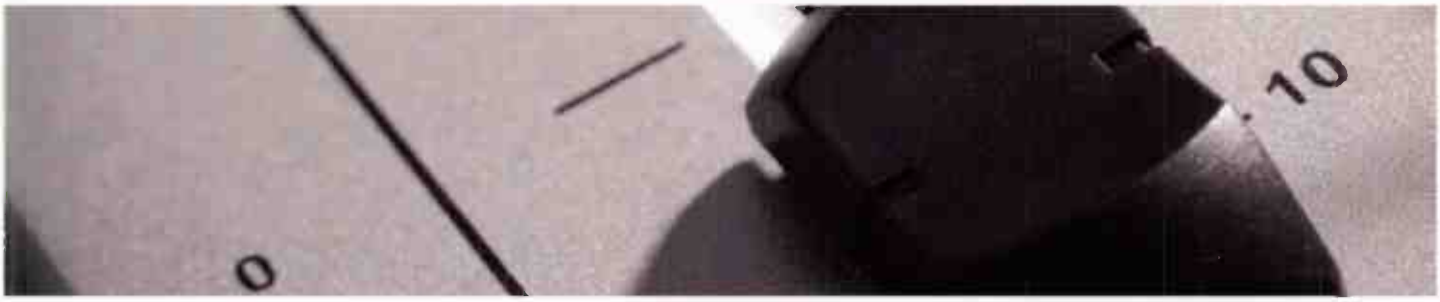
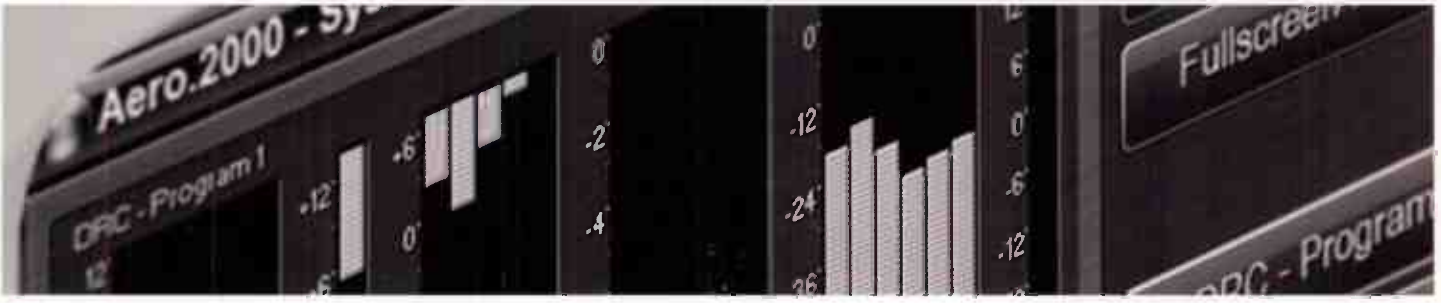
# NOW!

TELOS | OMNIA | 25-SEVEN | AXIA | LINEAR ACOUSTIC

TELEPHONY AND CODECS / AUDIO PROCESSING / TIME MANAGEMENT / CONSOLES AND IP AUDIO / DTV AUDIO




VOL.9 | 2013-14

Telos® Omnia® 25 seven™ Axia® LINEAR ACOUSTIC®

# THE TELOS ALLIANCE
## 1984-PRESENT

The history of the organization we call "The Telos Alliance" is well documented, and the individual companies it encompasses – Telos, Omnia, Axia, Linear Acoustic, and 25-Seven – have all developed game-changing technologies and products that have raised the bar for quality and innovation in the radio and television industries.

One would be hard pressed to walk into any radio station today and not find a Telos hybrid or a Zephyr codec hard at work simplifying and improving the quality of on-the-air phone calls and remote broadcasts. Omnia audio processors continue to be the choice of major (and not so major) market radio stations for their dial-dominating loudness and exceptional audio quality. Axia consoles with Livewire technology have re-defined how radio studios are built and how audio is routed and shared in the station. The 25-Seven lineup of products helps stations protect their licenses by "staying clean" on the air, and magically expand and contract time in ways that were unheard of just a few years ago. On the television side, local affiliates and major networks alike have embraced Linear Acoustic products as the industry standard for loudness control, upmixing, and metering digital television audio.

The Telos Alliance certainly fits the strict definitions of an "alliance" – "a bond or connection between parties or individuals," or "an association formed to further the common interests of the members." Indeed, the people who make up our organization do share a connection, and as members of that group, we have and do leverage our collective "brain trust" to further our success as a business.

But those words fail to capture the spirit of The Telos Alliance: Innovative, industry-changing, ground-breaking products developed for broadcasters by broadcasters. For nearly three decades, people who live and breathe radio and television have come into the fold to share their experience, their ideas, and their absolute passion for our industry. At one point, many of them were customers and fans of our products, and bring with them unique and valuable perspectives.

Even as our latest products come to market, we're already challenging ourselves to push further and make them even better, or asking "What if..." and experimenting with concepts and technologies that in some cases don't even exist yet. And while we love technology, we don't exploit it for its own sake. The world is full of "cool" gadgets that, while interesting, really don't serve a good purpose. In order for a product to bear the Telos, Omnia, Axia, Linear Acoustic or 25-Seven brand name, it must address a real need or somehow make a difference in the every day operation of the facilities in which it is installed, and it must do it better than anything else on the market.

For the people who make up The Telos Alliance, the "connection between parties" and the "furthering of common interests" doesn't stop at the outer walls of our buildings or at the end of the work day. It extends to you, our valued customers, everywhere and always, and is as deeply engrained in our DNA as it is in yours.

# BEHIND THE SCENES OF INNOVATION

## A LETTER FROM OUR CEO, FRANK FOTI

I know it sounds clichéd, but the phrase "when one door shuts, another opens" is ever so true! Looking back at our recent history we've experienced the unfortunate passing of our founder, Steve Church. Then, not long thereafter, we moved forward as 25-Seven merged into our Alliance. This latter move expands – yet again – our organization, and it brings together more of those "get it" guys who possess passion, drive, creativity, determination, and a willingness to win. Qualities Steve, Tim Carroll, and I shared, still possess, and it continues to be the basis of our livelihood.

Speaking of 25-Seven, wow, what an interesting group! They were one of our initial Livewire technology partners, and business buddies. Over the years, we realized, together, there might just be something special if we teamed up. Well, in typical Telos Alliance fashion, we didn't even wait for the ink to dry on the merger documents before beginning to play in the proverbial sandbox...we just jumped right in. As you'll soon read, the fruit of our collective labor is revealed later on inside this publication.

This year, 2013, marks the 25th anniversary of our radio processing division, which began its life as Cutting Edge Technologies, but you know it now as Omnia Audio. The combination of Omnia's upcoming birthday, and memories of my partner-in-crime, Steve, gave me reason to reflect on our past. Aside from all the obvious the industry knows about us, here's some behind the scenes reflections about a couple of individuals who were great influence upon us...Thomas Edison and Walt Disney.

Aside from the invention of the light bulb, motion picture, and phonograph, it's the style and manner of Thomas Edison that we, here at The Telos Alliance, identify with. He was known to surround himself with others who were experts in their field, he took chances – and sometimes failed, loved to challenge his colleagues, created an enjoyable yet interesting workplace, and was known to gather his troops together late at night for food and general merriment.

Steve, Tim, and I had many discussions about Mr. Edison, and we've each identified traits that occurred in Menlo Park, which occur today in our respective business environments. While Mr. Edison created a way to convey sound to the masses with the cylindrical recording, we've found a way to make sure it gets to those masses sounding the best possible for the transmission medium involved. Some things just never change, as they stand the test of time.

Walt Disney, a creative visionary, was also known to take risks, push the envelope, he created Imagineering, along with many of the world's greatest feature films, cartoons, a famous mouse, and incredible theme parks. Yet, his main desire was to offer mankind a happier place on earth. He did all that and more during sixty-five years of an incredible life. To me, he's been an inspiration for a very long time. I've studied Walt's life, and recently visited the family museum in San Francisco. It's interesting to note a number of parallels between Mr. Disney, his company, and our Alliance.

Walt and his brother Roy grew up in the humble surroundings of the midwest, in Marceline, Missouri. Just as Steve and I also grew up in the humble midwest cities of Lansing, Michigan, and Cleveland, Ohio. Walt and Roy began their business with whatever meager resources they had, and then bootstrapped their way forward in order to grow their company. The Disney brothers were driven by passion, determination, and a willingness to do whatever it takes in order to get the job done, figure some-

thing out, or reach a new goal. Innovation, thinking outside the box, and raising the bar was a mantra for the Disneys, just as it remains a driving force within our company today.

History shows how two renegade broadcast engineers (now three with the inclusion of little brother Tim) followed a path in like fashion. Driven by our dreams, we knew in our hearts there were cool ideas to make broadcast telephony, radio and television sound better. Using good old elbow grease, and a refuse-to-lose mentality, our exciting adventure started. Fortunately, that journey continues today, and actually it has picked up speed with each passing year.

By the way, there's another parallel shared between Walt and yours truly, our love for live steam railroading! At the end of his day, Walt would head to his home, in the Holmby Hills of Southern California. There, he'd fire up and operate his 1/8th scale steam locomotive Lilly Belle, as this provided freedom, peace, and refuge, which fueled his creativity. No difference than the joy brought on by East Wind #3712, my 1/8th scale live steam locomotive.

Additionally, there's another extremely important quotient involved with the companies of Thomas Edison, Walt Disney, and ours, and it's the ability to possess an incredible foundation made up of amazing people. These are our co-workers, colleagues, associates, and trusted staff. If not for them, we're just thoughts, ideas, and dreams that may never have materialized.

While there's been reflection shared here, our vision is focused forward. As you'll see, once again, inside these pages, every one of our divisions is offering something new. Additionally, we're working behind the scenes as a driving force to improve the technical broadcasting landscape. Our efforts regarding the optional use of SSBSC (Single Sideband Suppressed Carrier) for FM-Stereo transmission has moved into the NRSC (National Radio Systems Committee) forum. Early test results show, what broadcasters who've tested it in the field already know, that it reduces the effects of multipath, and improves the listeners experience! Also, our efforts contributing to the AES work on X192 will bring standardization to AoIP (Audio over IP) technology.

It would be easy to ramble on about our past, and those who influenced us. But, if not for you, our customers, and the many of you whom are trusted customers, we'd be nothing more than a pipedream. We've been very fortunate to work with the best in the whole world. I'm continually humbled, and honored by this. Thank you, for what we've been able to do in the past, and thank you for the opportunity to offer you our best efforts as we move forward!

Sincerely,

**Frank Foti**
Il Padrino, The Telos Alliance

# BROADCAST AUDIO QUALITY

## A LETTER FROM OUR CTO, TIM CARROLL

Remember when those words also described audio of the highest caliber? Today we are surrounded by a growing number of regulations in radio and television designed to protect listeners and viewers. Although well-intentioned, they are arguably misguided and can have the negative side effect of shoving quality to the backseat in the name of hitting some target.

At first glance, dealing with audio quality may seem at odds with day-to-day challenges. It is difficult enough to deal with normal daily tasks and maintain enough staff to do so, let alone having staff that has the time and sufficient training to handle the details of audio quality.

However, in these hyper competitive times when ratings are the bellwether of success, it is imperative to understand that there is a valuable difference between a pacified consumer and a satisfied consumer. The former will likely not complain, but the latter will go above and beyond and spread the word about their good experiences and return to that broadcaster for more of the same. This is also the difference between compliance and quality.

Too often forgotten is the basic idea that it is the responsibility of our entire industry to balance consumer satisfaction with realistic protection of artistic intent. When we get it right, both sides are happy and compliance takes care of itself.

So, as The Telos Alliance focuses on refining existing products and adding some new ones, many of which can be seen in the pages of this catalog, we are also focusing on how these products can work together to ease the increased burden of compliance by improving broadcast quality. Our goal is to not only create individual products and technologies that are excellent in their own right but to create systems of products, both inside and across our brands, that lead the industry forward.

We look forward to the continued dialog with our customers and the challenge of raising the bar higher. Together we will continue to redefine broadcast quality and make consumers exceptionally satisfied. And, make regulators bored.

