers, screeners, etc. IC.1D has 20 preset stations presented on LED-backlit button caps; an economical way to add intercom function to any space. 20 LED-backlit film-cap buttons can be labeled with station names and programmed using a built-in Web interface and any browser. The OLED callback window lets users identify and answer calls from remote stations that aren't programmed on a local "speed" key. IC.1D includes a built-in speaker and front-panel 4-pin locking headset jack. All it takes to add it to your intercom network is a single CAT-5 connected to the rear-panel Ethernet port; a built-in auto-sensing power supply eliminates nasty "wall warts."

### IC.20D 20-Station OLED Intercom Pane

The IC.20D is the desktop version of the IC.20 rack-mount station we showed you earlier. The 20 station preset locations are equipped with high-resolution OLED displays; the OLED callback window and dialing pad let operators call any station not programmed to a preset location. Naturally there's a built-in speaker, front-panel 4-pin locking headset jack, front-panel mic input, an Ethernet port for fast hookup, and internal auto-sensing power supply.

# IN DEPTH

## Console Modules



You don't need to own an Axia console to use IP Intercom — rack-mount and desktop stations integrate with any broadcast mixer to route intercom traffic to air instantly — full-bandwidth, broadcast-quality audio, not tin-can-and-string noise. But if you do own an Axia Fusion or Element mixing console, these drop-in modules make communications even easier by turning your board into an intercom station!

Built-in Talkback functions enable seamless communication between board ops, hosts and studio guests.

### 20 Station OLED Intercom Module

# IN DEPTH

The 20-Station OLED intercom module requires two frame positions and provides access to 20 pre-programmed intercom stations. Individual talk and listen buttons are combined with high-resolution OLED displays for fast access to frequently-called stations; auto-answer functions are also provided. Mic audio is taken directly from the console operator's microphone; speaker audio is directed to the console's preview speaker. There's a dedicated listen volume control, individual mic and speaker mute keys and group talk functions; the overbridge display works with the console's monitor module numeric keypad to give direct access to any station systemwide. Station presets and GPIO functions are programmed using any standard Web browser. Available for Element consoles..

## 10 Station OLED Intercom Module

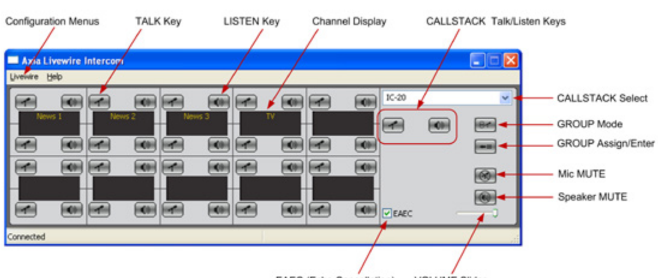
The 10-Station OLED intercom module occupies one console frame position and includes ten preset locations with 10-character OLED displays, auto-answer functions, dedicated listen volume control, and mute keys for speaker and mic. Available for Axia Element console only.

# IN DEPTH

## 10 Station Film-Cap Intercom Module


This economical 10-Station Film-Cap intercom module features ten LED-backlit film-cap buttons for single-button calling of up to 10 preset stations. This module occupies one frame position, and also provides a dedicated listen volume control, speaker and mic mute buttons. It uses a single frame position. Available for Element consoles..

## SoftCom IP Intercom for Windows




Axia Softcom Intercom for Windows makes any networked PC a part of your IP Intercom system! The easy user interface mimics the IC-20 control panel, with preset locations for 20 frequently-called

# IN DEPTH

stations. Auto-answer and hands-free functions are supported, and a drop-down station finder gives instant access to stations not pre-programmed. All your PC needs is a sound card with mic & speakers, and a 100Base-T Ethernet connection to your Axia IP-Audio network. Purchase includes a site license for all PCs.

# SPECIFICATIONS

Like all Axia products, IP Intercom uses only premium, studio-grade audio components to guarantee maximum performance.

## Microphone Preamplifiers

- Source Impedance: 150 ohms
- Input Impedance: 4 k ohms minimum, balanced
- Nominal Level Range: Adjustable, -75 dBu to -20 dBu
- Input Headroom: >20 dB above nominal input
- Output Level: +4 dBu, nominal

## Analog Line Inputs

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

## Analog Line Outputs

- Output Source Impedance: <50 ohms balanced
- Output Load Impedance: 600 ohms, minimum
- Nominal Output Level: +4 dBu
- Maximum Output Level: +24 dBu

# SPECIFICATIONS

## Frequency Response

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

## Dynamic Range

- Analog Input to Analog Output: 102 dB referenced to 0 dBFS, 105 dB "A" weighted to 0 dBFS
- Analog Input to Digital Output: 105 dB referenced to 0 dBFS
- Digital Input to Analog Output: 103 dB referenced to 0 dBFS, 106 dB "A" weighted
- Digital Input to Digital Output: 125 dB

## Equivalent Input Noise

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

## Total Harmonic Distortion + Noise

- Mic Pre Input to Analog Line Output: <0.005%, 1 kHz, -38 dBu input, +18 dBu output
- Analog Input to Analog Output: <0.008%, 1 kHz, +18 dBu input, +18 dBu output
- Digital Input to Digital Output: <0.0003%, 1 kHz, -20 dBFS
- Digital Input to Analog Output: <0.005%, 1 kHz, -6 dBFS input, +18 dBu output

## Crosstalk Isolation and CMRR

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

## Power Supply AC Input, rackmount and desktop stations

- Auto-sensing supply, 90VAC to 240VAC, 50 Hz to 60 Hz, IEC receptacle, internal fuse
- Power consumption: 35 Watts or less

# SPECIFICATIONS

## Operating Temperatures

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

## Dimensions

- IC.20: 3.5 inches x 19 inches x 8.5 inches, 5 pounds
- IC.10X: 1.75 inches x 19 inches x 8.5 inches, 4 pounds
- IC.10: 1.75 inches x 19 inches x 8.5 inches, 4 pounds
- IC.1: 1.75 inches x 19 inches x 8.5 inches, 4 pounds
- IC.20D: 18.25 inches x 6 inches x 5.75 inches, 6 pounds
- IC.1D: 13.5 inches x 8.5 inches x 4.5 inches, 6 pounds

## SoftCom PC Hardware Requirements

- Windows XP or higher
- 20MB free hard drive space
- 100Base-T Ethernet connection to Axia network
- Sound card and mic/earphone headset

# AERO.2000™

## Audio/Loudness Manager


## OVERVIEW

The Linear Acoustic AERO.2000 provides all of the features and functionality of the AERO.100, but takes things a step further. Handling up to 16 channels of baseband or encoded audio via both AES or HD/SD-SDI I/O, it offers AEROMAX® audio processing, upmixing, encoding, and metering for over the air, over the top, and mobile television broadcasts in a 2RU chassis with a front panel that includes a full-color screen, controls, and headphone output.

Base configuration includes a single processing instance in a 5.1+2+2 channel configuration, with or without Dolby® encoding and Nielsen® Watermark encoding. A second processing instance (5.1+2+2 or 2+2+2) with or without the Dolby and Nielsen options can be enabled at any time by simply entering a factory provided key.

Local audio insertion, SAP/DVS with downmix replacement, full-time downmix, and support for Audio Description (warble tone) is provided.

Failover bypass relays on all I/O maintain signal continuity and dual auto-ranging power supplies enable redundancy and worldwide compatibility.

Designed and built in the USA, the 2RU AERO.2000 combines performance, flexibility and convenience.

A full color front panel screen with rotary navigation cluster and headphone output allow for local set-up and adjustment.

# FEATURES

- Linear Acoustic AEROMAX loudness and dynamic range control
- UPMAX® II upmixing with AutoMAX-II™
- Intelligent Dynamics™ with advanced ITU (AI) Limiter
- Standard single processing instance in a 5.1+2+2 configuration
- Available second instance in a 5.1+2+2 or 2+2+2 configuration, with or without Dolby encoding and Nielsen Watermark encoding
- Automatic downmix output
- Support for SAP/DVS
- Local audio/voiceover insertion
- Optional Audio Description (warble tone)
- Standard Dolby® Digital (AC-3), Dolby Digital Plus and Dolby E decoding
- Available Dolby Digital (AC-3), Dolby Digital Plus, and Dolby E encoding
- 3GHz HD/SD-SDI I/O with included video delay (16 channels)
- AES I/O with reference input (16 channels)
- Dual PSU and relay bypass
- Extensive TCP/IP remote control and HTTP control

# IN DEPTH

## Comprehensive TCP/IP Remote

Provides control over all system settings, processing and coding parameters plus extensive metering of signal presence, processing and coding activity, and audio loudness. System status reports physical I/O details along with system, power supply and environmental health. The remote application also delivers remote audio, including 5.1 channels, to the user so that signal quality can be auditioned anywhere link bandwidth permits. The built-in HTTP server provides for simple get/set control of all parameters and status.

# IN DEPTH


## Nielsen Encoding

Generates revenue-critical Nielsen NAES II and the new Nielsen Watermarks audience measurement codes. AERO.2000 precisely inserts these signals for maximum code recovery – after audio decoding and processing and before transmission encoding.

## Linear Acoustic Intelligent Dynamics

### The Basics

Linear Acoustic Intelligent Dynamics hybrid metadata processing is a patented hybrid of traditional multiband techniques and metadata. Because it is created during transmission encoding, this metadata requires no operator intervention or special tools and is a new version of the DRC part of the Dolby Digital encoder that has always been there. Except it is now effective and uncomplicated.

Featured at no additional cost, Intelligent Dynamics is actually a combination of two revolutionary technologies.

### How it Works

The first portion of Intelligent Dynamics is the Emmy® award-winning process developed by Linear Acoustic which generates new metadata dynamic range control via traditional processing techniques. This control can be set as permanent, reversible, or anywhere in between.  This technology is already present in every AERO.100, AERO.1000, AERO.2000, AERO.asi and AERO.soft product.

The second portion of the process involves verifying and marking content as compliant upstream. This is accomplished using a portion of the new Dolby Intelligent Media Framework called Evolution which enables measurement and incorporation of authenticated loudness data within the audio at each stage of the content delivery chain from production onwards. Compliant programming can be passed through with minimal or no additional processing, while content that cannot be verified can still be made compliant.

# IN DEPTH

Most current television audio processors control loudness by managing dynamic range in real time. This effectively manages "boundary issues" such as commercials that follow a quiet program segment, but also compromises the impact and excitement of intentionally dynamic scenes because all programming gets some degree of permanent correction.

Audio transmission formats such as Dolby Digital (AC-3) and Dolby Digital Plus (E-AC-3) already incorporate metadata dynamic range control (DRC) profiles, which can provide some control, but these profiles are not ideally suited for broadcast audio and often result in significant over and under processing if other metadata parameters are not correct.

Intelligent Dynamics overcomes these compromises and enables audio to be processed to the degree and permanence dictated by the programming itself. The content effectively and automatically controls the processing and eliminates the need for further downstream changes that could irreparably change it.

### The Result

Consumers can now enjoy the benefits of audio that is tailored to their liking using existing Dolby Digital and Dolby Digital Plus decoders. From the default of controlled dynamic range for noisy environments and small television speakers to full dynamic range for well-produced programming and multi-channel home theater systems, Linear Acoustic Intelligent Dynamics delivers the best of both worlds without compromise.

# SPECIFICATIONS

## Processing

AEROMAX multistage adaptive wideband and multiband loudness and dynamic range control with ITU-R BS.1770 and EBU R128 loudness metering.

Dual UPMAX II two-channel to 5.1-channel upmixers plus main channel downmixing with automatic bypass of discrete content.

Linear Acoustic Intelligent Dynamics with Advanced ITU
(AI) Limiter

# SPECIFICATIONS

## Audio Encoding/Decoding

Standard Dolby Digital (AC-3), Dolby Digital Plus and Dolby E decoding

Available Dolby Digital (AC-3), Dolby Digital Plus, and Dolby E encoding

## Reference

48kHz via AES DARS (or any AES signal applied to the Ref In connector), AES In 1, SDI, or from the internal 48kHz clock (standalone use only).

## Sample Rate/Resolution/Frequency Response

48kHz, 24-bit, 20Hz to 20kHz below threshold

## AES I/O

Eight main inputs plus reference via 75-Ohm BNC female connectors. Eight main outputs plus encoder output. Eight additional channels of auxiliary digital I/O on DB-25 female connector. All digital inputs are 75 Ohm internally terminated, unbalanced. Signal levels per SMPTE 276M/ AES-3ID-2001.

## HD/SD-SDI I/O

Auto-sensing 3GHz HD/SD-SDI (SMPTE 292M/259M) inputs, up to 1080i/60/59.94/50Hz. De-embed up to 16 channels from applied SDI signal, process and/or encode, re-embed up to 16 channels. Supports SMPTE 2020 A and B VANC metadata.

## Headphone Output

1/4'' (6.35mm) front panel connector with volume control.
Parallel GPI/O Parallel Control Port
25-pin female D connector, 0-5V TTL levels for 8 inputs and 8 outputs; controls simple preset recalls plus voiceover/EAS insertion

## Serial Metadata Input

9-pin female D connector; 115.2 kbps; pinout per SMPTE 207M (RS-485); Designed to directly interface with Dolby serial metadata (SMPTE RDD6)

# SPECIFICATIONS

## Ethernet

Gigabit Ethernet via RJ45 supports included TCP/IP remote control application; HTTP server included for get/set control of all parameters.

## Remote Control

Windows®-compatible TCP/IP remote control Included application for full setup and control, ITU-R BS.1770 metering for all programs, encoder statistics, and return audio for remote monitoring (network speed permitting). HTTP server allows get/set control from PC Front Panel Controls and Indicators.

## Front Panel Controls

Rotary encoder and control keys plus color display and headphone output.

## Power Requirements

Dual redundant power supplies, each rated at 100-264 VAC, auto-sensing, 50/60 Hz, 175W each maximum

## Dimensions and Weight

2RU: 3.50"H x 19"W x 17"D (89mm X 483mm X 432mm)
Net weight: 13 lbs. (5.9 kg), approximate.

## Shipping Dimensions and Weight

22"W x 20"D x 9"H (559 x 508 x229 mm)
Net weight: 18 lbs. (8.2 kg), approximate.

## Environmental

Fan cooled. Operating: 0 to 50 degrees C, non-operating -20 to 70 degrees C

## Regulatory

North America: Tested to comply with the limits for a class A digital device pursuant to Part 15 of the FCC rules (CFR). Power supplies are UL tested and approved.

Europe: Tested to comply with the requirements and harmonised Low Voltage Directive 2006/95/EC and EMC Directive 2004/108/EC as indicated by the affixed CE marking; RoHS and WEEE compliant

## Warranty

Standard Telos Alliance 5-Year Warranty

# AERO.1000™
## Audio/Loudness Platform


## OVERVIEW

AERO.1000 is a fresh, revolutionary approach to balancing channel density, control, and quality. Award-winning loudness control tools plus extensive I/O in a flexible, expandable, high-density package make the AERO.1000 a wise investment.

AERO.1000 supports up to eight instances of processing in two flavors: 5.1+2+2 or 2+2+2. Additional instances can be added as needed via software key.

Handling up to 64 baseband or encoded audio channels via AES or HD/SD-SDI I/O, AERO.1000 offers extremely high density. Dolby® encoders and decoders can be optionally enabled for each input and output.

Local audio insertion, SAP/DVS with downmix replacement, full-time downmix, and support for Audio Description (warble tone) is provided.

Failover bypass relays on all I/O maintain signal continuity and dual auto-ranging power supplies enable redundancy and worldwide compatibility.

Designed and assembled in the USA, the lightweight and rugged AERO.1000 is a solid investment in performance and flexibility.

A bright yellow OLED display shows unit IP address and status for remote control while a front panel headphone output allows for local monitoring.

# FEATURES

- Linear Acoustic AEROMAX® loudness and dynamics control
- UPMAX® II upmixing with AutoMAX-II™
- Intelligent Dynamics™ with advanced ITU (AI) Limiter
- Up to eight available processing instances in a 5.1+2+2 or 2+2+2 configuration.
- Available Dolby® Digital (AC-3), Dolby Digital Plus, and Dolby E decoding (per instance).
- Available Dolby Digital (AC-3), Dolby Digital Plus, and Dolby E encoding (per instance)
- Available Nielsen® Watermark encoding (per instance)
- Automatic downmix output
- Support for SAP/DVS
- Local audio/voiceover insertion
- Optional Audio Description (warble tone)
- 3GHz HD/SD-SDI I/O with included video delay (16 channels)
- AES I/O with reference input (16 channels)
- Dual PSU and relay bypass
- Extensive TCP/IP remote control and HTTP control

# IN DEPTH

## Comprehensive TCP/IP Remote

Provides control over all system settings, processing and coding parameters plus extensive metering of signal presence, processing and coding activity, and audio loudness. System status reports physical I/O details along with system, power supply and environmental health. The remote application also delivers remote audio, including 5.1 channels, to the user so that signal quality can be auditioned anywhere link bandwidth permits. The built-in HTTP server provides for simple get/set control of all parameters and status.

# IN DEPTH

## Nielsen Watermark Encoding

Generates revenue-critical Nielsen NAES II and the new Nielsen Watermarks audience measurement codes. AERO.1000 precisely inserts these signals for maximum code recovery – after audio decoding and processing and before transmission encoding.

## Linear Acoustic Intelligent Dynamics

### The Basics

Linear Acoustic Intelligent Dynamics hybrid metadata processing is a patented hybrid of traditional multiband techniques and metadata. Because it is created during transmission encoding, this metadata requires no operator intervention or special tools and is a new version of the DRC part of the Dolby Digital encoder that has always been there. Except it is now effective and uncomplicated.

Featured at no additional cost, Intelligent Dynamics is actually a combination of two revolutionary technologies.

### How it Works

The first portion of Intelligent Dynamics is the Emmy® award-winning process developed by Linear Acoustic which generates new metadata dynamic range control via traditional processing techniques. This control can be set as permanent, reversible, or anywhere in between.  This technology is already present in every AERO.100, AERO.1000, AERO.2000, AERO.asi and AERO.soft product.

The second portion of the process involves verifying and marking content as compliant upstream. This is accomplished using a portion of the new Dolby Intelligent Media Framework called Evolution which enables measurement and incorporation of authenticated loudness data within the audio at each stage of the content delivery chain from production onwards. Compliant programming can be passed through with minimal or no additional processing, while content that cannot be verified can still be made compliant.

Most current television audio processors control loudness by managing dynamic range in real time. This effectively manages “boundary issues” such as commercials that follow a quiet program segment, but also compromises the impact and excitement of intentionally dynamic scenes because all programming gets some degree of permanent correction.

Audio transmission formats such as Dolby Digital (AC-3) and Dolby Digital Plus (E-AC-3) already incorporate metadata dynamic range control (DRC) profiles, which can provide some control, but these profiles are not ideally suited for broadcast audio and often result in significant over and under processing if other metadata parameters are not correct.

# IN DEPTH

Intelligent Dynamics overcomes these compromises and enables audio to be processed to the degree and permanence dictated by the programming itself. The content effectively and automatically controls the processing and eliminates the need for further downstream changes that could irreparably change it.

### The Result

Consumers can now enjoy the benefits of audio that is tailored to their liking using existing Dolby Digital and Dolby Digital Plus decoders. From the default of controlled dynamic range for noisy environments and small television speakers to full dynamic range for well-produced programming and multi-channel home theater systems, Linear Acoustic Intelligent Dynamics delivers the best of both worlds without compromise.

# SPECIFICATIONS

## Processing

AEROMAX multistage adaptive wideband and multiband loudness and dynamic range control with ITU-R BS.1770 loudness metering

UPMAX® II two-channel to 5.1 channel upmixing and downmixing automatically bypasses discrete content

Linear Acoustic Intelligent Dynamics with Advanced ITU (AI) Limiter

## Audio Encoding/Decoding

Available Dolby Digital (AC-3), Dolby Digital Plus and Dolby E decoding and encoding.

## Reference

48kHz via AES DARS (or any AES signal applied to the Ref In connector), AES In 1, SDI, or from the internal 48kHz clock (standalone use only).

## Sample Rate/Resolution/Frequency Response

48kHz, 24-bit, 20Hz to 20kHz below threshold

# SPECIFICATIONS

## AES I/O

Eight main inputs plus reference via 75-Ohm BNC female connectors. Eight main outputs plus encoder output. Eight additional channels of auxiliary digital I/O on DB-25 female connector. All digital inputs are 75 Ohm internally terminated, unbalanced. Signal levels per SMPTE 276M/ AES-3ID-2001.

## HD-SD SDI I/O

Auto-sensing 3GHz HD/SD-SDI (SMPTE 292M/259M) inputs, up to 1080i/60/59.94/50Hz. De-embed up to 16 channels from applied SDI signal, process and/or encode, re-embed up to 16 channels. Supports SMPTE 2020 A and B VANC metadata.

## Analog I/O

10K Ohm balanced stereo inputs, +4dBu nominal, +24dBu Max.; Balanced outputs +4dBu nominal, +24dBu Max into 600 Ohms.

## Headphone Output

1/4" (6.3mm) front panel connector, +12 dBu max into 600-Ohms

## Parallel GPI/O Control Port

25-pin female D connector, 0-5V TTL levels for 8 inputs and 8 outputs; controls simple preset recalls plus voiceover/EAS insertion

## Serial Metadata Input

9-pin female D connector; 115.2 kbps;; pinout per SMPTE 207M (RS-485); Designed to directly interface with Dolby serial metadata (SMPTE RDD6)

## Ethernet

Gigabit Ethernet via RJ45 supports included TCP/IP remote control application; HTTP server included for get/set control of all parameters

## Remote Control

Windows®-compatible TCP/IP remote control Included application for full setup and control, ITU-R BS.1770 metering for all programs, encoder statistics, and return audio for remote monitoring (network speed permitting). HTTP server allows get/set control from PC.