With my best regards,

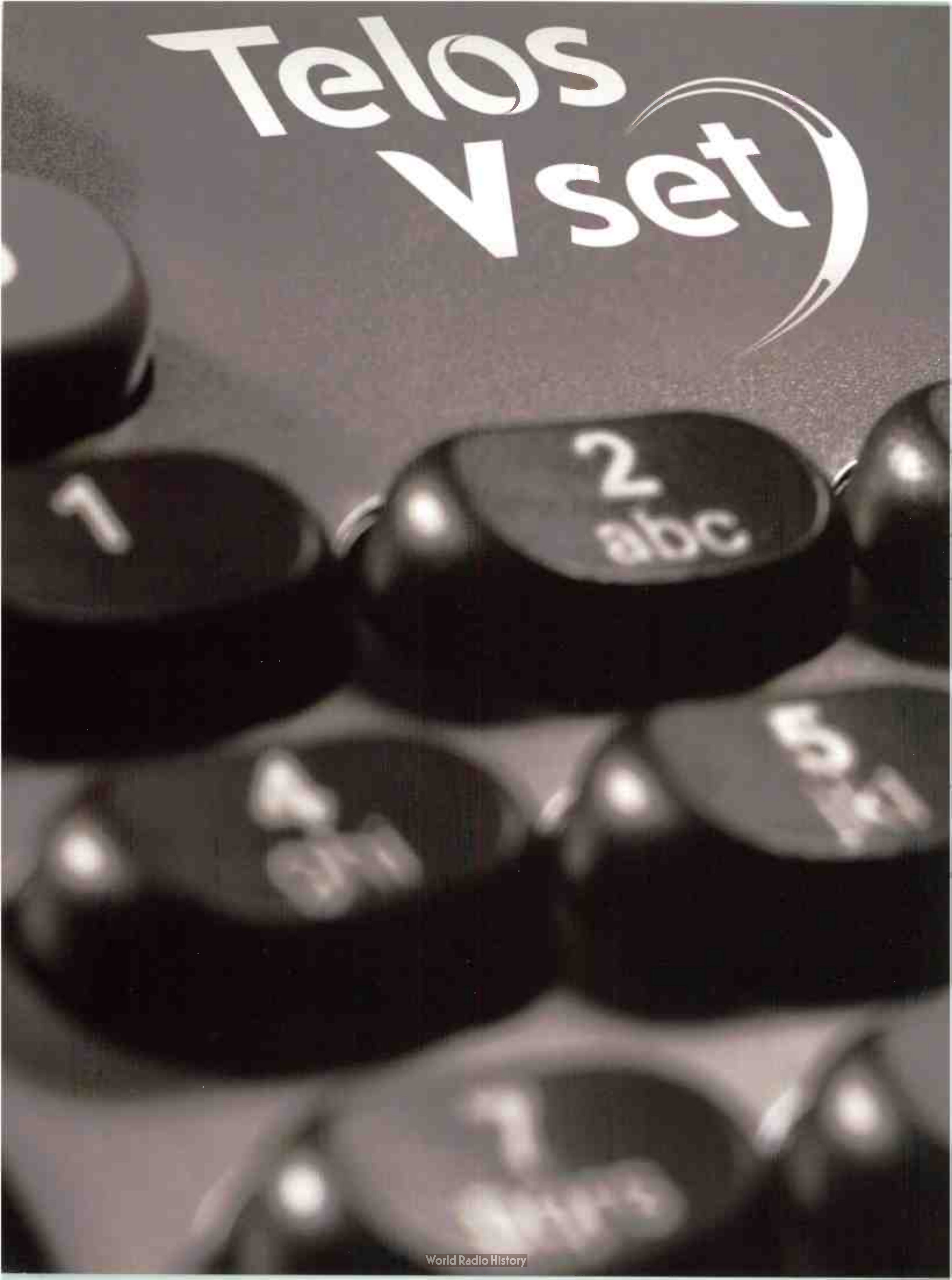
**Tim Carroll**
CTO
The Telos Alliance

# NOW!
## TABLE OF CONTENTS

## AXIA



## LINEAR ACOUSTIC


Telos
Vset
1
2
abc

## TELOS

The most compelling audio content and programming originates with the human voice. Pundits, perpetrators, experts, celebrities, newsmakers, victims, moderators, and listeners - they all have opinions. Many want to be heard. And listeners want to hear them.

It's for this reason that voice quality over the challenging and changing Public Switched Telephone Network is an engineering passion for us. Telos' founder, the late Steve Church, introduced Digital Signal Processing to broadcasters for the first time with the revolutionary Telos 10 phone hybrid. Since that invention, Steve and our engineering/development teams have brought multi-line talkshow systems, ISDN capabilities, and even large-scale T1- and PRI-based talkshow systems for multiple studios.

Now that Voice Over IP is becoming commonplace, broadcasters benefit again from a Telos innovation - the Telos VX talkshow system. VX connects directly to VoIP over SIP connections and uses Livewire AoIP to bring those clear caller voices right into your audio console or production workflow. VX's IP connectivity is allowing broadcasters to upgrade to "HD Voice" quality for on-the-spot reports and priority listener calls.

IP connections are quickly becoming the go-to standard for broadcast remotes, studio-transmitter links, and audio content distribution. Whether for main or backup service, permanent or short-term use, Telos codecs are built for professional use and jaw-dropping audio quality. The Telos Z/IP ONE is designed to work well over the Public Internet, and includes Agile Connection Technology, enabling fast responsiveness to changing and challenging bandwidth conditions. The Z/IP ONE is perfect for remote talent, with low-latency and simplified, two-button connections. Z/IP ONE also affords super-reliable operation for STL and other program links; and works over any IP link: IP radios, fiber, Internet, or private WAN.

On the pages that follow, you'll see tools, equipment, and systems designed to connect your listeners with compelling audio content. They'll connect your talent with listeners, experts, and events with hardly a second thought to the amazing technology inside.

Thank you for your own dedication, ideas, and comments. Please tell me how you're creating compelling content with Telos, and how we can help you do that even better!

My best,

**Kirk Harnack**
Vice President and Executive Director
Telos Systems

# TELOS VX

## THE FUTURE OF BROADCAST TALKSHOW SYSTEMS


### 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.

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.

## FEATURES AT A GLANCE

There are lots of VoIP systems for business, but only VX is built with broadcasters in mind. Here are a few of the benefits VX brings to your studio:

» Works with any type of phone lines - POTS, T1/E1, ISDN and SIP Trunking telco services - via standard gateways 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, resulting in cost savings and a simplified studio infrastructure. » Connection of up to 100 control devices — phone sets, console controllers or call screening software — 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. » 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 lets you instantly reassign call-in lines to studios where they're needed most. » 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. » Drop-in modules can integrate VX phone control directly into your Axia mixing consoles. » Clear, clean caller audio from fifth-generation Telos 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.

# TELOS VX COMPONENTS

## BUILD YOUR IDEAL PHONE SYSTEM

### VX ENGINE

The VX Engine is the heart of the system. A fan-free 2RU rack-mount device with enormous processing power, the VX Engine provides all the call control and audio processing needed for the system. VX is Web-based, so remote control and configuration are a snap – you can work with it from any place you can get online.

The VX Engine's platform is so powerful, it provides a hybrid for every line, allowing multiple calls to be conferenced and aired simultaneously with excellent quality. Incredibly advanced DSP hybrids make caller audio sound its best, no matter what kind of line or phone the caller uses. Smart AGC coupled with Telos three-band adaptive Digital Dynamic EQ and a three-band adaptive spectral processor are part of the toolkit; send audio gets a frequency shifter, AGC/limiter and FhG's Advanced Echo Cancellation technology to eliminate open-mic feedback. Call ducking and host override round out the package.

With VX, choice comes standard. Want to use traditional phone services, like T1/E1, ISDN, and POTS? The VX Engine works with standard telco gateways from Patton, Cisco, Grandstream and others. Want to use a VoIP-based PBX or SIP Trunking telco service? VX Engine uses standard SIP (Session Initiation Protocol) and RTP (Real-time Transport Protocol).

## VSET 12

The VSet12 phone controller is an IP-based phoneset with two large, high-contrast color LCD panels that provide line status and caller information using easy-to-understand Status Symbol displays. Use them like a traditional Telos controller; select, hold and drop calls the same way you've always done. You can also map phone lines to individual faders for greater control — even assign a group of lines to a single fader. Additional controls lock calls on-air, start external recording devices, and queue calls for sequential airing. There's a built-in address book and call history log, and the display screens deliver detailed line status, caller information, caller ID and time ringing-in or on-hold – perfect for error-free production of fast-paced shows. And VSet hookup is easy, too: just connect it to your network switch with an Ethernet cable and you're done.

## VSET 6

The VSet6 phone controller is a six-line version of the VSet12. Like its big brother, it has a large, friendly color screen with animated Status Symbol icons, and controls for 6 phone lines. With all the control functions of the VSet12, it's great for secondary studios or other locations where only six lines of control are needed.

## VSET 1

The VSet1 phone controller provides convenient single-line access to your VX system in news booths, voiceover stations, etc., where control of multiple phone lines is not necessary. Its bright display screen and intuitive controls let operators easily hold, drop and step through queued calls.

## CONSOLE CONTROLLERS

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. Wouldn't it be great if talent could take control of phones without ever having to take their focus from the board?

They can: Axia Element and iQ IP-Audio consoles can be outfitted with built-in phone controllers. This sophisticated integration helps shows run smoother, since phone controls fall immediately to hand. Talent enjoys phone controls right on the board to dial, answer, screen, and drop calls without ever diverting attention from the console.

And, since the console now communicates directly with the phone hybrid, mundane tasks such as mix-minus generation, recording device activation and playback of pre-recorded conversations can all be automated, allowing talent to focus on doing what they do best — their show.

JOE MAUK, PEAK BROADCASTING

"VX is exciting technology that scales to support facility clustering. In our environment, VX connects to an Asterisk PBX, the 'firewall' between it and our PRI/VoIP services. Integration of VX and Asterisk was seamless...Thanks to embedded Omnia processing, call quality is superb. Even low bit-rate cell calls are greatly improved."

## VX INTERFACES

VX Audio & 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.

## PhoneBOX VX By Broadcast Bionics

Broadcast Bionics asked if they could use the VX as a platform for their popular PhoneBOX product. It took us about a millisecond to say "yes!" Partly because we believe in open standards and the benefits of partnerships. But also because we think PhoneBOX is very cool!

Starting with PhoneBOX VX Lite, free for all VX customers, PhoneBOX VX scales to cover the most demanding of formats and the largest of networks. PhoneBOX VX harnesses the power of the Livewire network. It offers stunning visual talkback. It integrates phones with email, SMS & social media. It's powerful and scalable, offering multi-studio control complete with SQL caller database, call history, call recording & editing, codec and console integration, all standard.


## VX + By Broadcast Bionics

Putting a VX phone system into a really large plant? VX+ from Broadcast Bionics allows you to link multiple VX Engines together to provide the power, scale and resilience required for even the most demanding of installations. VX+ facilitates networking and transfer of shows between multiple sites, and adds enhanced functionality to VX phone systems, including outgoing announcement and voice mail capability. And it doesn't need exotic hardware: VX+ simply runs as a service on Windows PCs.

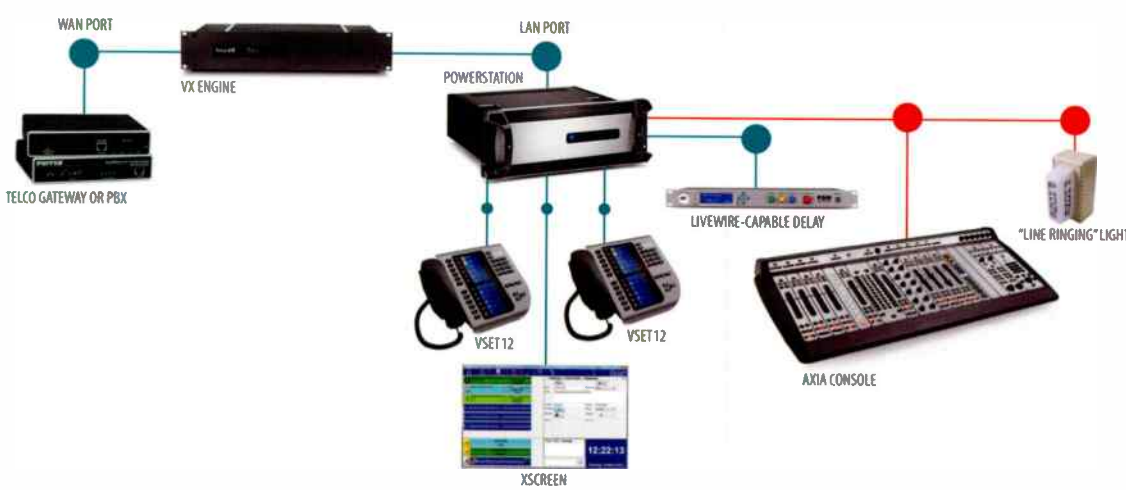
# PAINLESS HOOKUP

## READY. SET. BROADCAST.

### AXIA INSTALLATION

Telos VX is a "facility wide" on-air telephone system. That means multiple studios, multiple stations, and multiple shows with minimal hardware requirements. VX eliminates the maze of discrete cables once required by a multi-line talkshow system. All VX components are linked with Ethernet, so a single CAT-5 cable provides connection to the telco interface, line switching commands, data communication between the VX Engine and VSet phones, transport of caller audio to mixing consoles, return of mix-minus and program-on-hold audio to the caller, data messages (such as call notes and IM) between producer and talent, Livewire audio call recording, and transfer of recorded call files from the producer to the studio.

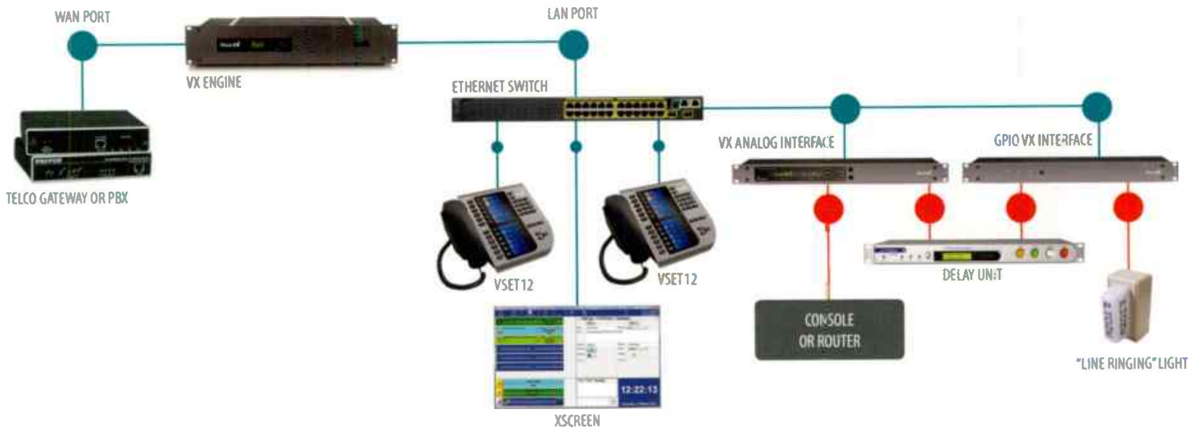
Telco is delivered via IP through a POTS, ISDN or T1 gateway device, a SIP PBX, or a dedicated IP circuit using SIP Trunking. Got an Axia Livewire AoIP studio network? Telos VX will plug right in. Audio inputs and outputs are Livewire real-time audio channels and travel over your existing Axia system just like the rest of your audio. Axia console GPIO ports can be used for "phone ringing" tallies or remote control of profanity delay units. It's the seamless integration of studio phones, mixing consoles and routing network you've dreamed about! The diagram below shows just how easy it is to combine VX with your Axia network.



COX RADIO, TAMPA, FLORIDA

## NON-AXIA INSTALLATION

Don't have IP-Audio networking yet? Don't worry: VX will work with all console brands, networked or not, via VX Audio and Logic interfaces – compact 1RU breakouts that put multiple I/O channels right where you need them. This diagram shows a typical studio with an analog mixer, using VX Analog and GPIO logic interfaces to connect the console and other broadcast equipment.