# SPECIFICATIONS

## Front Panel Controls and Indicators

Graphical OLED display and headphone output.

## Power Requirements

Dual power supplies, each rated at 100-264 VAC, 50/60Hz, auto-sensing, 150W max

## Dimensions and Weight

One rack unit- 1.75"H x 19"W x 15.5"D (44 x 483 x 394 mm)
Net weight: 9 lbs. (4 kg), approximate.

## Shipping Weight and Dimensions

22"W x 20"D x 7"H (559 x 508 x 178 mm)
Net weight: 15 lbs. (6.8 kg), approximate.

## Environmental

Fan cooled. Operating: 0 to 50 degrees C, non-operating -20 to 70 degrees C

## Regulatory

North America: Tested to comply with the limits for a class A digital device pursuant to Part 15 of the FCC rules (CFR). Power supplies are UL tested and approved.

Europe: Tested to comply with the requirements and harmonised Low Voltage Directive 2006/95/EC and EMC Directive 2004/108/EC as indicated by the affixed CE marking; RoHS and WEEE compliant

## Warranty

Standard Telos Alliance 5-Year Warranty

# AERO.100™
## DTV Audio Processor


## OVERVIEW

Handling up to 16 channels of baseband or encoded audio via AES (10 channels) or HD/SD-SDI (16 channels) I/O, the Linear Acoustic AERO.100 offers all-inclusive AEROMAX® audio processing, upmixing, encoding, and metering for over the air, over the top, and mobile television broadcasts in a compact 1RU chassis.

Base configuration includes a single processing instance in a 5.1+2+2 channel configuration, with or without Dolby® encoding and Nielsen® Watermark encoding. A second processing instance (5.1+2+2 or 2+2+2) with or without the Dolby and Nielsen options can be enabled at any time by simply entering a factory provided key.

Local audio insertion, SAP/DVS with downmix replacement, full-time downmix, and support for Audio Description (warble tone) is provided.

Failover bypass relays on all I/O maintain signal continuity and dual auto-ranging power supplies enable redundancy and worldwide compatibility.

Designed and built in the USA, the lightweight and rugged 1RU AERO.100 is a solid investment in performance and flexibility.

# FEATURES

- Linear Acoustic AEROMAX loudness and dynamics control
- UPMAX® II upmixing with AutoMAX-II™
- Intelligent Dynamics™ with advanced ITU (AI) Limiter
- Standard single processing instance in a 5.1+2+2 configuration
- Available second instance in a 5.1+2+2 or 2+2+2 configuration, with or without Dolby encoding and Nielsen Watermark encoding
- Automatic downmix output
- Support for SAP/DVS
- Local audio/voiceover insertion
- Optional Audio Description (warble tone)
- Standard Dolby Digital (AC-3), Dolby Digital Plus and Dolby E decoding
- Available Dolby Digital (AC-3), Dolby Digital Plus, and Dolby E encoding
- 3GHz HD/SD-SDI I/O with included video delay (16 channels)
- AES I/O with reference input (8 channels)
- Dual PSU and relay bypass
- Extensive TCP/IP remote control and HTTP control

# IN DEPTH

## Comprehensive TCP/IP Remote

Provides control over all system settings, processing and coding parameters plus extensive metering of signal presence, processing and coding activity, and audio loudness. System status reports physical I/O details along with system, power supply and environmental health. The remote application also delivers remote audio, up to 5.1 channels, to the user so that signal quality can be auditioned anywhere link bandwidth permits. HTTP server is also included for simple get/set control of all parameters and status.

# IN DEPTH


## Nielsen Watermark Encoding

Generates revenue-critical Nielsen NAES II and the new Nielsen Watermark audience measurement codes. AERO.100 precisely inserts these signals for maximum code recovery after audio decoding and processing and before transmission encoding.

## Linear Acoustic Intelligent Dynamics

### The Basics

Linear Acoustic Intelligent Dynamics hybrid metadata processing is a patented hybrid of traditional multiband techniques and metadata. Because it is created during transmission encoding, this metadata requires no operator intervention or special tools and is a new version of the DRC part of the Dolby Digital encoder that has always been there. Except it is now effective and uncomplicated.

Featured at no additional cost, Intelligent Dynamics is actually a combination of two revolutionary technologies.

### How it Works

The first portion of Intelligent Dynamics is the Emmy® award-winning process developed by Linear Acoustic which generates new metadata dynamic range control via traditional processing techniques. This control can be set as permanent, reversible, or anywhere in between.  This technology is already present in every AERO.100, AERO.1000, AERO.2000, AERO.asi and AERO.soft product.

# IN DEPTH

The second portion of the process involves verifying and marking content as compliant upstream. This is accomplished using a portion of the new Dolby Intelligent Media Framework called Evolution which enables measurement and incorporation of authenticated loudness data within the audio at each stage of the content delivery chain from production onwards. Compliant programming can be passed through with minimal or no additional processing, while content that cannot be verified can still be made compliant.

Most current television audio processors control loudness by managing dynamic range in real time. This effectively manages "boundary issues" such as commercials that follow a quiet program segment, but also compromises the impact and excitement of intentionally dynamic scenes because all programming gets some degree of permanent correction.

Audio transmission formats such as Dolby Digital (AC-3) and Dolby Digital Plus (E-AC-3) already incorporate metadata dynamic range control (DRC) profiles, which can provide some control, but these profiles are not ideally suited for broadcast audio and often result in significant over and under processing if other metadata parameters are not correct.

Intelligent Dynamics overcomes these compromises and enables audio to be processed to the degree and permanence dictated by the programming itself. The content effectively and automatically controls the processing and eliminates the need for further downstream changes that could irreparably change it.

### The Result

Consumers can now enjoy the benefits of audio that is tailored to their liking using existing Dolby Digital and Dolby Digital Plus decoders. From the default of controlled dynamic range for noisy environments and small television speakers to full dynamic range for well-produced programming and multi-channel home theater systems, Linear Acoustic Intelligent Dynamics delivers the best of both worlds without compromise.

# SPECIFICATIONS

## Processing

AEROMAX multistage adaptive wideband and multiband loudness and dynamic range control with ITU-R BS.1770 and EBU R128 loudness metering

Dual UPMAX II two-channel to 5.1-channel upmixers plus main channel downmixing with automatic bypass of discrete content

Linear Acoustic Intelligent Dynamics with Advanced ITU (AI) Limiter

# SPECIFICATIONS

## Audio Encoding/Decoding

Standard Dolby Digital (AC-3), Dolby Digital Plus and Dolby E decoding

Available Dolby Digital (AC-3), Dolby Digital Plus, and Dolby E encoding

## Reference

48kHz via AES DARS (or any AES signal applied to the Ref In connector), AES In 1, SDI, or from the internal 48kHz clock (standalone use only)

## Sample Rate/Resolution/Frequency Response

48kHz, 24-bit, 20Hz to 20kHz below threshold

## AES I/O

Eight main inputs plus reference via 75-Ohm BNC female connectors. Eight main outputs plus encoder output. All digital inputs are 75 Ohm internally terminated, unbalanced. Signal levels per SMPTE 276M/AES-3ID-2001

## HD/SD-SDI I/O

Auto-sensing 3GHz HD/SD-SDI (SMPTE 292M/259M) inputs, up to 1080i/60/59.94/50Hz. De-embed up to 16 channels from applied SDI signal, process and/or encode, re-embed up to 16 channels. Supports SMPTE 2020 A and B VANC metadata

## Parallel GPI/O Control Port

25-pin female D connector, 0-5V TTL levels for 8 inputs and 8 outputs; controls simple preset recalls plus voiceover/EAS insertion

## Serial Metadata Input

9-pin female D connector; 115.2 kbps; pinout per SMPTE 207M (RS-485); Designed to directly interface with Dolby serial metadata (SMPTE RDD6)

## Ethernet

Gigabit Ethernet via RJ45 supports included TCP/IP remote control application; HTTP server included for get/set control of all parameters

# SPECIFICATIONS

## Remote Control

Windows®-compatible TCP/IP remote control Included application for full setup and control, ITU-R BS.1770 metering for all programs, encoder statistics, and return audio for remote monitoring (network speed permitting). HTTP server allows get/set control from PC

## Front Panel Controls and Indicators

Graphical OLED display

## Power Requirements

Dual power supplies, each rated at 100-264 VAC, 50/60Hz, auto-sensing, 150W max. total

## Dimensions and Weight

1RU - 1.75"H x 19"W x 15.5"D (44 x 483 x 394 mm)
Net weight: 9 lbs. (4 kg), approximate.

## Shipping Dimensions and Weight

22"W x 20"D x 7"H (559 x 508 x 178 mm)
Net weight: 15 lbs. (6.80 kg), approximate.

## Environmental

Fan cooled. Operating: 0 to 50 degrees C, non-operating -20 to 70 degrees C

## Regulatory

North America: Tested to comply with the limits for a class A digital device pursuant to Part 15 of the FCC rules (CFR). Power supplies are UL tested and approved

Eurpoe: Tested to comply with the requirements of harmonised Low Voltage Directive 2006/95/EC C and EMC Directive 2004/108/EC as indicated by the affixed CE marking; RoHS and WEEE compliant

## Warranty

Standard Telos Alliance 5-Year Warranty

# AERO.lite™
## Loudness Controller


## OVERVIEW

AERO.lite delivers CALM- and R128-compliant audio via the renowned Linear Acoustic AEROMAX® processing algorithm, but in a cost-effective 2-channel package for applications where 5.1-channel audio is not needed. Combined with comprehensive I/O choices, this makes AERO.lite the perfect choice for stereo-only main or backup transmission paths.

Like all Linear Acoustic processors, AERO.lite applies preset-driven loudness and dynamic range control using adaptive, look-ahead wideband and multiband algorithms. Standard presets are simplified for easy setup, and GPI control allows compliant content to pass untouched.

Failover bypass relays on all I/O maintains signal continuity while the auto-ranging power supply ensures worldwide compatibility. A sealed, locking 2.5mm DC input connector is provided for the available redundant power supply.

AERO.lite is designed and assembled in the USA and built on a broadcast-quality Linear Acoustic platform.

A bright yellow OLED display and integrated rotary navigation cluster provide straightforward menu navigation and adjustment.

# FEATURES

- Simple preset-driven loudness control
- Front panel display of I/O and processing activity
- HD/SD-SDI I/O with access to all audio pairs
- Built-in routing and audio format conversion
- AES I/O with reference input
- Analog I/O (+4dBu)
- Front panel headphone output
- GPI/O for alarms and control
- Available built-in ITU-R BS.1770 LKFS utility meter for simplified output level setup
- Available redundant external power supply
- Available Ethernet connectivity for logging and SNMP

# IN DEPTH

## Flexible I/O and Audio Extraction

Audio can be extracted from any pair of an applied HD/SD-SDI signal or taken from the AES or balanced analog inputs and routed to the processing core. Output is simultaneously provided on the front-panel headphone output, balanced analog outputs, AES outputs, and re-embedded into any/all SDI pairs. Analog or AES inputs can be used for SDI embedding with or without processing. Since all 16 channels are available for de-embedding and re-embedding, pair shuffling is easily accomplished.

# SPECIFICATIONS

## Processing

AEROMAX multistage adaptive wideband and multiband loudness and dynamic range control.

## Reference

48kHz via AES DARS (or any AES signal applied to the Ref In connector), AES In 1, SDI, or from the internal 48kHz clock (standalone use only)

## Sample Rate/Resolution/Frequency Response

48kHz, 24-bit, 20Hz to 20kHz below threshold

## AES I/O

All connectors are 75-Ohm BNC female; Main inputs with 75-Ohm internal termination; Signal levels per SMPTE 276M/AES-3ID-2001

## HD/SD-SDI I/O

Auto-sensing HD/SD-SDI (SMPTE 292M/259M) inputs, up to 1080i/60/59.94/50Hz, access to all 16 audio channels

## Analog I/O

10K Ohm balanced XLR female stereo inputs, +4dBu nominal, +24dBu Max.; Balanced XLR male outputs, +4dBu nominal, +24dBu Max into 600 Ohms

## Headphone Output

6.3mm front panel connector, +12 dBu max into 600-Ohms

## Parallel GPI/O Parallel Control Port

9-pin female D connector, 0-5V TTL levels

## Front Panel Controls and Indicators

Rotary navigation cluster plus graphical OLED display and headphone output

# SPECIFICATIONS

## Power Requirements

90-264 VAC, 50/60Hz, auto-sensing, 35W maximum; Backup redundant DC Input 12VDC, 2.5A maximum via 2.5mm sealed, locking connector with protective cover

## Dimensions and Weight

1RU - 1.75"H x 19"W x 11.5"D (44 x 483 x 293 mm)
Net weight: 6 lbs. (2.72 kg), approximate.

## Shipping Dimensions and Weight

22"W x 17"D x7"H (559 x 432 x 178mm)
Net weight: 11 lbs. (5.0 kg), approximate.

## Environmental

Convection cooled. Operating: 0 to 50 degrees C, non-operating -20 to 70 degrees C

## Regulatory

North America: Tested to comply with the limits for a class A digital device pursuant to Part 15 of the FCC rules (CFR). Power supplies are UL tested and approved

Europe: Tested to comply with the requirements of harmonised Low Voltage Directive 2006/95/ED and EMC Directive 2006/95/EC and EMC Directive 2004/108/EC as indicated by the affixed CE marking; RoHS and WEEE compliant

## Warranty

Standard Telos Alliance 5-Year Warranty

# AERO.one™
## Audio/Loudness Manager


## OVERVIEW

AERO.one accepts baseband audio via AES (8 channels) or HD/SD-SDI (16 channels) I/O in a 5.1+2 channel configuration. It includes AEROMAX® processing, dual UPMAX® upmixing engines, and a downmixed output.

It can be optionally configured with Dolby Digital (AC-3) and Dolby Digital Plus encoding and SNMP monitoring.

Failover bypass relays on all I/O maintain signal continuity and dual auto-ranging power supplies enable redundancy and worldwide compatibility.

Designed and built in the USA, the 1RU AERO.one is a solid and proven performer.

# FEATURES

- Linear Acoustic AEROMAX loudness and dynamics control
- UPMAX upmixing with AutoMAX-II™
- 5.1+2 channel configuration
- Available Dolby Digital (AC-3) and Dolby Digital Plus encoding
- Available SNMP monitoring

# IN DEPTH

## Processing

AERO.one uses the renowned AEROMAX processing algorithm. This multistage adaptive wideband and multiband loudness and dynamic range control delivers audio that is both compliant and pleasing to the viewer .

## Upmixing

2-channel to 5.1-channel upmixing is provided by the Hollywood-approved UPMAX algorithm which provides a compelling 5.1-channel audio experience while remaining completely downmix compatible. AERO.one includes the AutoMAX-II auto-detection algorithm to smoothly and automatically bypass upmixing when 5.1-channel audio is applied.

## Metadata

Metadata can be applied, if available, via the VANC space of an applied HD-SDI signal or from a standard serial input for control of upmixing and processing functions. Extensive protection is provided to protect viewers from the audible effects of incorrect or missing metadata.

# SPECIFICATIONS

## Processing Structure

AEROMAX® multistage adaptive wideband and multiband loudness and dynamic range control.

Dual UPMAX upmixers plus main channel downmixing with automatic bypass of discrete content.

## Audio Encoding

Available Dolby Digital (AC-3) and Dolby Digital Plus encoding

## Reference

48kHz via AES DARS (or any AES signal applied to the Ref In connector), AES In 1, SDI, or from the internal 48kHz clock (standalone use only)

## AES I/O

All connectors are 75-Ohm BNC female; Four main inputs with 75-Ohm internal termination; Four main outputs; 48kHz AES DARS reference input and passive loop out; Bypass input and encoder output; Signal levels per SMPTE 276M/AES-3ID-2001

## HD/SD-SDI I/O

Auto-sensing HD/SD-SDI (SMPTE 292M/259M) inputs, up to 1080i/60/59.94/50Hz. De-embed up to 16 channels from applied SDI signal, process and/or encode, re-embed up to 16 channels. Supports SMPTE 2020 A and B VANC metadata

## Processing Delay

16 msec fixed (any mode), add 137msec for AC-3 encoding

## Parallel GPI/O Control Port

9-pin female D connector, 0-5V TTL levels, used to control upmixing and audio switching

## Serial/Metadata Input

9-pin female D connector, 115 kbps, pinout per SMPTE 207M (RS-422/485)

# SPECIFICATIONS

## Front Panel Controls and Indicators

Highly visible LED display and simple navigation cluster for setup and adjustment. Status indicators for reference, SDI, metadata, codec, and fault condition

## Power Requirements

Dual power supplies, each rated at 100-264 VAC, auto-sensing, 40 W maximum

## Dimensions and Weight

1RU – 1.75"H x 19"W x 12.5"D (44 x 483 x 317 mm),
Net weight: 6 lbs. (2.72 kg), approximate.

## Shipping Dimensions and Weight

22"W x 17"D x 7"H (559 x 432 x 178 mm)
Net weight: 13 lbs. (5.90 kg), approximate.

## Environmental

Convection cooled. Operating: 0 to 50 degrees C, non-operating –20 to 70 degrees C

## Regulatory

North America: Tested to comply with the limits for a class A digital device pursuant to Part 15 of the FCC rules (CFR). Power supplies are UL tested and approved

Europe: Tested to comply with the requirements of harmonised Low Voltage Directive 2006/95/EC and EMC Directive 2004/108/EC as indicated by the affixed CE marking; RoHS and WEEE compliant

## Warranty

Standard Telos Alliance 5-Year Warranty

# AERO.mobile™

## Mobile DTV Audio/Loudness Manager


## OVERVIEW

Traditional audio processing alone cannot enable diverse audio content to be reproduced effectively on mobile and handheld devices, including content from Mobile DTV and live internet streaming.

Mobile viewing must overcome physical constraints including small speakers and environmental issues such as background noise. Additionally, program audio can range from mono to 5.1 channels and from whispery soft to screaming loud. These factors combine to impair intelligibility, making viewing tedious and frustrating.

AERO.mobile solves these issues by employing specialized processing algorithms to ensure the audio comes through with exceptional clarity in these challenging environments.

The rugged 1RU AERO.mobile is intended to be installed directly before the mobile audio encoder in either the AES or SDI paths.

Failover bypass relays are provided to ensure continuous service in the unlikely event of failure, while a bright LED display, rotary encoder, and four control keys provide straightforward menu navigation and adjustment.

# FEATURES

- Linear Acoustic AEROMAX® loudness and dynamics control with Mobilizer™ technology for up to two programs
- Input downmixer supports 5.1 and stereo sources
- 5.1+2 (downmix plus alternate channel), 5.1+2 (downmix with Mobilizer plus alternate channel), or 2+2 (dual stereo with dual Mobilizer) channel configurations.
- Six factory presets for standard programming and sports
- Automatic, metadata, or GPI control
- AES and HD/SDI-SDI I/O

# IN DEPTH

## Mobilizer™ Technology

Linear Acoustic Mobilizer technology was developed based on extensive research into normal and impaired hearing in both quiet and noisy environments. By using technology from the renowned CrowdControl™ algorithm to isolate dialog elements and combining new multiband techniques designed to preserve critical audio cues, program intelligibility is enhanced without the need for heavy handed processing. Mobilizer also provides pre-conditioning for the low bit rate HE AAC Mobile DTV audio encoder to maximize its performance at even the lowest rates.

# SPECIFICATIONS

## Processing

AEROMAX multistage adaptive wideband and multiband loudness and dynamic range control with Mobilizer technology

## Sample Rate/Resolution

48kHz, 24-bit

# SPECIFICATIONS

## Latency

15msec fixed

## AES I/O

All connectors are 75-Ohm BNC female; Four main inputs with 75-Ohm internal termination; Four main outputs; 48kHz DARS reference input/passive loop out; Signal levels per SMPTE 276M/AES-3ID-2001

## HD/SD-SDI I/O

Auto-sensing HD/SD-SDI (SMPTE 292M/259M) inputs, up to 1080i/60/59.94/50Hz, access to all 16 audio channels plus VANC metadata per SMPTE 2020M methods A and B

## Parallel GPI/O Control Port

9-pin female D connector, 0-5V TTL levels

## Serial Metadata Input

SMPTE 207M (RS485/RS422) connection per SMPTE RDD-6

## Front Panel Controls and Indicators

Rotary encoder and control keys plus bright dot matrix LED display

## Power Requirements

Dual, redundant 100-264 VAC, 50/60Hz, auto-sensing, 35W maximum

## Dimensions and Weight

1RU - 1.75"H x 19"W x 12.5"D (44 x 483 x 318 mm)
Net weight: 6 lbs. (2.72 kg), approximate.

## Shipping Weight and Dimensions

22"W x 17"D x 7"H (559 x 432 x 178 mm)
Net weight: 13 lbs. (5.90 kg), approximate.

# SPECIFICATIONS

## Environmental

Convection cooled. Operating: 0 to 50 degrees C, non-operating -20 to 70 degrees C

## Regulatory

North America: Tested to comply with the limits for a class A digital device pursuant to Part 15 of the FCC rules (CFR). Power supplies are UL tested and approved

Europe: Tested to comply with the requirements of harmonised Low Voltage Directive 2006/95/EC and EMC Directive 2004/108/EC as indicated by the affixed CE marking; RoHS and WEEE compliant

## Warranty

Standard Telos Alliance 5-Year Warranty

# AERO.asi™

## TS Audio Processor


## OVERVIEW

AERO.asi brings legendary Linear Acoustic loudness control and upmixing to the transport stream for DVB-ASI signals.

Handling up to 10 channels of baseband or encoded audio via both AES or HD/SD-SDI I/O, AERO.asi provides AEROMAX® audio processing, UPMAX®-II automatic upmixing, Dolby® Digital (AC-3) coding, and metering for over the air, over the top, and mobile television broadcasts.