# TELOS Nx TALKSHOW SYSTEMS

## CLEAN, CONSISTENT CALL QUALITY — EVERY LINE, EVERY TIME.


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, 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 the Console Director drop-in module. 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.

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, and 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.

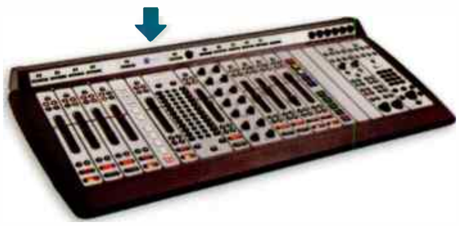
Nx systems will work happily with either POTS, ISDN-S, or ISDN-U lines — just tell us when you order. Or, split the difference: with Nx12, you can specify half POTS, half digital line interfaces.

## DESKTOP DIRECTOR

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.

## CALL CONTROLLER MODULE FOR ELEMENT AND IQ CONSOLES

Call Controller modules for Axia Element and iQ mixing consoles let talent take command of Nx6 or Nx12 right from the board, making for smoother, more error-free shows. Two rows of hybrid controls with easy-to-read Status Symbols icons are flanked by dedicated faders so talent can make, take and bring callers to air without ever taking their eyes off the board.



## XSCREEN By Broadcast Bionics

VX, NX & iQ6 come complete with XScreen from Broadcast Bionics. When they asked if they could use these products as a platform for their new XScreen product, it took us about a millisecond to say "yes!" Partly because we believe in open standards and the benefits of partnerships, but also because we think XScreen is very cool. XScreen'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 VX 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 plus more. XScreen, deployed as part of a Livewire network, also enables call recording, editing and console integration directly over the network.

# TELOS Hx6

## GIVE YOUR PHONES AN INSTANT UPGRADE


Say hello to Hx6, the new, advanced six-line broadcast phone system from Telos. Equipped with powerful Telos 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, whether song requests or morning show phoners, or intensive call-in 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 third-generation 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 spectral processor, analyzes and adjusts received audio spectral characteristics so that calls sound smooth and consistent despite today's wide variety of phone sets and connection types. Adjustable Omnia smart-level AGC with noise gating provides spectrally consistent audio from call to call — even on notoriously tough cellular calls. A sophisticated caller override allows precision adjustment of the degree to which talent audio "ducks" the caller audio, and exclusive feedback reduction functions help eliminate open-speaker howl.

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

Around back, you'll find audio I/O, GPIO, and Telco connections. Hx6 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; there's also an Ethernet port on the back panel for connecting from 1 to 6 Telos VSet6 or other phone controllers. This Ethernet port is also Livewire-qualified for connecting to Axia consoles and networks. Additional features include a Program-On-Hold input, GPIO connections for speaker muting or ring tallies, and Ethernet-based remote setup and administration.

The Telos VSet6 six-line phone controller is nothing short of beautiful (if we say so ourselves). It's an IP-based phoneset with a large, high-contrast color LCD panel that provides line status and caller information. There's almost no learning curve; VSet phones work like traditional Telos controllers, with calls selected, held, and dropped in the way to which operators have

TELOS VSET6 SIX-LINE PHONE CONTROLLERS

Beautiful new Telos VSet6 six-line phone controllers with large, colorful VGA LCD displays provide Hx6 users with 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, ensuring that phone segments are always smooth and error-free.

grown accustomed. Exclusive animated Telos Status Symbol™ icons show producers and talent at-a-glance line and caller status. Talent can also control Hx6 using controls built into Axia consoles, which means faster, more precise phone segments — since operators' eyes never need to leave the console.

For PC-based call screening, Hx6 comes complete with XScreen Lite from Broadcast Bionics. XScreen Lite's interface gives screeners and hosts tons of information and control using sophisticated visual talkback, a drag and drop caller database, and visual warnings for persistent or nuisance callers.

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.

## FEATURES 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 incoming call types. » 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. » Precision, adjustable caller override. » VSet6 phone controllers with VGA LCD displays and Status Symbols for fast, intuitive operation. » Caller ID for both analog and digital telephone connections. » Standard analog and Livewire IP-Audio I/O, plus optional AES digital audio I/O.

# TELOS iQ6 TELCO GATEWAY

## MULTI-LINE PHONES MADE SMARTER

A multi-line phone system that connects to your console with just one cable? Smooth, detailed caller audio — even from cellular callers? Meet iQ6, the dedicated telco gateway for Axia iQ consoles.

iQ6 plugs right into your Livewire AoIP network, saving you money and time by eliminating the cost and labor of old-fashioned discrete I/O, cabling and soldered connectors. iQ6 transfers audio, hybrid control and mix-minus for six phone lines over a single CAT-5 cable. Setup is easy: plug it into your Axia network, do some fast web-based configuration, and voila! You're taking calls.

Talent can control iQ6 right from their Axia iQ console — there's an iQ Telco expansion frame that puts hybrid control and Status Symbols information icons right on the mixer's surface, so talent never has to take their eyes off the board to take a call.

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

How does iQ6 sound? Like a Telos, of course! Inside, two of our most advanced hybrids handle up to six 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 high-quality conferencing between callers at the touch of a button — no external equipment needed.

# Hx1 AND Hx2 DIGITAL POTS HYBRIDS

## POTS PHONES NEVER SOUNDED SO GOOD

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

We've pioneered 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.

Hx1 and Hx2 have Auto-Answer, caller disconnect detection, sophisticated new audio-leveling and anti-feedback routines for enhanced open-speaker applications, call screening and line-hold features, and front-panel send and receive audio metering — plus 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. Studio adaptation and a subtle pitch-shifter help prevent feedback in open-speaker situations. 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.

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.

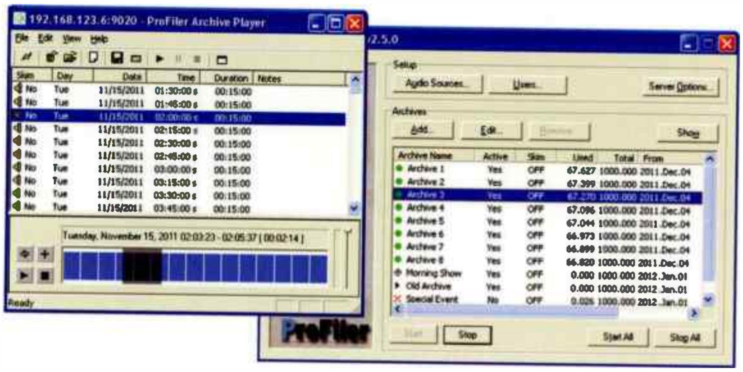
Around back, you'll find a switchable mic/line input, balanced analog receive-out output, RJ ports for telco input and phoneset, input level adjustment, and a DB9 remote control connector with GPIO closures for hybrid control and status indicator lamps. Need digital I/O? No problem — Hx comes in an AES/EBU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

BLAKE THOMPSON, WZIP-FM

"Telos' Hx1 hybrid is all new, and with the built in processing, is one of the best I've heard. I have it in 2 stations and both are VERY happy with the improvement...Save the money AND get better sounding calls!"

# TELOS PROFILER

## MULTI-CHANNEL, MULTI-STREAM AUDIO ARCHIVING FOR YOUR PC




Telos ProFiler is the efficient, set-and-forget way to automatically log your radio station's program audio. Who needs tape decks or expensive logging hardware? ProFiler runs on a Windows PC, and produces time-stamped audio archives you can listen to on standard MP3 players.

Each ProFiler package includes a Telos audio interface with pro-level balanced audio inputs and buffered GPIO inputs that you can use as a start/stop trigger, or to let an active microphone trigger a high-quality capture mode. The starter kit lets you log one stereo or two mono streams; add more audio interfaces to capture as many as eight mono streams simultaneously on a single PC. ProFiler 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.

**JOE BURKE, LIFT FM RADIO NETWORK**

"Just wanted to give you a heads' up regarding our Telos ProStream. We are forever grateful to you...the sound quality is outstanding! Thank you for making our streaming audio sound better than most over-the-air radio broadcasts!"

# TELOS PROSTREAM

## THE DO-IT-ALL, ONE-BOX WEB STREAMING SOLUTION


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 delivered.

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.

Broadcasters know Telos is the codec expert. ProStream puts all our expertise into an 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 a Shoutcast, Wowza, Icecast, Live365 or Adobe Flash Media server for distribution.

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 ¼" 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.

## FEATURES AT A GLANCE

» 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 and Live365 streaming services.

MIKE SCHWEITZER, IMPROBABLE MISSIONS FORCE

"You can have my Zephyr Xstream when you pry it from my cold, dead fingers!"

# ZEPHYR XSTREAM

## THE BEST WAY TO HEAR FROM THERE™

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.

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.

## FEATURES AT A GLANCE

» 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. Exclusive Error Concealment technology prevents occasional network glitches from being heard. » Low-Delay MPEG AAC-LD coding: crystal-clear audio quality and greatly reduced encoding delay for smooth, natural bidirectional 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. » 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 standard G.711 phone calls right from your Zephyr Xstream

# ZEPHYR XPORT

## THE "GO ANYWHERE" ZEPHYR

When you're traveling to an important remote, "small and light" is the name of the game. Of course, excellent audio quality is a must-have, too. Zephyr Xport is the on-the-go broadcast codec that wins on both counts.

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 that 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 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. Got to get to a remote on the double? Grab an Xport and go!

## FEATURES AT A GLANCE

» Superior sound: aacPlus™ coding gives you the best-quality audio of any POTS codec, even at low bit rates. » ISDN capability, standard. Plug Xport into any POTS jack and dial your studio's Zephyr Xstream; plug into digital phone lines when available for ultimate flexibility. » Easy, intuitive operation: even the greenest intern can have a remote broadcast up and running in minutes with Zephyr Xport. » 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; Xport has an internal auto-ranging power supply that works anywhere in the world. » Built-in mixer with mic and line inputs and selectable DSP processing by Omnia for sweetening when you need it. » 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.

DAVE ANDERSON, JOY-FM

"The difference between satellite feeds and Z/IP ONE is amazing. Even more impressive though, is the difference between raw audio and Z/IP ONE. There is none."

# Z/IP ONE

## THE IP CODEC THAT DROPS JAWS. (NOT AUDIO.)

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.

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.

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. Around back, you'll find analog XLR ins and outs, a Livewire port for quick connection to Axia networks, separate LAN and WAN jacks for safe connection to "the outside world", and a parallel port for GPIO contact closures. All in a compact, 1RU package.

## FEATURES AT A GLANCE

» 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. Optional Enhanced aptX codec. » 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 geo-locations, 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.

# ZEPHYR iPORT

## LIVEWIRE TO IP MPEG GATEWAY

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

A pair of Zephyr iPorts 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, iPort is perfect for audio transmission over VPNs, satellite links, Ethernet radio systems, and Telco or ISP-provided QoS-controlled IP services. 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 iPort to generate multiple channels of MP3 or AAC for Internet streaming, broadcasting to mobile phones, and audio distribution systems... the possibilities are limited only by your imagination.

iPort uses 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 with Axia xNodes, or a Telos VX analog or digital audio interface to make a standalone high-density audio codec package.

Zephyr iPort 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 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 delivers more power than a legacy cardframe design, at only a fraction of the cost.

# AUDIO RELIABILITY OVER THE PUBLIC INTERNET:

## DESIGNING ROBUST IP-STREAMING FOR OUTSIDE BROADCASTS, STL, AND PROGRAM DISTRIBUTION.

### ABSTRACT

One of the challenges in transferring audio programming over IP networks, especially over the Public Internet, is data loss caused by two factors: congestion-related packet loss and varying latency or jitter. IP links may drop packets for several reasons; though some transmission protocols are designed to mitigate or correct such losses, they require extra bandwidth and extra time to make these corrections. In this paper we examine several data transmission error mitigation techniques in the context of their application to real-time, low-latency IP-audio transport. We suggest how different techniques may be applied to different data loss or jitter scenarios. And, we assist the reader in analyzing his own data transmission path, characterizing any difficulties, then selecting the best technical solution to mitigate or eliminate any final effects on the delivered audio product.

### INTRODUCTION

Many radio engineers are wary, even skittish, about relying on an Ethernet/IP network as the conduit for critical program audio. Perhaps too many unresponsive "Print" commands, unresponsive web pages, IP networking confusion, and wonky Internet connections have led to this skepticism about real-time, low-latency audio transmission over IP.

In recent years, however, we've come to accept and even embrace localized Audio over IP (AoIP) within and among local studios. Here we have end-to-end control of the network environment. Local Audio over IP (AoIP) is enabling low-latency, Ethernet/IP-connected audio at about 4,000 radio studios at this very moment. Localized AoIP works very well when all the equipment, wiring, and configuration are under our purview - our direct control.

### CHALLENGES OUTSIDE THE LOCAL STUDIO

What typically happens when we encode audio data into IP packets, and then hand those packets over to a third party for transport? What's the impact to IP-Audio when Real Time Protocol (RTP) packets must compete with a rush of other packet traffic, either outbound or inbound? And, how can we optimize an AoIP system for end-to-end low-latency, while assuring no perceivable audio dropouts?

For a couple decades or more, radio engineers have used ISDN, T1 and E1 circuits, satellite links, and various RF point-to-point solutions to transport audio to and from the studio. While not without their own issues, at least these telco-provided services are tariffed and come with a basic guarantee of performance. And, the RF links tend to be under the engineer's direct control with no ambiguities.

More recently, economics and availability are often dictating that we use third-party IP links in one form or another to transport real-time audio. There's a perception that such links are inherently less reliable than the technologies they replace. And while that may seem true anecdotally, it doesn't have to be.

Private or dedicated IP transport connections can be every bit as reliable as older technologies. They're usually less costly, available from multiple, competitive providers in the same area, and are far more flexible in terms of capability, utility, and pricing. Moreover, private or dedicated IP links usually offer one or more forms of Quality of Service (QoS) as well as optional Service Level Agreements (SLAs), placing reliability as high as other, older technologies, and offering the flexibility to mix data types that older tech couldn't do. Private or dedicated IP connections such as these rarely require extraordinary technology from the endpoint equipment - in this case the IP-Audio codecs. Over an essentially perfect and known IP connection, even the most basic and static IP codecs can work reliably.