Up to four available 5.1- or 2.0-channel PIDs can be individually specified with or without Nielsen® Watermark encoding in 5.1+2+2 or 2+2+2 channel configurations at the time of purchase or later in the field with a factory-provided key.

Failover bypass relays on all I/O maintain signal continuity and dual auto-ranging power supplies enable redundancy and worldwide compatibility.

Designed and built in the USA, the lightweight and rugged 1RU AERO.asi is a solid investment in performance and flexibility.

# FEATURES

- Linear Acoustic AEROMAX loudness and dynamic range control
- UPMAX®-II upmixing with AutoMAX-II™
- Intelligent Dynamics™ with advanced ITU (AI) Limiter
- Support for up to four available 5.1- or 2.0-channel PIDs
- Available Nielsen Watermark encoding for each PID
- Standard Dolby® Digital (AC-3) decoding/encoding
- DVB-ASI I/O
- Dual PSU and relay bypass
- Extensive TCP/IP remote control and HTTP control

# IN DEPTH

## Comprehensive TCP/IP Remote

Provides control over all system settings, processing and coding parameters plus extensive metering of signal presence, processing and coding activity, and audio loudness. System status reports physical I/O details along with system, power supply and environmental health. The remote application also delivers remote audio, including 5.1 channels, to the user so that signal quality can be auditioned anywhere link bandwidth permits. The built-in HTTP server provides for simple get/set control of all parameters and status.

## Nielsen Watermark Encoding

Generates revenue-critical Nielsen NAES II and the new Nielsen Watermarks audience measurement codes. AERO.asi precisely inserts these signals for maximum code recovery – after audio decoding and processing and before transmission encoding.

## Linear Acoustic Intelligent Dynamics

### The Basics

Linear Acoustic Intelligent Dynamics hybrid metadata processing is a patented hybrid of traditional multiband techniques and metadata. Because it is created during transmission encoding, this metadata requires no operator intervention or special tools and is a new version of the DRC part of the Dolby Digital encoder that has always been there. Except it is now effective and uncomplicated.

# IN DEPTH

Featured at no additional cost, Intelligent Dynamics is actually a combination of two revolutionary technologies.

## How it Works

The first portion of Intelligent Dynamics is the Emmy® award-winning process developed by Linear Acoustic which generates new metadata dynamic range control via traditional processing techniques. This control can be set as permanent, reversible, or anywhere in between. This technology is already present in every AERO.100, AERO.1000, AERO.2000, AERO.asi and AERO.soft product.

The second portion of the process involves verifying and marking content as compliant upstream. This is accomplished using a portion of the new Dolby Intelligent Media Framework called Evolution which enables measurement and incorporation of authenticated loudness data within the audio at each stage of the content delivery chain from production onwards. Compliant programming can be passed through with minimal or no additional processing, while content that cannot be verified can still be made compliant.
Most current television audio processors control loudness by managing dynamic range in real time. This effectively manages “boundary issues” such as commercials that follow a quiet program segment, but also compromises the impact and excitement of intentionally dynamic scenes because all programming gets some degree of permanent correction.

Audio transmission formats such as Dolby Digital (AC-3) and Dolby Digital Plus (E-AC-3) already incorporate metadata dynamic range control (DRC) profiles, which can provide some control, but these profiles are not ideally suited for broadcast audio and often result in significant over and under processing if other metadata parameters are not correct.

Intelligent Dynamics overcomes these compromises and enables audio to be processed to the degree and permanence dictated by the programming itself. The content effectively and automatically controls the processing and eliminates the need for further downstream changes that could irreparably change it.

## The Result

Consumers can now enjoy the benefits of audio that is tailored to their liking using existing Dolby Digital and Dolby Digital Plus decoders. From the default of controlled dynamic range for noisy environments and small television speakers to full dynamic range for well-produced programming and multi-channel home theater systems, Linear Acoustic Intelligent Dynamics delivers the best of both worlds without compromise.

# SPECIFICATIONS

## Processing

5.1+2+2 or 2+2+2 AEROMAX multistage adaptive wideband and multiband loudness and dynamic range control with ITU-R BS.1770 and EBU R128 loudness metering.

Dual UPMAX-II two-channel to 5.1-channel upmixers plus main channel downmixing with automatic bypass of discrete content.

Linear Acoustic Intelligent Dynamics with Advanced ITU (AI) Limiter

## Audio Encoding/Decoding

Standard Dolby Digital (AC-3) decoding/encoding

## Sample Rate/Resolution/Frequency Response

48kHz, 24-bit, 20Hz to 20kHz below threshold

## DVB-ASI I/O

DVB-ASI (ETSI TR 101 891 v1.1.1) Transport Stream input

## Parallel GPI/O Parallel Control Port

25-pin female D connector, 0-5V TTL levels for 8 inputs and 8 outputs; controls simple preset recalls

## Ethernet

Gigabit Ethernet via RJ45 supports included TCP/IP remote control application; HTTP server included for get/set control of all parameters.

## Remote Control

Windows®-compatible TCP/IP remote control Included application for full setup and control, ITU-R BS.1770 metering for all programs, encoder statistics, and return audio for remote monitoring (network speed permitting). HTTP server allows get/set control from PC Front Panel Controls and Indicators

## Front Panel Controls

Graphical OLED display

# SPECIFICATIONS

## Power Requirements

Dual redundant power supplies, each rated at 100-264 VAC, auto-sensing, 50/60 Hz, 150W maximum total

## Dimensions and Weight

1RU - 1.75"H x 19"W x 15.5"D (44 x 483 x 394 mm)
Net weight: 9 lbs. (4 kg), approximate.

## Shipping Dimensions and Weight

7"H x 22"W x 20"D (178 x 559 x 508 mm)
Net weight: 15 lbs. (6.8 kg), approximate.

## Environmental

Fan cooled. Operating: 0 to 50 degrees C, non-operating -20 to 70 degrees C

## Regulatory

North America: Tested to comply with the limits for a class A digital device pursuant to Part 15 of the FCC rules (CFR). Power supplies are UL tested and approved.

Europe: Tested to comply with the requirements and harmonised Low Voltage Directive 2006/95/EC and EMC Directive 2004/108/EC as indicated by the affixed CE marking; RoHS and WEEE compliant

## Warranty

Standard Telos Alliance 5-Year Warranty

# AERO.soft™

## Enterprise-wide Audio Processing


## OVERVIEW

Enterprise-wide audio and loudness management deployed on standard third party servers.

Trusted Linear Acoustic AEROMAX® loudness control, UPMAX® upmixing, and ITU-R BS.1770-3 metering plus the very latest Dolby® encoding and decoding guarantees compliance with world-wide audio loudness regulations and the highest audio quality.

Audio over IP connectivity via Livewire®/AES-67 enables I/O of any flavor and combination or integration with third party video and/or audio encoding software.

AERO.soft includes the AEROMAX adaptive wideband and multiband, multistage ITU compliant loudness control algorithm. Upmixing is provided by the Hollywood-approved UPMAX algorithm, which delivers engaging 5.1-channel audio experience for two-channel sources. Loudness, spectral balance, and image shifts are controlled while preserving more of the original content than previously possible. AERO.soft uses the latest Dolby codecs which support Evolution, part of their new Intelligent Media Framework.

Each AERO.soft service is capable of handling up to eight processing instances. Each instance can be either 5.1+2+2 with upmixing and downmixing, or 2+2+2. Both types include Auto Downmix output, support for SAP/DVS, Voiceover, and optional Audio Description (warble tone) functionality. ITU-R BS.1770-3 or EBU R128 meters are present on each output (24 total). Dolby Digital, Dolby Digital Plus, and Dolby E decoding and encoding is optionally available for each instance.

Multiple services can be deployed on servers with enough horsepower, so audio channels per rackspace are maximized.

# FEATURES

- Very high channel capacity software runs real-time on standard Windows® servers; Channel count constrained only by server processing capabilities
- I/O Independence via Livewire/AES-67 Audio over IP
- Linear Acoustic AEROMAX loudness/dynamics control
- UPMAX II automatic upmixing and downmixing
- Intelligent Dynamics™ with New Advanced ITU (AI) Limiter
- 5.1+2+2 (Voiceover/Local) 10-channel and 2+2+2 (voiceover/Local) engines available
- ITU-R BS.1770-3 compliant metering
- Extensive remote control enables metering, control, and local audio monitoring (up to 5.1 channels)
- Dolby Digital/Plus/E encoding and decoding option
- Nielsen® Watermark encoder option

# IN DEPTH

## Livewire

Broadcast facilities have been using Axia Livewire AoIP (audio over IP) technology to route hundreds of audio channels using standard Ethernet switches, cables, and other components. With thousands of devices on the air every day, reliability is proven.

Livewire is lightning fast, simple to implement, easy to manage, conforms to the AES-67 standard, and is the technology upon which Ravenna is built. It also significantly reduces the number of cables and wires needed to transport and share audio throughout the broadcast plant.

Nearly Any audio source – analog or AES, and now SDI – and myriad equipment from phone systems to codecs to satellite receivers – can be a part of the Livewire network.

## Linear Acoustic Intelligent Dynamics

### The Basics

Linear Acoustic Intelligent Dynamics hybrid metadata processing is a patented hybrid of traditional multiband techniques and metadata. Because it is created during transmission encoding, this metadata requires no operator intervention or special tools and is a new version of the DRC part of the Dolby Digital encoder that has always been there. Except it is now effective and uncomplicated.

Featured at no additional cost, Intelligent Dynamics is actually a combination of two revolutionary technologies.

# IN DEPTH

## How it Works

The first portion of Intelligent Dynamics is the Emmy® award-winning process developed by Linear Acoustic which generates new metadata dynamic range control via traditional processing techniques. This control can be set as permanent, reversible, or anywhere in between. This technology is already present in every AERO.100, AERO.1000, AERO.2000, AERO.asi and AERO.soft product.

The second portion of the process involves verifying and marking content as compliant upstream. This is accomplished using a portion of the new Dolby Intelligent Media Framework called Evolution which enables measurement and incorporation of authenticated loudness data within the audio at each stage of the content delivery chain from production onwards. Compliant programming can be passed through with minimal or no additional processing, while content that cannot be verified can still be made compliant.

Most current television audio processors control loudness by managing dynamic range in real time. This effectively manages “boundary issues” such as commercials that follow a quiet program segment, but also compromises the impact and excitement of intentionally dynamic scenes because all programming gets some degree of permanent correction.

Audio transmission formats such as Dolby Digital (AC-3) and Dolby Digital Plus (E-AC-3) already incorporate metadata dynamic range control (DRC) profiles, which can provide some control, but these profiles are not ideally suited for broadcast audio and often result in significant over and under processing if other metadata parameters are not correct.

Intelligent Dynamics overcomes these compromises and enables audio to be processed to the degree and permanence dictated by the programming itself. The content effectively and automatically controls the processing and eliminates the need for further downstream changes that could irreparably change it.

## The Result

Consumers can now enjoy the benefits of audio that is tailored to their liking using existing Dolby Digital and Dolby Digital Plus decoders. From the default of controlled dynamic range for noisy environments and small television speakers to full dynamic range for well-produced programming and multi-channel home theater systems, Linear Acoustic Intelligent Dynamics delivers the best of both worlds without compromise.

# SPECIFICATIONS

## Processing

AEROMAX multistage adaptive wideband and multiband loudness and dynamic range control with ITU-R BS.1770-3 loudness metering

UPMAX II two-channel to 5.1 channel upmixing and downmixing, automatically bypasses discrete content

Includes Auto Downmix output, support for SAP/DVS, Voiceover, and optional Audio Description (warble tone) functionality

Included Linear Acoustic Intelligent Dynamics works with Dolby Evolution as part of the their Intelligent Media Platform

## Software Benchmark

Eight 5.1+2+Voiceover cores with eight Dolby E decoders/Dolby Digital (AC-3) encoders and 24 ITU-R BS.1770-3 or EBU R128 metering instances require one Intel Core i7 3700 with 4GB RAM.

## Remote Control

Dedicated TCP/IP remote control application provides extensive metering, control, system management and remote monitoring of one or many instances; HTTP server included for get/set control of all parameters

## Options

- Up to 8 Dolby E/Digital/Plus Decoders and Encoders
- Up to 8 AEROMAX 5.1+2+2 Engines with dual UPMAX II upmix/downmix
- Up to 8 AEROMAX 2+2+2
- Up to 8 Audio Description Engines (warble tone)
- Up to 8 Nielsen Watermark Encoders

# AERO.x™

## High Density Loudness Control and Metering


## OVERVIEW

Compact and powerful, the half-rack AERO.x features trusted Linear Acoustic AEROMAX® loudness control, UPMAX® upmixing, and ITU-R BS.1770-3 metering, now in an ultra-compact form factor.

Dual, independent HD/SD-SDI I/O plus Livewire®/AES-67 Audio over IP provides flexible local and networked audio connectivity.

AERO.x is designed for high-density installations and can also be used where only a few processors are required or even as a stand-alone desktop unit. High quality audio is provided in a compact, feature rich, cost effective, expandable manner.

The AERO.x contains up to two processing engines, each of which can be either 2+2+Dual ITU Meters or 5.1+Upmix+Downmix+ Dual ITU Meters. Audio sources can be applied to either or both engines from any channels present on the SDI inputs or via Livewire/AES-67 Audio over IP. Outputs can be from either or both engines. Audio channel pairs can also be shuffled and passed between any input and any output. Audio/video timing can be matched using the built-in video delays.

# OVERVIEW

AERO.x includes the AEROMAX adaptive wideband and multiband, multistage ITU compliant loudness control algorithm. Upmixing is provided by the Hollywood-approved UPMAX algorithm which provides an engaging 5.1-channel audio experience while AutoMAX®-II smoothly and automatically bypasses upmixing for 5.1-channel content.

Loudness, spectral balance, and image shifts are controlled while preserving more of the original content than previously possible. AERO.x is also compatible with Dolby Evolution, part of its new Intelligent Media Framework.

# FEATURES

- Dual independent SDI inputs and outputs
- Adjustable compensating video delay
- Livewire/AES-67 AoIP
- Up to 16 channels of audio
- Linear Acoustic AEROMAX and UPMAX
- Up to four ITU-R BS.1770-3 compliant meters
- Support for Dolby Evolution
- AC power and backup Power over Ethernet
- Failover relays on SDI I/O

# IN DEPTH

## Livewire

Broadcast facilities have been using Axia Livewire AoIP (audio over IP) technology to route hundreds of audio channels using standard Ethernet switches, cables, and other components. With thousands of devices on the air every day, reliability is proven.

# IN DEPTH

Livewire is lightning fast, simple to implement, easy to manage, conforms to the AES-67 standard, and is the technology upon which Ravenna is built. It also significantly reduces the number of cables and wires needed to transport and share audio throughout the broadcast plant.

Nearly Any audio source – analog or AES, and now SDI – and myriad equipment from phone systems to codecs to satellite receivers – can be a part of the Livewire network.

# SPECIFICATIONS

## Processing Structure

ITU-compliant loudness and control using multiband/multi-stage AEROMAX dynamic range engine and upmixing via UPMAX with AutoMAX-II. Compatible with Dolby Evolution for automatic processing control.

## HD/SD-SDI I/O

Dual auto-sensing 3GHz HD/SD-SDI (SMPTE 292M/259M) I/O, up to1080i/60/59.94/50Hz with independent compensating video delay. Enables up to 16 channels of audio to be de-embedded, shuffled, re-embedded between the SDI and Audio over IP connections.

## Audio over IP

Axia Livewire and AES-67 compatibility

## Ethernet

Two RJ-45 100 BASE-T connections

## Sample Rate/Resolution/Frequency Response

48kHz, 24-bit, 20Hz to 20kHz below threshold

## Front Panel Controls and Indicators

Graphical OLED display, menu keys

# SPECIFICATIONS

## Power Requirements and Consumption

95-240 VAC, 50/60 Hz, 15W maximum. Redundant power via Power over Ethernet (PoE).

## Dimensions and Weight

1RU high x 1/2RU width - 1.72"H x 8.5"W x 11.75"D
(44 x 216 x 298mm) Net weight: 7 lbs. (3.2 kg), approximate.

## Shipping Dimensions and Weight

12"W x 16"D x 8"H (305 x 406 x 203mm)
Net weight: 11 lbs. (5 kg), approximate.

## Environmental

Convection cooled. Operating: 0 to 50 degrees C, non-operating –20 to 70 degrees C.

## Regulatory

North America: Tested to comply with the limits for a class A digital device pursuant to Part 15 of the FCC rules (CFR). Power supplies are UL tested and approved.

Europe: Tested to comply with the requirements of harmonised Low Voltage Directive 2006/95/EC and EMC Directive 2004/108/EC as indicated by the affixed CE marking; RoHS and WEEE compliant

## Warranty

Standard Telos Alliance 5-Year Warranty

## Options

- Second HD/SD-SDI I/O with Compensating Video Delay
- 5.1 AEROMAX+UPMAX+Downmix+dual ITU-R BS.1770-3 Meters
- 2.0+2.0 AEROMAX + dual ITU-R BS.1770-3 Meters

# SDI xNode™

## Livewire/AES-67 AoIP Interface


## OVERVIEW

SDI xNode builds upon the success of the popular AXIA xNode by de-embedding two separate HD/SD-SDI inputs and converting up to 16 channels of audio to the Livewire®/AES-67 Audio over IP formats. The audio is available as a Livewire network source or can be re-embedded into two separate SDI output streams. Compensating video delay for each SDI input ensures audio/video synchronization is maintained.

SDI xNode brings the power and flexibility of the Axia Livewire AoIP (audio over IP) format to broadcast television for the first time by de-embedding two incoming SDI streams and making that audio available on a Livewire network.

Facilities can now easily and efficiently de-embedded audio from SDI input (up to 16 channels) and route to any or all Livewire/AES-67 compliant devices in the plant.

Additionally, it is possible to take audio from either of the two incoming SDI streams, shuffle pairs, and create two unique outgoing SDI streams with matched audio/video latency.

# FEATURES

- Two relay-bypassed 3Gb/s SDI inputs with access to all audio channels
- Dual, independent compensating video delays
- Up to 16 channels of audio via Livewire/AES-67
- Includes all standard xNode controls including front panel and web interface

# IN DEPTH

## Livewire

Broadcast facilities have been using Axia Livewire AoIP (audio over IP) technology to route hundreds of audio channels using standard Ethernet switches, cables, and other components. With thousands of devices on the air every day, reliability is proven.

Livewire is lightning fast, simple to implement, easy to manage, conforms to the AES-67 standard, and is the technology upon which Ravenna is built. It also significantly reduces the number of cables and wires needed to transport and share audio throughout the entire broadcast plant.

Nearly Any audio source – analog or AES, and now SDI – and myriad equipment from phone systems to codecs to satellite receivers – can be a part of the Livewire network.

# SPECIFICATIONS

## HD/SD-SDI I/O

Auto-sensing 3GHz HD/SD-SDI with de-embedding for up to 8 channels from each SDI input for a total of 16 channels; de-embedded audio can be routed to Livewire output and/or re-embedded to either SDI output with SMPTE 292M (HD-SDI) and SMPTE 259M (SD-SDI) support

## Livewire/AES-67

Full Livewire and AES-67 compatibility

# SPECIFICATIONS

## Reference

Livewire at audio input, SDI-delivered audio clock at output

## Ethernet

Two RJ-45 100Base-T connections.

## Power Requirements and Consumption

95-240 VAC, 50/60 Hz, 15W maximum. Redundant power sourcing available via Power over Ethernet (PoE).

## Dimensions and Weight

1RU high x 1/2RU width - 1.72"H x 8.5"W x 11.75"D
(44 x 216 x 298mm) Net weight: 7 lbs. (3.2 kg), approximate.

## Shipping Dimensions and Weight

12"W x 16"D x 8"H (305 x 406 x 203mm)
Net weight: 11 lbs. (5 kg), approximate.

## Warranty

Standard Telos Alliance 5-Year Warranty

# LQ-1000™
## Loudness Quality Monitor

## OVERVIEW

Accepting up to 8 channels of baseband or Dolby® Digital (AC-3) encoded audio via AES or HD/SD-SDI I/O, LQ-1000 provides simultaneous LKFS loudness measurement for two independent programs (5.1+2) per ITU-R BS.1770-3 and EBU R128 standards. Dolby Digital Plus and Dolby E decoding is optionally available.

The vibrant front-panel display features a large, color-coded numeric loudness value for the primary 5.1-channel program or secondary 2-channel program. The secondary meter can also be used to display loudness values for an internally created Lo/Ro or Lt/Rt downmix.

Additional bar graph meters with adjustable integration times, a loudness histogram to track loudness trends, True Peak readings, and dialnorm and coding information is also displayed.

Additionally, LQ-1000 provides comprehensive logging to a USB or network drive allowing stations to keep records of their loudness measurements.

A rear-panel VGA output allows information to be viewed on an external monitor or multi-viewer.

# FEATURES

- LKFS metering for two programs – 5.1+2
- ITU-R BS.1770-3 and EBU R128 compliant
- HD/SD-SDI I/O
- AES I/O
- Dolby Digital (AC-3) decoding standard
- Available Dolby Digital and Dolby E decoding
- Large color-coded numeric display shows current loudness value
- Three additional bar-graph meters with adjustable integration times
- Histogram shows loudness history and trends
- Front-panel buttons for Start, Stop, and Reset
- Logging to USB drive or network for all meters and programs
- Built-in HTTP server for retrieval of loudness logs
- Front panel headphone output
- GPI/O for external start/stop/reset and alarms
- Dual PSU

# IN DEPTH

## Color Display

The bright LCD front panel display presents a wealth of important loudness information all at once, but remains easy to read at a glance. Critical loudness parameters like short, medium, and long term loudness, loudness history, current peak level, maximum peak level, and the loudness target are displayed.