This paper concerns the use of IP codecs over imperfect links. Such links usually include the Public Internet, but also encompass congested and variable wireless links like 3G and 4G, WiFi, and WiMax services. We must also include the data impairments that occur within a Local Area Network (LAN) as competing packets are routed to and from the local Internet Service Provider (ISP) and on to the Public Internet.

IP-Audio is reliable and robust in a controlled network environment. Latency can be very low with easy routing of channels and superlative operational conveniences. Our challenge - and opportunity - comes with getting excellent audio performance across highly-trafficked, imperfect links that we don't control. And that is where clever technology steps in.

Regular (non-real-time) IP traffic that must be one hundred percent reliable - bit for bit reconstructed at the receiving end - is commonly transported using TCP/IP. This protocol can assure that, eventually, there's a perfect transfer of data across any usable network, as long as it doesn't matter how much time the process takes. Whether a file downloads in five seconds or five minutes is of less consequence than making sure the file is one hundred percent complete and bit-for-bit identical to the source. TCP/IP will slave away, requesting and re-requesting a complete and error-free transmission, packet-by-packet, from the far-end source. TCP/IP doesn't give up until every bit, byte, and packet is transferred, no matter how long it takes, notwithstanding wholesale connection timeouts. If you've ever downloaded a huge file over a slow connection, you've experienced the relentless robustness that hallmarks TCP/IP.

"No matter how long it takes", then, is precisely why TCP/IP is not appropriate for real-time audio or video. Indeed, transferring real-time media and its metadata over a lossy, jittery, packet-switched network appears counter-intuitive on its face.

One-way media distribution, such as music streaming or video entertainment over IP can use TCP/IP, however. Thanks to a large receive buffer, playout applications can request and buffer upwards of 30 to 60 seconds worth of streaming data, then back off further requests until needed to keep the buffer full. The application meters out the buffered data in real-time, but only locally to the user.

For the balance of this paper, we'll refer only to audio over IP and it metadata, but similar concerns and solutions apply to real-time video streaming as well.

Meaningful audio performance for most radio broadcast uses implies two-way audio and low delay while maintaining the highest possible audio quality. When a two-way, low-delay audio connection is desired, there isn't time for TCP/IP's handshaking and retransmission. Neither is there the security of large data buffers to even out the flow of data at our applications' user interfaces.

The protocols of choice for most real-time audio streaming scenarios are User Datagram Protocol (UDP) and Real Time Protocol (RTP). UDP is free of time-consuming handshakes. It's a one-way stream of data, sent at the request of the receiving end. RTP offers synchronizing, timing, and prioritizing information, useful to keep the receiver's playout in sync with the encoder.

A typical IP-codec transport path will present two major delay components and several minor ones. Cumulatively, these delay components comprise the total one-way audio delay. Those major contributors to delay are the summed encoding/decoding, or "codec" delay, the packet's network transit time, and the appropriate receive jitter buffer delay.

Minor delay contributors include the codecs' A/D/D/A conversions or Sample Rate Converters (SRC's), along with the audio handling delay of the underlying operating system (OS), and packetizing/depacketizing delay.

We'll see how different approaches toward our goal may be used individually, or perhaps together, to mitigate the data impairments presented in typical IP networks.

One only need glance at a histogram of dropped or delayed packets in an IP-Audio stream to identify the challenges that a given IP connection will present.

Following are two histograms showing test results from different connections to the Public Internet. The test shows upstream packet jitter between the testing computer and a server.

Figure 1 shows a test via a hard-wired Internet connection in a low-congestion environment. Packet jitter is 2 ms or less. This is quite good and likely to present a very usable path for high-quality IP-audio carriage, assuming similar performance over a longer time horizon.

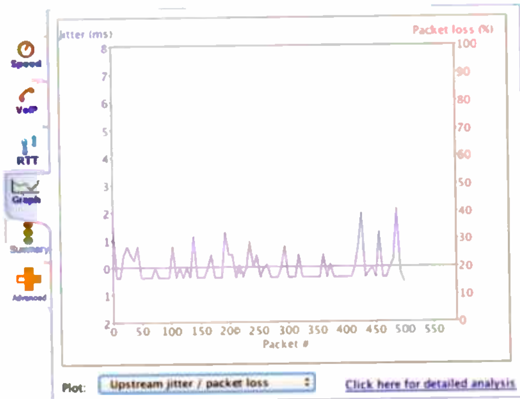



Figure 1

Figure 2 was conducted with the same PC and the same software, but the Internet connection is via a wireless WiMax service. The packet jitter reaches nearly 40 ms. This level of jitter is still usable, but will require additional buffering in the receiving IP-audio codec. 40 ms typically represents two packets of compressed audio data. Using a connection with 40 ms of jitter, we would be wise to set at least 80 to 100 ms of additional buffer time.

If a given IP path is exhibiting large amounts of or large variations in jitter, we should endeavor to find out the cause. High jitter figures often indicate an overloaded link or a misconfigured router. Jitter may also indicated simply a poor physical connection somewhere and the TCP/IP protocol is tirelessly working to deal with it, resending packets until they successfully reach their destination.

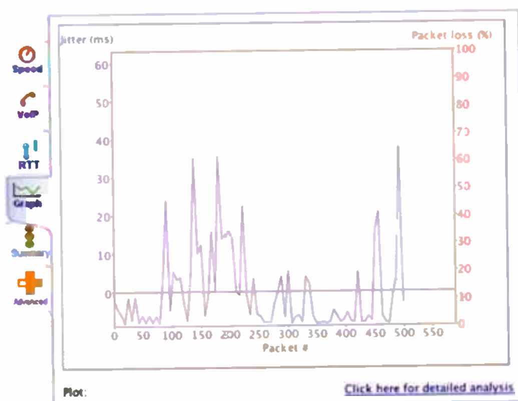



Figure 2

The tests shown above are parts of a comprehensive suite of free tests intended to troubleshoot Voice over IP (VoIP) difficulties, or to pre-qualify an IP connection for VoIP service. The web URL is: http://www.whichvoip.com/voip/speed_test/ppspeed.html. VoIP service and high-quality IP-audio are rather similar, so these tests are useful to broadcast engineers, at least for testing from a local Internet connection to the test servers.

Next we'll examine different approaches to making IP-Audio more reliable. Some work over less-than-perfect IP connections, while others may use two separate connections to be assured of a good connection by dint of aggregation.

## APPROACHES TO RELIABLE IP AUDIO

### Use Perfect IP Networks

Employing a more-or-less perfect or wholly transparent IP Network is certainly simplest from a codec design perspective. Over a perfect IP network, codecs don't have to be smart in any way; they simply need to be compatible. A perfect IP network will transport IP-audio packets from the encoder to the decoder with no packet loss, no packet mis-ordering, very low latency, and sub-packet-interval jitter performance.

Well-managed private Wide Area Networks (WANs) can deliver this network experience, especially with the assistance of end-to-end Quality of Service (QoS) implementation. QoS is often achieved though Multi Packet Label Switching (MPLS) and assures that the desired packets are handled first at each network device. Perfect or near-perfect IP networks may be an option for program contribution/distribution networks within the same corporate infrastructure, as well as critical point-to-point audio transmissions.

### Generate Fully Redundant IP Streams

If one IP-audio stream is mostly reliable, then adding another in parallel for redundancy should prove exemplary. It's possible to design a codec to send identical IP-audio streams through two different physical Ethernet interfaces. Ideally, these distinct interfaces are networked through disparate equipment connected to separate WAN or Internet providers. At the far end, the same disparate and redundant connection scheme completes the provisioning of two thoroughly redundant IP transport paths. Due care is taken to ensure that two different local Internet providers are not themselves connecting to the same Internet backbone. After all, partial redundancy is not true redundancy.

A benefit of enabling, within a single bi-directional codec, use of redundant IP transport paths is this: The codec's decoder can seamlessly switch between incoming packet sources. Shown in Figure 3, if a packet - or two or ten - are missing from Stream and Path "A", then replacement packets should be timely available from Stream and Path "B". We configure the decoder's receive buffer to accommodate the worst-case latency offset plus jitter from the more latent of the two streams. Then the decoder can be configured to simply replace missing, late, or defective packets with good ones from the redundant stream, affording no interruption in the decoded audio.

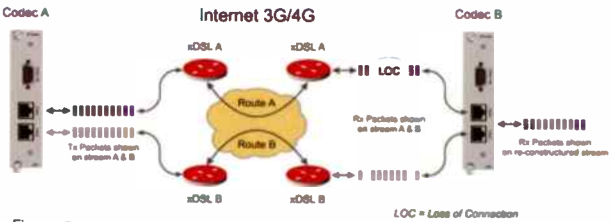



Figure 3

However, even with redundant data paths this approach isn't truly redundant unless there are two completely disparate codecs at each end. This gives rise to considering how much redundancy is enough.

An alternative to purpose-built redundant-interface codecs is to use one single-interface codec at each end, then using Virtual Private Network (VPN) technology, create IP tunnels over separate IP transport providers. The work of creating and then switching between two data paths is left to the IP routers, which can be inexpensive. Note that this router-based data path redundancy offers switching of the data transport path, but not packet-by-packet replacement.

The redundant IP stream approach to reliability can also offer some degree of benefit even when implemented over a single connection. Assuming adequate end-to-end bandwidth, sending the same data twice over a data path characterized by only occasional, single-packet dropouts can benefit from having a replacement stream readily available, especially if we can also include some temporal diversity.

The advantages of sending and receiving redundant IP-audio streams fully depends upon having the requisite bandwidth to do this. One requires either disparate data paths or a single data path that's twice as big as would be needed with single-stream transmission. When using wireless connections, or a smaller DSL connection, or any connection shared with other users, there's the likelihood of instances when the real-time audio packets will have wait or be dropped by a router. In some situations one must ask, "If the end-to-end system lacks reliability due to occasional bandwidth bottlenecks, then how will sending twice the data , causing more congestion, be helpful?"

### Generate Temporally Diverse Redundant IP Streams

What if the bandwidth is available for redundant streams over a single data path, but we want to obtain more useful redundancy though time diversity? This question gives rise to the approach of sending redundant streams that are separated in time by some appropriate amount, depicted in Figure 4. If one analyzes a given data path and discovers regular or irregular short-term flow interruptions or jitter anomalies, then temporal diversity can provide a solution.

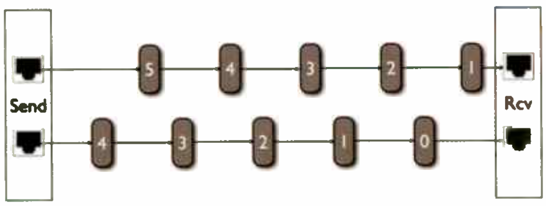


Figure 4

The delay of the 2nd IP-audio stream would necessarily be slightly longer then statistically significant interruptions or periods of increased jitter. The receive buffer would also need to be slightly longer than the stream offset.

Radio engineers familiar with HD Radio transmission know that this data is sent using both frequency and temporal diversity to mitigate multipath and shadowing effects, especially in mobile reception environments.

One codec manufacturer is experimenting with sending packets out of order in the redundant stream, then setting the receive buffer for the maximum random offset of a redundant stream packet.

### Employ IP Forward Error Correction

Forward Error Correction (FEC) as understood by many broadcast engineers, is best applicable to non-packetized serial data. Such has been the case with digital satellite transmission and other serialized data transports. Noise burst or other data loss episodes are expected to be very short-lived in this kind of data path. As such, one can reasonably apply Reed-Solomon Convolutional coding and recover these short-lived data losses appearing at the decoder.

IP-audio is packet-based. Typically, an entire packet will be lost - or several packets in quick succession. Even partial packet loss implies loss of the entire packet. Either way, much data is lost, and recovery using common FEC techniques requires the FEC data be spread over quite a few packets. As such, one must employ a transmission buffer as well as a receive buffer. Overall delay increases dramatically and could be in the hundreds of milliseconds when aggregated. Clearly, employing FEC becomes counter to our goal of low-latency reliable audio transmission.

Some research is ongoing employing Reed-Solomon Erasure codes in both single and double column RSE algorithms. Packet losses of up to twenty percent are showing promising recovery rates with only a twenty percent increase in overall data rate for the FEC data. Still, the costs of such application of clever FEC algorithms are both additional time delay and computational complexity.

Dynamically-Controlled IP Stream Management Theres's a sharp reality about employing an IP data path for real-time data: the jitter and available bandwidth can be quite variable from moment to moment.

Unless some automatic encoder bit-rate control is implemented, the date path must be over-provisioned and, hence, under-utilized. Additionally, the codecs' receive buffers must be configured for the worst-case packet jitter. In other words, both usable bit-rate and buffer delay cannot be better than the worst-case, in order to maintain a reliable audio stream.

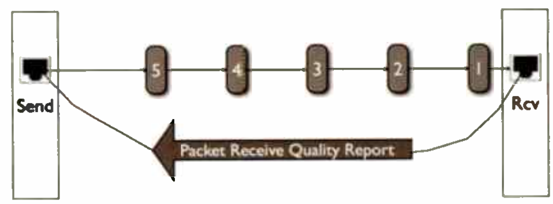


Figure 5

Said another way, a single IP data path can be used optimally only when real-time adjustments to receive buffer size and encode bit rate are managed dynamically by a smart algorithm. This basic feedback path is shown in Figure 5.

Keeping the goal of low-latency audio transport in mind, the encoder/decoder pairs are managed such that buffer size is persistently minimized, consistent with reliable audio decoding. However, if the buffer size reaches some acceptable maximum and some packets are still lost or untimely received, the encoder can reduce the encode data rate in increments until a reliable data rate is found. From time to time, the management algorithm will attempt to increase the encoding rate and decrease the buffer sizes within pre-configured limits.

A key element to dynamic management of the codecs is employing a coding algorithm that conceals occasional data errors or packet losses. The more recent AAC family of audio codecs do exactly this. Indeed, AAC's error concealment allows packet loss of five to ten percent with no obvious degradation to decoded audio quality. We may exploit this sophisticated error concealment as part of a comprehensive algorithm that dynamically seeks to set the receive buffer size just above the minimum required for low packet loss. An algorithm seeking the minimum acceptable receive buffer size can occasionally reduce the buffer to the point where packets are not arriving timely due to packet jitter in the IP data path.

### Selection and Hybrid Approach

Which of the preceding approaches is appropriate for a given operational scenario?

We consider the following factors and requirements:

- Required overall reliability
- Acceptability of short-term or long-term audio dropouts
- Use timeline - permanent or temporary
- Availability of IP connectivity
- Audio quality expectation

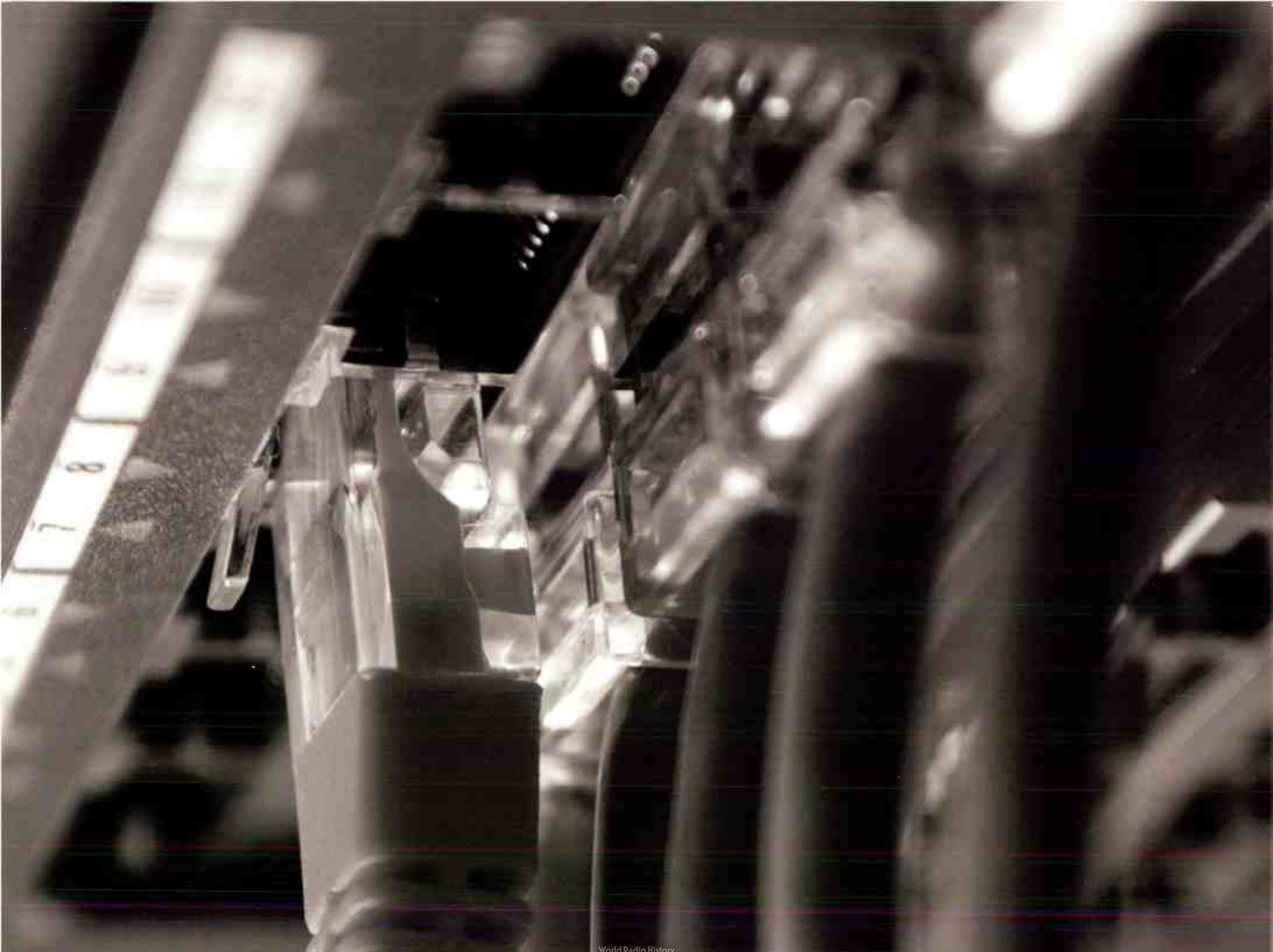
Ultimate reliability is best addressed by using two or more IP data paths. An IP codec designed with dual ports and appropriate software can do this, even to the level of packet-by-packet replacement. An external solution, such as a multi-WAN router can perform similarly in terms of reliability, except that switching will be on a path basis and not a packet basis. Such a multi-WAN router may be capable of switching among two or more WAN connections.

Permanent installations are made most reliable by appropriate port-forwarding through routers, allowing direct, peer-to-peer connections between codec pairs. Temporary use cases, such as outside broadcasts are assisted greatly with help from a "rendezvous server", making connection through unknown routers and firewalls convenient.

The availability of appropriate IP connectivity varies widely. Even where one form of IP connection, such as DSL or cable, is available, obtaining a backup may prove troublesome. Clever engineers are using inexpensive point-to-point IP radios to extend alternate IP connectivity to remote transmitter sites. In some cases, wireless 4G LTE service is serving as a backup path for permanent installations, and as the main (or only) IP connection for outside broadcasts.

Key to selecting the best options in IP codecs and connection schemes is understanding the nature of IP codec operation. Factoring technical requirements with operational goals and connection options will afford high-quality audio and reliable operation is nearly every scenario.

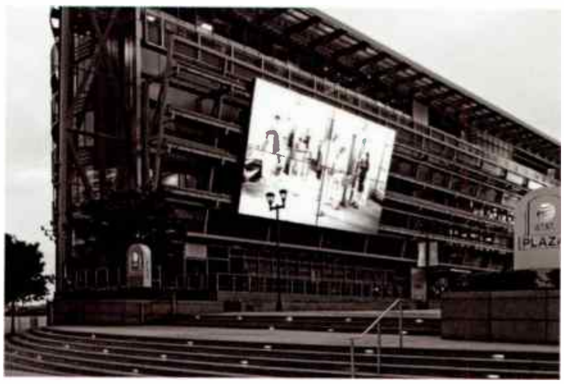
**Kirk Harnack**
Vice President and Executive Director
Telos Systems



# TELOS AND AXIA CONVERGE IN BIG D

## HOW IP-AUDIO AND BROADCAST VOIP POWER SIX STATIONS IN THE US' 5TH LARGEST MARKET.

Completing a cluster relocation in a major market is a major undertaking, even in the best of times. Completing a cluster relocation with a corporate merger in the middle of the project, well, that's taking it to another level altogether. Such was the challenge presented to Gary Kline, Senior Vice President of Engineering for Cumulus Media. Just as construction had begun on new facilities for the Cumulus Dallas group, the merger between Cumulus and Citadel was announced. The project had to be stopped until the newly-acquired Citadel stations could be incorporated into the building. It was a bumpy ride for a while, but the story has a happy ending.

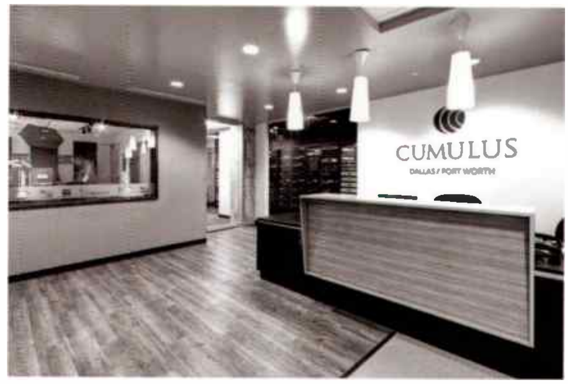

Cumulus' new home in Dallas, at Victory Park on AT&T Plaza

The original relocation project was to bring Cumulus stations from two buildings together. The Cumulus group now includes: KLIF News/Information 570; KLIF-FM, I93; KPLX, 99.5 The Wolf; KSCS, New Country 96.3; KTCK 1310, The Ticket; and WBAP, News/Talk 820 AM & 96.7 FM.

The project was challenging not only because of its size, but because of the diverse organizations that were brought together under one roof. The mix of stations and formats ran the full gamut of operational requirements for facilities. On the human resources side, the cultures that came together in the move ranged from the hip-hop generation to sports talkers to serious news journalists.

The selected location for the new facilities are the Plaza Towers in Victory Park, a 75-acre master planned development northwest of downtown Dallas.

Cumulus occupies the fourth and fifth floors, with facilities separated by product type. The fourth floor houses music and sports-talk stations, as well as business and sales offices. The fifth floor is home to news and information talkers and the news, online and promotions offices.



Lobby includes an equipment rack showcase with plenty of Telos and Axia equipment on display.

Close to Plaza Towers is the American Airlines Center (AAC), home of the Dallas Mavericks and Dallas Stars, and also a venue for concerts and other events. "Our showcase studios are in the major pedestrian traffic pattern for AAC, so we're very visible to concert-goers and sports fans. The plan is to capitalize on that visibility," notes Kline.

Plans for the move actually began two years ago, when the Cumulus stations were on two floors of an older downtown building. "The original buildout was during the analog days, there were lots of multi-pair cables, punch blocks and custom analog switching gear. When we thought about rebuilding the facilities, it quickly became apparent that it would be easier to start from scratch," recalls Kline.


Talent at the Axia Element console in I93's air studio. Note the Telos Call Controller built into the console, which directly controls the Telos VX Broadcast VoIP phone system.

The relocation brought the staffs of both groups into the 21st century. Cumulus Dallas had an interesting mix of analog and digital consoles, while Citadel was mostly analog with older Pacific Recorders consoles. The move transitioned everybody to AoIP with Axia equipment.

One of the unique features of the Dallas build is a talk studio on wheels. A table with mics headphones, tally and a self-contained snake can be rolled out into the mall so that pedestrians can get an up-front look at the action. Ins and outs from the table are terminated in Axia nodes, and Cat 6 cable connects to studios upstairs.

Large talk studio for KTCK ("The Ticket") hosts 5 mic positions with room for more.

The talk operations make extensive use of VoIP using the Telos VX Talkshow Systems. Telos engineer Joe Talbot installed VX engines in all studios and did the initial checkout. Cumulus also uses five Zephyrs, which are shared between all stations. The Livewire outputs on the Zephyrs are another example of the tight integration between Telos Alliance product lines.

Cumulus engineers did most of the Pathfinder programing themselves, but Telos engineers Bryan Jones and Jeff McGinley stopped by for a week at the end of the build for some of the custom programming. As Jones recalls, one Pathfinder project was particularly challenging. "The Ticket does lots of remotes, some of them are quite involved. The producer needs to talk to the announcers in the studio, the editor and people at the remote site, but not all at once. Different situations involve different people." Jones created a Pathfinder program that allows the producer to dynamically group who is on the communications channel.

Edit bays show off Axia RAQ consoles, Telos VSet12 phones, Axia IP-Intercom desktop stations. All connect to the same Livewire network

In a facility the size of Cumulus Dallas, interpersonal communications becomes a challenge. Axia IP Intercom modules were installed in all of the consoles, and desktop stations were placed in several offices. With its broadcast-quality audio specs, IP Intercom provides a seamless bridge between program audio and communications channels. Any intercom station can feed a program channel with the touch of a button. Cumulus also purchased Axia's Soft Com, which enables any PC with speakers and a mic to act as an intercom station.

With the facility is in daily use, Kline reflects that "The beauty of AoIP with Axia gear is that the infrastructure requirements are much less, no punch blocks at all. Everything runs through Cat 6 cables that are terminated in the rack room. That means a vastly simpler infrastructure and faster construction time, which equals less money."

Operator at an Axia Radius 8-fader console looks into the WB AP talk studio. Telos VSet12 phone stands ready to the left

Tom Vernon
Reporter-at-large
The Telos Alliance

192.168.2.199
Fault: None
Config: <none>
Preset: CGSmooth
Input: Livewire 1
Pri
Sec
Input
AGC
ST
Wide Band
AGC
Limiters
FM
B
M
HD
S
L
Analysis
Preset
Wizard
Basic
Advanced
Lim Thresh
0.0 dB
SL to LO Sync
-12.0 dB
Attack Offset
8.3
Release Offset
5.6
LO
ML
MH
HI
7.5 kHz

## OMNIA

The past year has certainly been a busy one for Team Omnia.

Omnia is now an extensive team of individuals who are dedicated to the belief that innovative technology in the world of traditional and digital audio can truly benefit the processing experience for both listeners and the engineering community.

Our team has been busy in the past year coming up with new ideas designed to take the entire Omnia line to the next level and beyond.

Over the past couple of years, we've been working hard to bring a long-time "back of the table napkin" idea to fruition: Single Sideband Suppressed Carrier technology, currently available in the Omnia.11, Omnia.9, and the new Omnia ONE.SG. In many cases, Single Sideband Suppressed Carrier deployment can reduce multipath in challenging terrain conditions and in large cities featuring tall buildings and compactly built neighborhoods.

This year, we have made major steps toward another concept that has been bubbling under the surface in our labs for many years: the digital composite audio link. Several members of our team have been developing and testing this exciting technology which will enable the use of composite audio over digital links to (numerically) feed to digital FM exciters for increased performance.

Another unit of Team Omnia is actively developing a new microphone processor that promises to bring the flexibility of our broadcast processing to human voice reproduction.

As always, every area of Omnia's product line will see updates to enhance old ideas and to innovate with brand new ones.

Be assured that when you purchase an Omnia product, your device can always be updated with the latest improvements, so your investment will pay dividends for many years to come.

It is in our DNA to always try to push the art of audio processing forward. We wouldn't have it any other way.

We appreciate your consideration of Omnia products at your facility and we will continue to work hard to earn your trust.

My best,

Cornelius Gould
Omnia Audio Brand Manager
Omnia Audio

### OMNIA CLIENTS WIN BIG AT THE 2012 NAB MARCONI AWARDS

Congratulations to Omnia clients **KSTP-FM St. Paul** (Legendary Station Of The Year); **WTOP Washington** (Major Market Station Of The Year and News Talk Station Of The Year), **WYCD Detroit** (Country Music Station Of The Year), **KMVK Dallas** (Spanish Station Of The Year) and **WVAQ Morgantown, WV** (Small Market Station Of The Year).

# OMNIA.11

## GAME CHANGER


Before you read any further, it is best to understand that the Omnia.11 is not for everyone.

Probably not even for most.

Omnia.11 is strictly for mission critical processing, where maximum firepower is required in an extremely competitive environment.