## Loudness Speedometer™

The most important information – the current LKFS loudness value – is boldly displayed as a numeric value and is color-coded to represent the roughly 16dB-wide loudness "comfort zone" which is aligned around the adjustable target level. The visual is simple: blue is too quiet, green is just right, yellow is getting loud, and red is too loud.

# IN DEPTH

## Network Logging

Readings from each of the four meters for each program (5.1-channel and 2-channel) can be saved to a USB drive or external network drive. Loudness data for the past 24 hours, 48 hours, 7 days, plus a user-defined period of time is stored in the efficient and universal .csv format and can easily be retrieved from the built-in HTTP server.

# SPECIFICATIONS

## Metering Standards

ITU-R BS.1770-3 and EBU R128

## Audio Input and Decoding

Baseband input and Dolby Digital (AC-3) decoding standard
Available Dolby Digital Plus and Dolby E decoding

## AES I/O

Four 75-Ohm AES inputs and outputs via female BNC connectors; Outputs of selected inputs: AES, De-embedded SDI, Dolby decoded. Signal levels per SMPTE 276/AES-3ID-2001

## HD/SD-SDI I/O

Auto-sensing HD/SD-SDI (SMPTE 292M/259M) inputs up to 1080i/60/59.94/50Hz, access to all 16 audio channels plus VANC metadata per SMPTE 2020M methods A and B.
Re-clocked HD/SD-SDI output

## Headphone Output

l/4'' (6.35mm) front panel connector with volume control

## Parallel GPI/O Control Port

25-pin female D connector, 0-5V TTL levels for external start/stop/reset and alarms

# SPECIFICATIONS

## Serial Metadata Input

9-pin female D connector, 115 kbps per SMPTE 207M (RS-422/485): Directly interfaces with Dolby metadata (SMPTE RDD6)

## VGA Output

640x480 for connection to external monitor or multi-viewer

## Ethernet

Gigabit Ethernet via RJ45 for HTTP access and network logging

## Front Panel Controls

Rotary encoder and control keys plus color LCD display and headphone output

## Power Requirements

Dual power supplies, each rated at 100-240 VAC, auto-ranging, 100 W maximum

## Dimensions and Weight

3.5”H (2RU) x 19”W x 17”D; (89 x 483 x 432mm)
Net weight 10.8 lbs. (4.9 kg), approximate.

## Shipping Weight and Dimensions

22”W x 20”D x 9”H (559 x 508 x 229 mm)
Net weight:16 lbs. (7.3 kg), approximate.

## Regulatory

North America: Tested to comply with the limits for a class A digital device pursuant to Part 15 of the FCC rules (C FR). Power supplies are UL tested and approved.

Europe: Tested to comply with the requirements of harmonised Low Voltage Directive 2006/95/EC and EMC Directive 2004/108/EC as indicated by the affixed CE marking; RoHS and WEEE compliant

## Warranty

Standard Telos Alliance 5-Year Warranty.

# LQ-1™
## Loudness Meter

## OVERVIEW

LQ-1 combines ITU-R BS.1770 and EBU R128-compliant metering, comprehensive I/O options, an easy to read display, and a host of other features into a single, cost-effective, compact, and rugged 1RU package.

LQ-1 offers HD/SD-SDI, DVB-ASI, AES and analog I/O plus a TOSLINK® optical input, and accepts baseband and encoded signals with standard decoding for Dolby® Digital (AC-3), Dolby Digital Plus, and Dolby E.

Setup is simple: Select the desired input signal and choose to apply Dolby decoding and metadata if needed. Presets store diverse configurations and can be recalled from the front panel or by GPI.

Extensive alarm capabilities can indicate out of tolerance loudness, missing audio channels, corrupt or missing reference or metadata signals, and errors in Dolby-encoded bitstreams can be detected and logged.

# FEATURES

- LKFS metering for 5.1 or 2-channel audio with Dolby Dialogue Intelligence™
- ITU-R BS.1770-3 and EBU R128 compliant
- HD/SD-SDI, DVS-ASI, AES, analog, and TOSLINK optical inputs
- HD/SD-SDI (re-clocked), AES, and analog outputs
- Front panel headphone output
- Accepts up to 8 channels of baseband or encoded audio, with standard decoding for Dolby Digital (AC-3), Dolby Digital Plus, and Dolby E signals
- Downmixed AES and analog output
- LTC and ATC (SDI) timecode inputs for time-stamped logging
- GPI/O alarms and control
- Available SNMP via Ethernet for alarms and logging
- Available redundant external PSU

# IN DEPTH

## Dolby Dialog Intelligence

Included Dolby Dialog Intelligence provides the most accurate estimate of loudness possible in an automatic meter. By pausing integration during non-dialog sections and reverting to BS.1770-2 over time, loudness can finally be measured with accuracy independent of program dynamic range.

## Comprehensive Metering

Input and output signal levels are displayed alongside ITU-R BS.1770 measured LKFS loudness and dialnorm metadata to give instant verification of loudness compliance.

## Flexible I/O and Audio Extraction

Baseband or coded audio can be extracted from any pair of an applied HD/SD-SDI signal, AES, optical or balanced analog inputs and routed to the decoding and metering core. In DVB-ASI mode, audio can also be sourced from a selected PID and then decoded and measured.

# IN DEPTH

## Downmix Output

A selectable Lt/Rt or Lo/Ro downmixed output is provided simultaneously via the front panel headphone connector and rear panel AES and stereo +4dBU balanced analog outputs. Functions such as metadata emulation and Associated Dialog (AD or Visual Descriptive) mixing can also be auditioned on the downmixed output.

# SPECIFICATIONS

## Metering Standards

ITU-R BS.1770-1/2/3 with Dolby Dialog Intelligence and support for EBU R128.

## Audio Input Decoding

Baseband input and Dolby Digital (AC-3), Dolby Digital Plus, and Dolby E decoding standard. DVB-ASI mode to extract and meter audio from selected PID.

## AES I/O

Eight main input channels via four 75-Ohm BNC female connectors, internally terminated; Two-channel downmix AES output; Signal levels per SMPTE 276M/AES-3ID-2001.

## HD/SD-SDI I/O

Auto-sensing HD/SD-SDI (SMPTE 292M/259M) inputs, up to 1080i/60/59.94/50Hz, access to all 16 audio channels.

## DVB-ASI Input

DVB-ASI (ETSI TR 101 891 v1.1.1) Transport Stream Input (via SDI connector). Audio PID can be selected for decoding and measurement.

## Analog I/O

10K Ohm balanced XLR female stereo inputs, +4dBu nominal, +24dBu Max.; Balanced XLR male outputs, +4dBu nominal, +24dBu Max into 600 Ohms.

# SPECIFICATIONS

## Optical Input

Female TOSLINK connector for stereo or encoded input

## Headphone Output

6.3mm front panel connector, +12 dBu max into 600-Ohms

## Sample Rate/Resolution

48kHz, 24-bit

## Parallel GPI/O Control Port

9-pin female D connector, 0-5V TTL levels

## Serial Metadata Input

9-pin female D connector, 115 kbps per SMPTE 207M
(RS-422/485): Directly interfaces with Dolby metadata (SMPTE RDD6

## Ethernet

10/100 BASE-T via RJ45 for available external logging and SNMP reporting

## Front Panel Controls and Indicators

Rotary navigation cluster plus graphical OLED display

## Power Requirements

90-264 VAC, 50/60Hz, auto-sensing, 35W maximum; Backup redundant DC Input 12VDC, 2.5A maximum via 2.5mm sealed, locking connector with protective cover

## Dimensions and Weight

1RU - 1.75"H x 19"W x 11.5"D (44 x 483 x 293 mm)
Net weight: 6 lbs. (2.72 kg) approximate.

## Shipping Dimensions and Weight

22"W x 17"D x 7"H (178 x 559 x 432 mm)
Net weight: 11 lbs. (5 kg), approximate.

# SPECIFICATIONS

## Environmental

Convection cooled. Operating: 0 to 50 degrees C, non-operating -20 to 70 degrees C.

## Regulatory

North America: Tested to comply with the limits for a class A digital device pursuant to Part 15 of the FCC rules (CFR). Power supplies are UL tested and approved.

Europe: Tested to comply with the requirements of harmonised Low Voltage Directive 2006/95/EC and EMC Directive 2004/108/EC as indicated by the affixed CE marking; RoHS and WEEE compliant.

## Warranty

Standard Telos Alliance 5-Year Warranty.

# MT2000™
## Multichannel Bitstream Analyzer


## OVERVIEW

The Linear Acoustic MT2000 Bitstream Analyzer is a portable, handheld diagnostic tool that can monitor and generate Dolby® Digital, Dolby Digital Plus, Dolby E, and PCM bitstreams.

With its built-in test-signal generator, the MT2000 allows audio system integrators and service engineers to quickly monitor and check the integrity and composition of Dolby encoded and PCM signals routed through production, broadcast, cable, satellite facilities, or even a home theater system.

# FEATURES

- ITU-R BS.1770-1/2/3 loudness metering
- Built-in test-signal generator
- Accepts signals via AES, TOSLINK optical, 3GHz SDI, or HDMI connectors
- Outputs Dolby Digital/Plus/E and PCM bitstreams
- Bright OLED display
- Built-in processed monitor speaker
- Powered by a replaceable internal NiMH rechargeable battery pack or from its DC power port via included universal power supply

# IN DEPTH

The MT2000 accepts signals via MADI/AES, TOSLINK™ optical, 3GHz SDI, and HDMI connectors. The unit identifies the format of the selected input signal and activates the appropriate built-in decoder. Pass-through mode permits modification of the AES channel status bits of the input signal before passing it to the outputs. Monitoring capabilities include error detection at the AES3 layer and within the coded audio layers, including SMPTE 337 formatting information and Dolby E guard band position.

In addition to displaying audio signal statistics and metadata, the MT2000 includes ITU-R BS.1770-1/2/3 loudness measurement with selectable Dolby Dialogue Intelligence™ to support ATSC A/85 and EBU R128. Logging can be provided via the USB connector and/or via SNMP over Ethernet.

An extensive set of useful Dolby Digital, Dolby Digital Plus, and Dolby E test bitstreams is stored internally, and users can modify the set in the field via software download. The MT2000 can generate the selected bitstreams simultaneously on all output connectors, even while receiving and decoding an input signal. The MT2000 is also capable of generating two-channel PCM signals. In this mode, the user can select the output waveform type (white noise, pink noise, sine, square), amplitude, and frequency. Test signals and analysis are also provided for latency and basic lip sync.

Signals are provided simultaneously via the MADI/AES and TOSLINK optical outputs and can be re-embedded into any of the SDI pairs. Output can be the original input signal, a multichannel PCM decoded version of the input signal, test signals, or, in the case of the SDI output, a combination of all of these. Inputs can be used as sources for embedding even if not used for decoding thus channel shuffling can be easily accomplished.