Omnia.11 is effortlessly loud. Thunderous bottom end, sparkling highs, and crisp, clear voice reproduction. All with that trademark punch and clarity which makes Omnia the required audio processor of the highest-rated radio stations in the world.

A warning: Many adopters are genuinely startled by the lack of a traditional "processor sound" when the unit is first deployed. The low level distortion and artifacts, long accepted as part of the fundamentals of processing, are now almost completely gone and certainly not perceptible to the ear.

The firmware in Omnia.11 takes advantage of software capabilities never before possible. The results are dynamics algorithms that were once only a dream of the processing enthusiast. AGCs, Compressors, and Limiters analyze music in real time and adjust internal parameters for optimum performance across a broad range of material.

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.

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,

ANDREW JEFFRIES, VP OF PROGRAMMING, CLEAR CHANNEL MEDIA + ENTERTAINMENT, LOS ANGELES, CALIFORNIA

"I'm personally a huge fan of the Omnia.11. After comparing its sound quality to many other options, I found the Omnia.11 to have no equal. This new level of processing will deliver the sound your format requires, and the follow up technical expertise in audio processing from the team is second to none. I've ordered multiple units in the past 2 years"

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.

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.

Another advantage is selective SSBSC (Single Sideband Suppressed Carrier) technology, standard on Omnia.11. This is a method to potentially improve conventional FM-Stereo transmission performance, possibly reduce multipath, and provide increased protection to the baseband spectrum.

Read more at:
omniaaudio.com/downloads/white-papers/MPX-SSB-White-Paper.pdf

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 come standard. Headphone soft "patch points" are available for listening through the processing chain.

Fanless cooling design built into a rugged 4 RU chassis.

## CHOOSE YOUR OMNIA.11

**» OMNIA.11 FMHD**
Featuring independently controlled analog and HD/digital processing paths.

**» OMNIA.11 FM**
For those who have no need for HD/digital processing.
More information at: **OMNIAAUDIO.COM/11**

# OMNIA.9

## MORE STUFF. LESS SPACE.


Omnia.9 is based on a completely new platform which is unlike anything ever offered within the Omnia family. Not only is the platform completely different, but the feature list is far more extensive than that of any other processor available today.

There is literally NOTHING else on the market which comes close to its versatility and multiple functions.

Omnia.9 will simultaneously process FM analog and HD-1, plus encode and process streaming audio. The the 3HD version adds processing for HD-2 and HD-3, plus encoding and processing for the corresponding audio.

Omnia.9 features exclusive "Undo" technology, a source de-clipping algorithm and program-adaptive multiband expander which removes distortion from source material. This corrects over-processed CDs, so common in today's contemporary music.

The psychoacoustically-controlled distortion-masking clipper allows up to 140% within 100% total modulation and an extra 3dB of treble headroom.

Selectable patch points are available for convenient auditioning of the audio signal at any point of the processing chain without affecting listeners.

A built-in RDS encoder is standard and dynamically updateable, with HTTP push support for automation for dynamic RDS, streaming song titles and preset recall.

A separate low-latency studio processing core is provided for talent monitoring.

Selectable SSBSC (Single Sideband Suppressed Carrier) technology is standard on Omnia.9. to potentially improve conventional FM-Stereo transmission performance by reducing multipath distortion and provide increased protection to the baseband spectrum.

Read more at:
omniaaudio.com/downloads/white-papers/MPX-SSB-White-Paper.pdf

MORTEN BACH, PROGRAM DIRECTOR, RADIO ABC, DENMARK

"With Omnia.9, the songs sound even better than the original files. On top of that, the Omnia.9 not only replaces three other units (RDS, Streaming & Emergency audio) but also does their jobs better!"

## FEATURES AT A GLANCE

» Each processing core is separately fully adjustable and has selectable 3, 4, 5, 6 or 7 bands » 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 » On board streaming supports encoding to MP3 (Mpeg-1 Layer 3), MP2 (Mpeg-1 Layer 2), AAC, HE-AAC (including RTSP/3G for streaming to mobile phones), Ogg Vorbis, WMA and WMA Pro » 7 inch front panel touch screen with on-screen keyboard » Full remote control » Psychoacoustic-controlled distortion-masking clipper, enabling a full 3dB of added high-frequency headroom compared to previous processors, for both high fidelity and competitively loud FM audio even at 75us pre-emphasis » HTTP push support for automation, such as dynamic RDS and streaming song titles and preset recall » Dayparting (scheduled preset selection) » Composite pass-through (relay bypass) for your backup processor. **KEY NEW FEATURES:** » Dry voice detector improves quality of voice-only content. » L-R spectral limiter provides additional means of reducing multi-path distortion » MPX Power Mode for ITU BS-412

## CHOOSE YOUR OMNIA.9

**OMNIA.9 FM+HD+STREAM PROCESSING AND ENCODING**
FM plus HD-1 and single stream processing with multiple encoding.

**OMNIA.9 FM+3HD+STREAM PROCESSING AND ENCODING**
FM plus HD-1, HD-2 and HD-3 plus three separate streams processed with multiple encoding.

AL SERGI, GENERAL MANAGER OF SUMMIT MEDIA'S WAFD, WEST VIRGINIA'S 50,000 WATT HIT STATION

"We replaced another company's nearly top-of-the-line unit with the Omnia ONE FM, and you wouldn't believe the audio quality and kick that we have now. The five bands really perform. I don't know how you do it in a unit that is so attractively priced."

# OMNIA ONE

## ONE BIG PUNCH. ONE SMALL PRICE.

"Omnia ONE features the same processing topology that made the original Omnia processors famous, but at a price which is remarkably affordable. Available for FM, AM, Multicast and Studio Pro applications, or as a stereo generator. Application can be changed with simple software download, so your investment is ready for present and future utilization. Introduced in 2007, there are currently over 8,500 Omnia ONEs in service throughout the world."

## ONE FM

» Wide-band AGC for smooth, "hand on the pot" gain riding. » Four-Band AGC to add dynamic EQ enhancement for consistency and to build density before the limiter stages. » Five-Band peak limiter using feedback limiters for the lower two bands (optimized for bass punch and lower mid-range warmth) and feed-forward limiters for the upper two bands. (optimized for sparkling upper mids and highs) » Time-aligned, dynamically flat crossover. » Selectable phase rotator. » Advanced, fully distortion-controlled pre-emphasized final limiter / clipper. » Integrated digital stereo generator with advanced peak control, two composite MPX outputs, SCA input and 19kHz output. » Full remote control via RJ-45 Ethernet port using built-in web interface.

## ONE AM

» Livewire connectivity. » Web Browser remote interface. » Wide-band AGC provides smooth, "hand on the pot" gain riding. » Selectable phase rotator. » Time-aligned, dynamically flat crossover. Four-Band AGC to add dynamic EQ enhancement for consistency and to build density before the limiter stages. » Four-Band peak limiter using feedback limiters for the lower two bands (optimized for bass punch and lower mid-range warmth) and feed-forward limiters for the upper two bands. (optimized for sparkling upper mids and highs) » Advanced, fully distortion-controlled pre-emphasized final limiter / clipper. » Selectable Low Pass Filters suitable for NRSC, HD AM, or any ITU installation. » Full remote control via RJ-45 Ethernet port using built-in web interface.

## ONE MULTICAST

» Exclusive SENSUS™ technology to minimize artifacts as well as restore the fullness and depth that bit-reduction steals. Our DSP gurus teamed up with the codec experts at Telos, the folks who introduced broadcasters to MP3 and MPEG AAC, and together, they developed SENSUS, a unique way of processing your audio to pre-condition it for HD Radio multicasting, specifically tailoring it for the bitrate that you will be using. » Wideband gain rider followed by Four Bands of AGC, Four Bands of limiting, SENSUS and Omnia's proven low-distortion look-ahead Final Limiter. » Time-aligned, dynamically flat crossovers, selectable phase rotator, analog, AES3 and Livewire I/O, automatic input failover on loss of audio. » Full remote control via RJ-45 Ethernet port using built-in web interface.

## ONE STUDIO PRO

» Engineered for full-bandwidth studio applications that require minimal delay throughput and maximum audio quality. » Four Bands of AGC » Four Bands of limiting » New "Bypass" settings for the final look-ahead limiter and Bass EQ sections » Time-aligned, dynamically flat crossovers. » Selectable phase rotator. » Automatic input failover on loss of audio. » Analog, AES/EBU and Livewire inputs and outputs. » External AES/EBU Sync input to synchronize output sample rate to external reference. » Full remote control via RJ-45 Ethernet port using built-in web interface.

## ONE SG

» High-performance standalone stereo generator converts pre-processed analog, AES/EBU or Livewire input to composite baseband MPX output. » Built-in composite clipper to remove minor overshoot. » Single Sideband capability. » Selectable pre-emphasis: 75us or 50us plus OFF. » Selectable de-emphasis on audio outputs for monitoring. » Selectable low-pass filter. » Two composite MPX outputs with individual level adjustment. » Pilot level and pilot phase adjustments. » SCA input with level adjustment. » 19kHz output. » Save and load SG and I/O configuration presets. » Full remote control via RJ-45 Ethernet port using built-in web interface.

If you need to sync the 19kHz pilot frequency to a GPS 10Mhz reference signal, please contact Omnia Support for details.

JOEL LOSEGO, GENERAL MANAGER,
AVC COMMUNICATIONS GROUP IN CAMBRIDGE, OHIO

"The difference from what we had before is amazing, especially in the bottom end and overall loudness. We simply went with some standard presets and were on the air with major improvements on two stations immediately."

# OMNIA.11:
# THE CHOICE OF GREATER MEDIA, DETROIT
## A CONVERSATION WITH MIKE KERNEN, CHIEF ENGINEER.

### CAN YOU SUM UP YOUR IMPRESSIONS OF THE OMNIA.11?

The Omnia.11 represents a sea of change in the way audio is processed for radio. Radio listeners in general now have access to much higher quality receivers and speakers. Audio systems in vehicles are multitudes better than they were only ten years ago. We have to keep pace or risk losing the dashboard to alternative entertainment sources.

### DESCRIBE THE SPECIFIC PROCESSING NEEDS THAT YOU HAVE ON YOUR THREE FORMATS IN THE DETROIT GREATER MEDIA CLUSTER.

WCSX is a classic rock station: Classic rock demands a clean and well organized processing approach. The material varies widely, but much of it has been re-mastered and is still very dynamic. We wanted to keep the dynamics without losing the tonal balance of the source.

WRIF is "Everything That Rocks": Everything from Metallica and Led Zeppelin to Disturbed and Rage Against the Machine. We even have a talk-show in the mornings. To put it mildly, processing a legacy station like "The Riff" is a helluva challenge! The .11 gives me all the tools that I need and more with the ability to remote control it from my garage so I can listen on the killer Boston Acoustics sound system in my ride.

WMGC (Today's 105.1 as we call it) is an AC station with lots of variety. The Four Seasons followed by Katy Perry. Everything we push down the STL comes out sounding lively and bright without being harsh or too demanding. It also doesn't fade away when played in an office environment.

### WHAT WAS THE FIRST THING THAT YOU NOTICED WHEN THE OMNIA.11 WAS DEPLOYED?

Easy. We no longer sunk into the dial. We had the presence to compete without fatigue that plagues other boxes when you make them play loud.

### HOW IS IT AN ADVANCEMENT OVER YOUR PREVIOUS PROCESSOR(S)?

I liked what we had on the air. I'd worked very hard on it, spending hundreds of hours getting it to sound just right. I was in my comfort zone. Eventually, however, we needed to upgrade. We just weren't loud enough anymore and what we had wasn't up to the task. We didn't want to trade away our clean open sound for loudness and when the Omnia.11 gave us a option with no compromise, we took it.

### HAVE YOU NOTICED IMPROVEMENTS ON VOICE AS WELL AS MUSIC?

Music and voice are two very different challenges. Much of the time music is what you aim for and hope the voice works too. The Omnia.11 detects voice-only segments and applies different DSP to it, keeping the clippers from sounding harsh. It's not a new concept but the .11 does it better than I've experienced before in any other processor.

Frank (Foti) and Cornelius (Gould) of Team Omnia are both pros but are open to input from outsiders, too. Their roots as broadcast engineers make them relatable and allows them to apply a sharp focus to their products. These are also people that don't mind having a chat with the user or even dialing in a preset for a specific need.

# STREAMING SUMMARY

## OMNIA.9, OMNIA ONE MULTICAST AND PROSTREAM

### OMNIA.9

As well as processing for analog FM and HD/DAB, Omnia.9 can also encode your internet streams (HD-1 is standard, HD-2 and HD-3 optional). It has completely separate processing cores for analog and digital, and all popular codecs are built in. Setting up internet streams can be troublesome. Omnia.9 puts the good, stable encoders right in the audio processor. It makes sense since the audio is there already, it's on the network already, and Omnia.9 has plenty of CPU left over to do it. Not only does it make setup a breeze, it also ensures the absolute highest audio quality, because there is no chance of glitches or audio degradation beyond what the chosen codec itself does. Omnia.9 also has digital outputs to feed your existing HD encoders, however it does not do the on-air HD/DAB encoding.

### OMNIA ONE MULTICAST

Omnia ONE Multicast is a hardware solution, specially engineered for the challenges of coded audio distribution.

Even very low bitrates can be effectively managed by Omnia Audio's exclusive SENSUS™ technology, a standard feature in the Omnia ONE Multicast. SENSUS minimizes artifacts as well as restores the fullness and depth of the audio transmission unlike no other method. SENSUS enables the audio processor to modify its own architecture in real time and in response to ever-changing program content. Simply stated, 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. Features include Wideband gain rider followed by Four Bands of AGC, Four Bands of limiting and Omnia's proven low-distortion look-ahead Final Limiter. Time-aligned, dynamically flat crossovers, selectable phase rotator, analog, AES3 and Livewire I/O, automatic input failover on loss of audio. Full remote control via RJ-45 Ethernet port using built-in web interface.

### PROSTREAM

With ProSTREAM, the professional Internet streaming solution from Telos, you don't need separate devices for audio processing, encoding and streaming; all of these functions are included in a single 1RU device. To make your streaming audio sound its best, ProSTREAM first employs audio processing from Omnia to ensure the smoothest, cleanest possible audio for your Internet stream. Processing functions include wideband AGC, 3-band combined compressor/limiter, high-frequency equalization, an adjustable-bandwidth low-pass filter, and a final Look-Ahead limiter. There's also exclusive SENSUS technology that analyzes audio content and adapts processing algorithms to optimize it for your selected bit-rate. After processing, ProSTREAM uses genuine MPEG encoding algorithms from FhG, the inventors of MP3, to ensure the most artifact-free sound quality at whatever bit rate you choose. You can encode directly to MP3 or MPEG-AAC, and feed any Shoutcast-compatible media server, or a Wowza server for streaming to Flash clients. ProSTREAM comes standard with studio-grade analog I/O, and also works with Livewire IP-Audio systems, taking audio directly from your network.

# OMNIA A/XE

## THE COMPLETE SOLUTION FOR AUDIO STREAMING

**Hardware Requirements:**

» 32-bit Windows XP and later » Minimum 512MB RAM
» 20MB free hard-drive space » Network Interface Card


Attain total control of your audio streams with Omnia A/XE, featuring genuine Omnia processing for your audio workstation.

Sound which is pure, clean and compelling.

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.

Encode directly to MP3 or AAC, feed a Shoutcast-style or Windows Media Server in the MP3 format, or stream to Adobe Flash clients through a Wowza Media Server. You can also pair Omnia A/XE with your existing Windows Media, Real, mpgPRO or MP3 streaming encoder.

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.

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.

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.

**ANDY REBSCHER, CHIEF ENGINEER, WOZI, PRESQUE ISLE, MAINE**

"Even at low bitrates, the Omnia A/XE sounds great. We have it at many of our Townsquare Media stations"

# OMNIA F/XE

## SPECIALLY ENGINEERED FOR PODCAST AND FILE ORIGINATION

**Hardware Requirements:**

» Windows XP and later » 20MB free hard-drive space
» Microsoft.NET client framework 4.0 » Internet access


Omnia F/XE is a file-based audio processor and encoder application. It combines Omnia audio processing with the Fraunhofer MP3 and AAC codecs for high quality file prep for podcasting or filebased streaming. For live processing and encoding, see Telos ProStream (page 34) and Omnia A/XE (page 46). Omnia F/XE is software only, no special cards are required. Able to read PCM, WAV, MPEG Layer-2 and MPEG Layer-3 source files.

**PLUS:** » Featuring genuine Omnia processing to improve audio levels, loudness and perceived quality. » Can automatically send the output file to an FTP server. » Will notify the user by email if problems are detected. » Logs are kept during processing so you can find the source of a problem. » 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. **CODECS:** » MP3, AAC, HE-AAC, HE-AAC v2. The highest quality codecs from Fraunhofer.

# OMNIA.8x

## EIGHT DISCREET OMNIA PROCESSORS IN A SINGLE NETWORKED UNIT


Wherever multiple instances of coded audio processing is required, you will find the Omnia.8x.

Omnia.8x dramatically improves the sound of your streams with the power, punch and purity of Omnia audio processing.

But Omnia.8x doesn't process just one audio stream. Omnia.8x combines eight separate 3-band stereo Omnia processors in a single networked appliance for on-demand processing of multiple HD Radio channels, sat-casting, Internet audio, studio production, podcasting, headphone feeds, etc. Multiple audio processors for full-bandwidth studio applications that require minimal-delay throughput and maximum audio quality.

We started with algorithms modeled after those used in our popular Omnia platform, then refined them even further. Omnia.8x's unique processing architecture is designed to work ahead of any bit reduced audio coder to reduce artifacts and improve overall audio integrity. Omnia.8x conditions and enhances to make sure that your coded audio sounds as good to your listeners as it does in your studio.

Connects to your Axia Network using a single CAT-6 cable for all I/O; pair with an Axia Audio Node for use as a standalone multiple-stream audio processor. Second-generation DSP processing platform is fan-free for cool, silent in-studio deployment and is equipped with dual Gigabit Ethernet ports, dual-redundant internal power supplies with automatic switching for complete peace of mind. Auto-sensing power supplies, 90VAC to 240VAC, 50 Hz to 60 Hz. 100 Watts. Rackmount, 2RU. *(Not designed for final AM or FM transmitter processing).*

**ROB CHICKERING, TECHNICAL DIRECTOR, KIDD KRADDICK IN THE MORNING**

"We purchased the Omnia 8x for Kidd Kraddick's syndicated radio program, which is heard in more than 75 markets and the American Forces Radio Network. The 8x gives us eight processors in a 2 ru chassis and its got Livewire. It is easily inserted into network feeds and headphone chains. Web interface provides easy access to processing presets. It keeps us out of the 'red' and our affiliates love the clean audio and even levels."

# OMNIA.9:
# THE SUMMIT IN AKRON USES IT FOR EVERYTHING!
## IMPRESSIONS FROM BILL GRUBER, PROGRAM DIRECTOR

Bill Gruber, Program Director of one of America's most respected Adult Alternative stations, WAPS (The Summit) 91.3 in Akron, Ohio, recently made the commitment to process the entire WAPS transmission lineup (analog FM, HD-1, HD-2 and HD-3) with a single, versatile Omnia.9.

"It may be a cliché, or even the holy grail of radio audio processing, but the Omnia.9 really sounds very close to 'CD-quality' while maintaining competitive loudness.", he says. "The Omnia.9 is the closest to 'HD Radio' sound quality I've ever heard on conventional analog FM broadcasts"

An Adult Alternative Station like WAPS has to deal with a wide variety of different sounds. In a single hour, it is not uncommon to hear Melissa Etheridge, The Clash, Bob Marley, Fun, Norah Jones and Green Day. Rock, Pop, Folk, Acoustic, Local, Reggae... many musical genres. At other broadcast times, there are syndicated shows like "World Café", "Acoustic Café" and "The Chill Side Of The Summit" featuring softer, ambient music.

But this is no problem for the Omnia.9.

"We have experienced amazingly smooth song-to-song musical transitions. We all know the wildly varying audio levels and quality of production from one song to another, yet the Omnia.9 gracefully levels the audio playing field," says Gruber.

Sometimes audio processors which promote "clarity" tend to over emphasize high end. This is not the case with the Omnia.9. "It has the best high frequencies I've ever heard on FM, and I mean honest treble, not artificial sizzle. The Omnia.9 keeps the music clean and bright without adding annoying rasp or sibilance to announcer voices."

The Summit's HD-2 ("Summit Flashback 80s") and HD-3 ("Kid Jam") are also independently processed by the very same Omnia.9. "Our other HD services sound terrific, too. This one unit does everything. It is truly amazing"

How about Time Spent Listening?

"We noticed improvements in Time Spent Listening, due to the fact that we now get competitive loudness with zero fatigue."

And listener comments?

"In addition to all the sonic advantages the Omnia.9 provides under normal signal conditions, it really maintains your signal quality way out in the fringes of your coverage, including greatly-reduced annoying mono-blending on car stereos. We received several listener comments asking if we raised our power."

Bill has this concluding statement:
"I can highly recommend the Omnia.9, especially in formats where many diverse program elements need special attention".

# OMNIA DIRECT

## MPX DIGITAL CONNECTIVITY



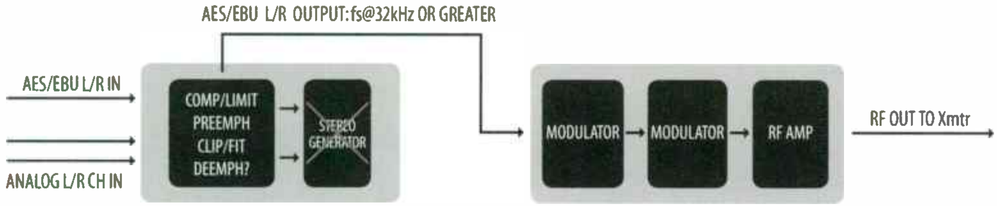
Prior to the advent of digital exciters for FM radio, integration of an audio processor (with integrated stereo generator) was done via a baseband or multiplexed (MPX) signal connected directly to the exciter's broadband (DC – 53kHz) input. The MPX signal would be routed directly to the modulator. This tried and true method is a present day standard. Any audio processor, be it analog or digital, that contains its own stereo generator offers an analog based MPX signal on a BNC connector. The connection of this signal, should it be directly out of the stereo generator itself or from the output of a composite STL system, is then connected to the MPX input on the exciter via another BNC connector. Anyone who has installed audio processing will understand this.


ANALOG PROCESSOR WITH MPX OUTPUT

ANALOG FM EXCITER

With the introduction of digital exciters, the only MPX input is analog and it must be passed through an analog-to-digital converter (ADC) before it can be digitally modulated. The sampling rate of the ADC must be relatively high due to the wide bandwidth of the MPX. Consider for a moment the audio spectrum of FM is 99kHz wide. Therefore, any analog signal connected to a digital exciter must be sampled at 200kHz or higher. Once digitized, no other processing is required before modulation.

To date, if an all digital transmission path is desired, the only interconnect between the audio processor and exciter requires using the AES/EBU discrete left/right input. This path is quite a bit more complicated than the MPX path because it must condition the audio signal in preparation for stereo generation and herein lies potential for audio degradation. In this configuration, the FM-Stereo signal must be generated in the exciter.

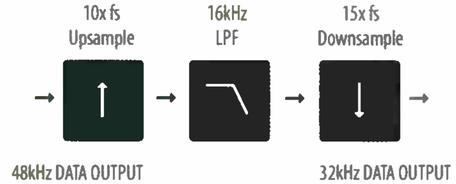



DIGITAL PROCESSOR WITH AES/EBU OUTPUT STEREO GEN NOT USED

DIGITAL FM EXCITER AES/EBU INPUT USED WITH INTERNAL STEREO GEN

Experience has proven this method less than ample in modulation efficiency. This is primarily due to the required sample rate converters (SRC) employed in the exciter's input section. The signal that is arriving at the AES/EBU input of the exciter might be operating at a different sampling rate than the exciter is expecting. If so, a rate converter is employed to make the proper transition. This device can pose problems as the digital filter in the rate converter can generate overshoots to the already tight peak controlled audio signal that is being adjusted.

The following example illustrates how a synchronous SRC works. However, interfaces between equipment are not synchronous and therefore require asynchronous SRCs. These are functionally similar but considerably more complex. In order to synchronously change 48kHz to 32kHz sampling, the conversion is accomplished by scaling up, or interpolating the original sampling rate, usually by a factor of ten. Then, at the 10x rate of 480kHz, the signal is low pass filtered at the Nyquist of the new desired sampling rate. This filter is required to 'smooth' out the 10x rate. If filtering not used, aliasing products would result. Finally, the signal is scaled down, or decimated by the factor needed, in this case ÷15 to achieve the new rate of 32kHz. Below, is a block diagram of an SRC. While this sounds quite simple, and basically it is, there are a few issues to consider. Of main interest is the interpolation filter.



All audio processors, both analog and digital apply some form of overshoot control to the output filtering section. In most designs, this function is a form of integrated protection clipper working around the final low pass filter to obtain control.

In each case the overshoot component can be calculated as a product of what is known as the 'Gibbs Phenomenon', which states that an overshoot will occur at one-third the cut-off frequency of any low pass filter whenever a non-linear waveform is passed through it. In the case of broadcasting, the non-linear waveform would be that of a clipped waveform.

Knowing that the audio bandwidth used in FM-Stereo is 15kHz, overshoot components will begin with any non-linear waveform above 5kHz. In this example, this would affect any signal above 5kHz that was clipped.

Should the slope of the previously described up-sampled interpolation filter appear greater than the slope of the final filter in the audio processor, then output overshoots may result in the sample rate conversion process! Unfortunately, these overshoots are generated after the processing unit. To remove them would require another limiting device. Some exciters provide an additional peak limiter, and the action of this limiter can degrade audio quality.

## ENTER OMNIA DIRECT

The desired goal for the digital interconnect between audio processor and exciter, has been to emulate the older analog method, where the audio processor generates the FM-Stereo MPX signal, and then connect to a wide spectral input on the exciter...a digital BNC-to-BNC connection so to speak.

Omnia Direct generates the FM-Stereo MPX signal inside the audio processor, and scales the sampling rate to 192kHz. Through recent firmware developments, the AES/EBU transom can accommodate a sampling rate at 192kHz, making it possible to transport the MPX signal directly from the audio processor to the input of the exciter...MPX over AES. Now it is possible to create a digital version of the older analog method. Below is an illustration of this concept.


There are benefits to this configuration. Now it is possible to couple the output of the audio processor directly to the input of the exciter's modulator in a full linear fashion. There are no SRC functions in the path, which can degrade modulation and sonic performance due to overshoots. The absolute peak level at the output of the stereo generator, will translate to the absolute maximum deviation of the FM carrier. Accomplishing this insures the modulator will faithfully mirror the performance of the audio processor. This reduces, and basically eliminates any coloration of the audio due to the exciter's input and modulator sections.

Since the connection between the audio processor and modulator is as close to theoretically perfect as possible, there will be an improvement in sonic bass performance heard on the radio. This has always been a weakness within the analog MPX connection to an exciter, as any unwanted signal components near DC, would negatively affect the performance of the AFC (automatic frequency control) section of the exciter. This usually required the use of a high-pass filter, or DC-servo of some sort. Both of these can degrade the low frequency performance of the modulator, which reduces or smears the aural texture of on-air bass.

Additionally, any specific functions unique to the audio processor/stereo generator are maintained, as they will not be forced to rely on any limitations imposed by the stereo generator in the exciter. By example, the alternative use of SSBSC (single sideband suppressed carrier) in the stereo generator, which has shown to reduce the effect of multipath, can be employed in the audio processor, and this alternative MPX signal will easily connect and operate correctly. Also, the AES/EBU signal is a standardized method. This now enables a digital MPX connection to be utilized as an industry standard.

Another benefit to this method is it now enables digital distribution of the MPX signal over wide broadcast networks. There are many such infrastructures of this type internationally. Any digital STL system capable of handling 192 kHz sampling should be able to carry this signal.

To date, Omnia Direct is available on the Omnia.11 and the Nautel NV series of FM transmitters.


25
seven

## 25-Seven Systems

I met fellow cofounders Barry Blesser and Derek Pilkington back in 1990, and the rest of the 25-Seven team shortly thereafter. The list of companies for which our collective team members have worked reads like a broadcast and audio company phone-book. If products were like record albums, you'd see our players are in a lot of the liner notes of both platinum hits and critically acclaimed gems.