# IN DEPTH

A bright yellow OLED display and integrated rotary navigation cluster provide straightforward menu navigation and function adjusted. A standard 1/8-inch stereo headphone jack can be switched to monitor any two decoded channels or a downmix of the whole program. A wide-range equalized and processed speaker provides a surprisingly loud output useful for quick checks and for emulating sound systems found in portable devices.

# SPECIFICATIONS

## Audio Formats

Dolby Digital, Dolby Digital Plus, and Dolby E inputs and outputs of test bitstreams; Stereo (AES/TOSLINK optical) and multichannel PCM (SDI and HDMI) input; Live generation of PCM waveforms such as white noise, pink noise, sine, and square waves, latency test signal, A/V sync test pulse (for beep/flash).

## Metering

ITU-R BS.1770 loudness measurement with Dolby Dialogue Intelligence; Error detection at the AES3 layer and within the coded audio layers, including SMPTE 337 formatting information and Dolby E guard band position.

## AES and MADI I/O

All connectors are 75-Ohm BNC female; Main inputs with 75-Ohm internal termination; Signal levels per SMPTE 276M/AES-3ID-2001. Compatible with consumer S/PDIF connections. AES I/O connectors also serve as I/O for MADI.

## HD/SD-SDI I/O

Auto-sensing 3GHz HD/SD-SDI (SMPTE 292M/259M/424M), up to 1080p/60/59.94/50Hz, access to audio and VANC metadata and timecode.

## DVB-ASI (Option)

DVB-ASI (ETSI TR 101 891 v1.1.1) Transport Stream Input (via SDI connector). Audio PID can be selected for decoding and measurement.

# SPECIFICATIONS

## HDMI Input (Option)

Multichannel baseband or encoded audio can be demultiplexed and analyzed from HDCP and non-HDCP signals.

## TOSLINK Optical I/O

Supports consumer IEC 61937 input and output.

Front Panel Controls and Indicators
Rotary joystick navigation cluster plus graphical OLED display

## Ethernet

10/100BT via RJ45, supports logging, updates, and optional SNMP

## USB

Connection for local storage for logging and metadata via adapter.

## Metadata Input

VANC from SDI input; serial metadata via USB to RS-485 adapter supporting Dolby Metadata (SMPTE RDD6)

## Headphone Output

3.5mm (1/8-inch) side connector, +12 dBu max into 600-Ohms

## Power Requirements

Internal NiMH rechargeable battery which can be field replaced if necessary; Power and charge via dedicated DC input

## Dimensions and Weight

7.9"H x 4"W x 1.6"D (200 x 100 x 41 mm)
Net weight: 3 lbs (1.36 kg), approximate.

## Shipping Weight

6 lbs. (2.7 kg), approximate.

# SPECIFICATIONS

## Environmental

Fan cooled. Operating: 0 to 50 degrees C, non-operating –20 to 70 degrees C.

## Regulatory

North America: Designed to comply with the limits for a class A digital device pursuant to Part 15 of the FCC rules (CFR). Power supplies are UL tested and approved.

Europe: Designed to comply with the requirements of harmonised Low Voltage Directive 2006/95/EC and EMC Directive 2004/108/EC as indicated by the affixed CE marking; RoHS and WEEE compliant.

## Warranty

Standard Telos Alliance 5-Year Warranty.  90 days for battery.

## Supplied Accessories

Carrying case, universal DC power supply, USB memory stick containing user manual.

# UPMAX™ v4
## Surroundfield Controller


## OVERVIEW

Drawing upon the success of its predecessors in the UPMAX line, this compact, lightweight, and rugged unit produces stunning 5.1 channel audio from two-channel sources and is perfectly suited for OB trucks and post-production facilities.

Offering AES and auto-sensing HD/SD-SDI I/O, the UPMAX v4 combines the renowned UPMAX® upmixing/downmixing algorithm with AutoMAX-II™ auto-detection for seamless transitions between stereo and native surround input sources, producing an upmix that is completely downmix compatible.

The UPMAX v4 comes standard with a built-in ITU loudness meter, dual power supplies, internal relay bypass, SNMP monitoring, and a monitor section with calibration tones and pink noise, eight +4dBu balanced analog outputs, and volume, mute, and return to reference control.

A remote volume and GPI I/O control is optional.

# FEATURES

- 5.1-channel audio from 2- and 3-channel sources
- Fulltime 2-channel downmix output
- Very low latency for live applications
- HD/SD-SDI and AES inputs
- HD/SD-SDI, AES, and multi-channel analog outputs
- Built-in ITU loudness meter
- SNMP monitoring
- Dual redundant internal PSU and relay bypass

# IN DEPTH

## The Most Stable and Trusted Upmixing Available

Tracing its lineage back to the original UPMAX 2251, UPMAX v4 offers the most stable and trusted algorithm in use today for both production and unattended upmixing. Critically, its output is completely downmix compatible and the resulting downmix is nearly indistinguishable from the original 2-channel source. For unattended operation, AutoMAX-II detection seamlessly and automatically transitions between discrete and upmixed audio, or control can be handled via GPI/O.

## Flexible Surroundfield

The upmixed "surroundfield" can be infinitely adjusted, allowing programming ranging from simple stereo audio to an LtRt downmix to be appropriately reproduced through a 5.1-channel playback system. Optional bass enhancement for the LFE channel allows a subwoofer channel to be created without compromising the downmix.

## Built-in Loudness Metering

A built-in LKFS meter provides loudness measurements per ITU-R BS.1770-3, simplifying the process of making sure output levels are properly adjusted to comply with loudness regulations around the world.

# SPECIFICATIONS

## Processing

Linear Acoustic UPMAX with AutoMAX-II

## Latency

4.6 msec (SRC bypass), 7 msec (SRC On)

## AES I/O

48kHz, 24-bit; Signal levels per SMPTE 276M/AES-3ID-2001; All connectors are 75-Ohm BNC female with internal 75-Ohm termination

## HD/SD-SDI I/O

Auto-sensing and de-embedding of up to 16 channels from applied SDI signal with upmixing and/or re-embedding up to 16 channels. Signal levels per SMPTE 292M/259M. Supports (SMPTE) 2020 A and B VANC metadata. Up to 1080i/60/59.94/50Hz supported

## Analog Audio Output

Eight channels of output, +4 dBu, balanced, via a DB-25 connector following the TASCAM™ pinout.

## Parallel GPI/O Control Port

9-pin female D connector, 0-5v TTL levels

## Serial/Metadata Input

9-pin female D connector, 115 kbps, pinout per SMPTE 207M (RS-422/485)/RDD6

## Remote Control Input

9-pin female D connector for optional remote control

## Front Panel Controls and Indicators

Rotary encoder and graphical OLED display

# SPECIFICATIONS

## Power Requirements

Dual, redundant 100-264 VAC, 50/60Hz, auto-sensing,
35W maximum

## Dimensions and Weight

1RU - 1.75"H x 19"W x 11.5"D (44 x 483 x 317 mm)
Net weight: 6 lbs. (2.72 kg), approximate.

## Shipping Dimensions and Weights

22"W x 17"D x 7"H (178 x 559 x 432 mm)
Net weight: 13 lbs. (5.9 kg), approximate.

## Environmental

Convection cooled. Operating: 0 to 50 degrees C, non-operating –20 to 70 degrees C.

## Regulatory

North America: Tested to comply with the limits for a class A digital device pursuant to Part 15 of the FCC rules (CFR). Power supplies are UL tested and approved.

Europe: Tested to comply with the requirements of harmonised Low Voltage Directive 2006/95/EC and EMC Directive 2004/108/EC as indicated by the affixed CE marking; RoHS and WEEE compliant.

## Warranty

Standard Telos Alliance 5-Year Warranty

# LA-5269

## Dolby® Digital/Plus Transcoder

## OVERVIEW

LA-5269 provides everything necessary to accomplish simple, reliable encoding or transcoding. Features include frame synchronized AES inputs, HD/SD-SDI, or AES-11 (DARS) reference, plus internal or external serial metadata. Coding is provided by a Dolby-manufactured encoder module featuring the latest versions of the codecs for superior sound quality.

Additional functionality can be added as needed via software, enabling support of additional encoding and/or transcoding flavors.

A bright LED display and rotary encoder with four control keys provide easy menu navigation. Dual redundant power supplies, GPI/O, and a hard relay bypass are standard, while SNMP monitoring is offered as an option.

# FEATURES

- Frame synchronized AES inputs
- HD/SD-SDI or AES-II (DARS) reference
- Internal or external serial metadata
- Available SNMP monitoring

# IN DEPTH

## Encoding

Encode to Dolby Digital (AC-3) from PCM, encode to Dolby Digital Plus (E-AC-3) from PCM, transcode Dolby Digital to Dolby Digital Plus, encode to AAC or HE-AAC (with or without Dolby metadata) from PCM.

Dolby E decoding is optional, and allows Dolby E sources to be decoded and encoded into Dolby Digital, Dolby Digital Plus, HE-AAC and AAC.

## Software Updates

The LA-5269 provides a low-cost encoding platform that allows features to be enabled without the need to replace hardware. A simple software update can add features or codecs at any time. System status and additional I/O features can also be added as needed, essentially allowing broadcasters to pay for only what they need.

# SPECIFICATIONS

## Supported CODECs

Dolby Digital (AC-3) Encode from PCM or Dolby E
Dolby Digital Plus (E-AC-3) Encode from PCM or Dolby E, Transcode from AC-3
Available Dolby E decoding

# SPECIFICATIONS

## AES I/O

All connectors are 75-Ohm BNC female; Four main inputs with 75-Ohm internal termination; Four main outputs; 48kHz AES DARS reference input and passive loop out; Bypass input and encoder output; Signal levels per SMPTE 276M/AES-3ID-2001

## HD/SD-SDI I/O

De-embedding, decoding/encoding, and re-embedding of 16 channels of applied HD/SD-SDI signal. Signal levels per SMPTE 292M/259M. Supports SMPTE 2020 A and B VANC metadata. Up to 1080i/60/59.94/50Hz supported.

## Latency

137 msec for Dolby Digital (AC-3) and Dolby Digital Plus (E-AC-3) encode -99/120 msec for Dolby E decode (NTSC/PAL)

## Parallel GPI/O Control Port

9-pin female D connector, 0-5V TTL levels

## Serial Metadata Input

9-pin female D connector, 115 kbps, pinout per SMPTE 207M (RS-485); protocol per SMPTE RDD6 metadata specifications and is Dolby compliant. HD/SD-SDI (SMPTE 292M/259M)

## Power Requirements

Dual 100-264 VAC, auto-sensing, 50/60Hz, 35 W total maximum

## Dimensions and Weight

1RU - 1.75"H x 19"W x 12.5"D (44 x 483 x 318 mm)
Net weight: 6 lbs. (2.72 kg), approximate.

## Shipping Dimensions and Weight

22"W x 17"D x 7"H (178 x 559 x 432 mm)
Net weight: 13 lbs. (5.9 kg), approximate.

# SPECIFICATIONS

## Environmental

Convection cooled. Operating: 0 to 50 degrees C, non-operating -20 to 70 degrees C

## Regulatory

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