While I get credit for starting 25-Seven Systems, it was self evident to me back in 2003 that the company already existed, it just didn't have a name, a logo, any capital or a project exciting enough to get "the band back in the studio."

Audio Time Manager was the project that put it all into play.

The idea to marry ultra-complex time compression algorithms to an interface any board op could run got the sparks flying. Derek and I came up with a spec and ran it past Barry. We worked up a plan. We rounded up the usual suspects. We borrowed some money from friends and family. We printed business cards.

Company names, logos and even capital are relatively cheap. What's rare and valuable is a tribe of self-selected individuals, endowing an intense collaboration with the currency of trust and respect for each other's strengths and weaknesses.

No MBA program will teach you this. But it's our story.

Audio Time Manager became radio's pre-eminent solution for real-time time compression and time shifting. It's used at some of the biggest talk stations in the world. I've had users tell me that they don't know how they did live radio before or how they could live without it. And ATM fit right in as one of the first hardware boxes to speak this new IP audio thing called Livewire.

Our second product, Program Delay Manager reinvented the profanity delay, taking an old process to a new level. PD-Alert, our patented system, captures before/after audio snapshots of every dump event and emails them to staff. Stations tell us this feature alone has given them instant visibility and an essential quality control tool in the quest for better air product and higher ratings. (Plus fewer fines!). Busy PD's and GM's love it; while some show hosts and producers hate it. Do you need more proof than that?

We're now focusing on solving other delay and time related problems. Precision Delay addresses time alignment issues between HD and analog broadcast, as well as other fixed delay applications. It also addresses the thorny issue of building in and out of HD alignment delay without trashing watermarking codes. All our delays offer separate means for delaying contact closures and serial data over IP or RS-232. After all, ancillary data really ought to stay synced to audio.

Now 2003 seems like yesterday. In the years our collective team has been working in this business, we have been peers, competitors and collaborators with our friends at The Telos Alliance. Now we're family and the next tour begins. We're already working on some hit records for you with the Alliance prodigies. We hope you'll love the harmonies.

Tipping my hat to the new constitution,

**Geoff Steadman**
President
25-Seven Systems

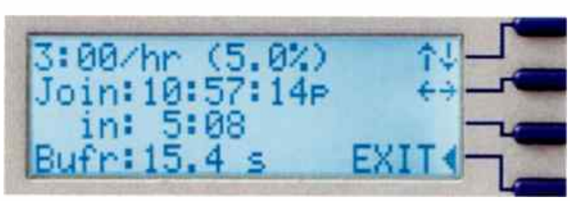
# AUDIO TIME MANAGER

## CREATE TIME... SHIFT TIME... CONTROL TIME.


**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.




Users can easily adjust time compression rates on-the-fly by percentage, or by entering a join time! 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.

**NOW 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.

**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.

BRIAN OLIGER, OPERATIONS MANAGER, WTOP, WASHINGTON DC

"The ATM solves an age-old problem for us: how to smoothly join a live network news update when breaking news makes it impossible to do in real time. Our news anchors hit the network seamlessly, whether it's on time or 20 seconds late."

KIPPER MCGEE, KIPPER MCGEE LLC

"25-Seven is the best tool I've ever seen to help ANY spoken word station meet and BEAT the clock! From the ability to seamlessly drop in call letters during a breaking news event to finding 'extra' time during network feeds of all sorts, 25-Seven can help your station add ratings AND revenues!"

### 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. Our Time/Rate Management Calculator adjusts to what you want to do.

### 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.

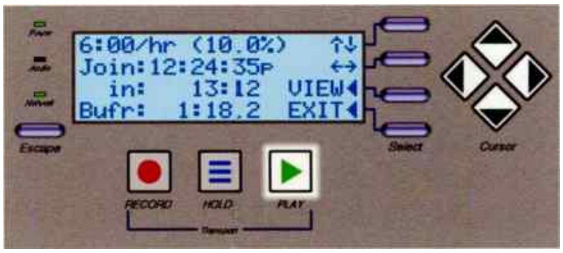
Some technologies can drastically cut into pauses, destroying a personality's careful delivery. Our proprietary algorithms intelligently process speech: pauses stay in proportion; pacing and inflection are maintained. Even stereo music stays clean and steady.

### "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

### 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.




Control ATM across your LAN or WAN, using any web browser.

### 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.

### AXIA LIVEWIRE COMPATIBLE.

ATM operates directly on Livewire AoIP networks.

# PRECISION DELAY

## YOUR STATION... IN SYNC AND ON TIME.


### 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 signal 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.

### KEEPING HD RADIO IN SYNC WITH ANALOG.

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.

### 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 solu-

**GEOFF STEADMAN, PRESIDENT, 25-SEVEN SYSTEMS**

"Your signal is more than your audio programming. Unlike other broadcast delays or systems that use pitch shifting, Precision Delay never alters frequencies. Our advanced algorithms and special Watermark Safe Mode leaves your watermarking data, such as embedded PPM codes, intact during delay builds and exits, preserving data critical to your ratings and revenue."

tion with no spinning hard drives and no complicated programming. With "set and run" simplicity and solid-state reliability.

### DELAY DATA & GPIO.

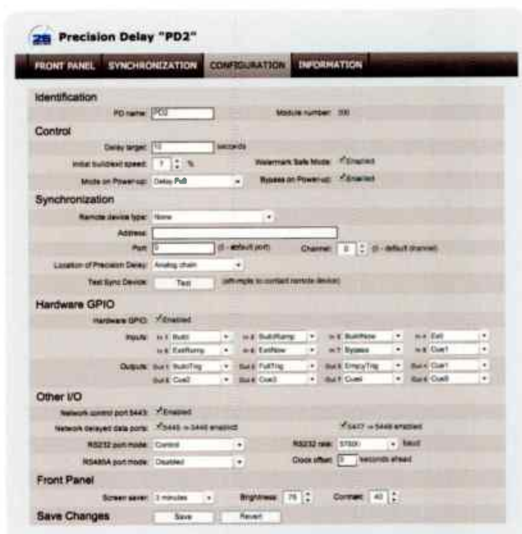
Serial data over IP or RS-232 such as "now playing" metadata can be delayed in sync with audio — even when delay time is in transition. Contact closures can also be delayed on input so that they trigger against the appropriate audio on output.

### 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.

### 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.




A built in web server provides password protected access. Configure and control your delay over a LAN or WAN.

BILL TRAUE, CHIEF ENGINEER, RIVERBEND COMMUNICATIONS, BLACKFOOT ID

"PDM is a beautiful design with features I never knew I could get in a delay unit. It performs as good as it looks, and is backed by some of the smartest guys in the business."

# PROGRAM DELAY MANAGER

## PROGRAM DELAY FOR THE 21ST CENTURY.


### 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 E-MAIL.

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 e-mailed 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.

### 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 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!

### AUDIO, RDS AND DATA STREAMS 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. The program ancillary data can be fed through PDM using its RS-232 or IP ports.

### 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.

PAUL SHULINS, DIRECTOR OF TECHNICAL OPERATIONS, GREATER MEDIA BOSTON

"In this age of tighter FCC regulations for program decency and content, the PDM has provided my station with an extra level of proof that we are on top of that issue, and gives us confidence our producers are trained properly and on top of their game."

**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.

**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.

**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.

**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?


A built in web server provides password protected access. Remote control your PDM, change configuration settings, or manage Overkill "fill" files and dumped-audio archive files easily over a LAN or WAN.

# PROGRAM LENGTH MANAGER

## FOR THE LONG AND SHORT OF RADIO.


### MAKE THE BEST USE OF YOUR TIME.

Radio. Every second counts. Program segments need to start and end exactly on time, with seamless transitions. Editing time is precious in the production room and the field. If your piece is just a bit long or a bit short, will the pressures of getting it on air force compromising edits or inappropriate filler?

Enter the 25-Seven Program Length Manager (PLM) to shrink or stretch your programs and program segments by several minutes per hour without pitch change, artifacts or glitches. Time-manipulated audio is clean enough to use on stereo music programs and live events. PLM is operator friendly and requires little training whether controlled from the front-panel or via web browser.

### YOUR PROGRAMMING—ON TIME, EVERY TIME.

PLM is the next best thing to being able to stop the clock or add more minutes to the hour. Now, network head-ends that rely on prerecorded programming are free to focus on creative radio and let PLM assure continuity. That syndicated or brokered program and your local content don't add up to the allotted slot? PLM can stretch the program to fit without the need for filler material or extra production steps. Need to insert some additional material without program interruption? Easily accomplished with PLM and not perceived by your listeners.

### FRONT AND CENTER OR BEHIND-THE-SCENES.

The front-panel LCD "dashboard" has a time readout and an audio level display that allow quick parameter changes. Automation-friendly remote control options include user assignable GPIO (switches, relay closures, or TTL), secure Web interface, RS-232 or serial control over an IP port. Network cues can be automatically delayed to match program audio and PLM's clock automatically synchronizes to a local or Network Time Server.

### WEB CONFIGURABLE.

Navigating though "set and forget" parameters is a breeze with our internal password-protected Web server. Connect PLM to your local area network and properly configured browsers can log in for an on-screen emulation of PLM's front panel and configuration functions. The network interface also lets you browse for and install software updates.

### FUTURE-PROOF AUDIO QUALITY.

Superior balanced analog I/O and AES digital are standard. Even during time manipulation, PLM boasts 84 dB s/n with 10 dB headroom, frequency response of 20 Hz-20 kHz (+0/-0.2dB) and 0.01% THD+N. Audio is always linear, with no lossy data reduction to mess up your sound.

DR. BARRY BLESSER, CTO, 25-SEVEN SYSTEMS

"As with all our products, PLM uses advanced time manipulation algorithms that don't affect pitch and don't add harmonic distortion or frequency limitations. Some technologies can drastically cut into pauses, destroying a personality's careful delivery. Our proprietary algorithms intelligently process speech, so pauses stay in proportion and both pacing and inflection are maintained. Even stereo music stays clean and steady. No choppiness, no stutters, no fuzziness. A tutorial in the PLM manual teaches you how to select the optimal speed change settings to meet your programming objectives."

**PLM OPERATIONAL MODES.**

**Shrink to fit:**

PLM applies the glitch-free time compression technologies developed for 25-Seven's groundbreaking Audio Time Manager. It offers two ways to shrink a program:

> BUFFER+PLAY provides a quick way of pausing your main program while inserting extra content. Trigger PLM from the front panel or by remote and the main program will be recorded until you command the unit to start playing out. PLM smoothly plays the buffered audio until it rejoins real time. During the playout, you can manually adjust the compression rate and rejoin time.
>
> SHRINK TO TARGET works like the program stretch function, allowing advance computation of program length and compression rate.

**Stretch to fill:**

Our all-new time expansion algorithms inaudibly stretch under-length programs to fit their time slots. Intuitive operator controls make it easy. Just enter the actual program length and target playout length. Then, PLM recommends a stretch rate and shows you the completion time of the stretch function. PLM looks at the entire program, eliminating the need to adjust individual program elements, such as spots or songs. You can then adjust the stretch rate and the point in time that PLM returns to real time to suit your preferences.


0
10

## AXIA

2013 marks a banner year for Axia, for this year we celebrate the first decade of the Connected Studio.

In retrospect, it's hard to believe that it's been 10 years since a start-up division of Telos grabbed everyone's attention with a radical idea about building radio studios with Ethernet. At a time when "state of the art" meant discrete digital audio, and a routing switcher was something only the richest stations could afford, Axia shocked the world by declaring that studio peripherals from the console to the audio processor could be networked, and that everyone could have the benefits of a routing switcher — for about a third of the cost of traditional technology.

When we launched Axia, Telos founder Steve Church made a prediction. "Three things will happen," Steve told us. "First, people will say that 'it will never work.' When they're proven wrong, they'll say 'It works, but you don't need that.' And finally, as they see Axia becoming successful, they'll say 'IP-Audio? We do that too!'"

Steve, as always, was right on the money. In the last 10 years, Axia Livewire has become the world's most popular IP-Audio networking technology, and our mixing consoles are the world's best-selling — they're on-air in over 4,000 studios, and counting. You'll find Axia equipment everywhere: at privately-owned stations, and large clusters run by big conglomerates. At established public broadcasters, and at newly-licensed International stations. At government broadcast facilities, and podcast studios. Needless to say, we're very humbled by the trust broadcasters place in us — and by the enthusiasm you show!

Being first with new technology is nice, but we've never been content to rest on our laurels. So the Telos R&D team (the industry's largest) continues to innovate and expand the scope of IP-Audio. Axia consoles are the first and only IP-Audio consoles with a pre-configured network switch built in, to save broadcasters the effort and expense of procuring and programming third-party switches. Our new compact xNode AoIP interfaces feature one-button setup, and can run on mains power or Power over Ethernet (PoE), for flexibility and redundancy. And we've engineered RAQ and DESQ, our newest mixing consoles, to be both space- and cost-efficient for easy deployment in studios of any size.

More than 40 Livewire partners believe in the vision of the Connected Studio too. Collectively, they make more than 65 hardware and software products (ranging from telephone systems to audio processors to program automation) that interoperate directly with Axia networks via Ethernet – making the connections "smart", and eliminating the need to purchase audio conversion devices. We've also partnered with RAVENNA, the high-performance audio standard backed by more than a dozen broadcast manufacturers, to expand broadcasters' networking options even further. And there are a lot more great new ideas in the cooker — so watch this space!

But in the final analysis, Axia's success isn't due to our efforts. It's you, the broadcast professional, who has embraced our vision, used our technology in ways we never dreamed of, and raved about Axia products to everyone who'd listen. We owe you a huge debt of thanks. And we promise to never stop trying to amaze and delight you.

So here's to the next 10 years. Excelsior!

**Marty Sacks**
Vice President and Executive Director
Axia Audio

# ELEMENT

## THE WORLD'S MOST POPULAR IP-AUDIO CONSOLE

SILVER

BRONZE

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 4,500 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.

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

UNIVERSAL STUDIOS RADIO, ORLANDO, FLORIDA

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 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.

# ELEMENT

## CHOOSE YOUR OPTIONS.



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 power supplies, and 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, EQ and voice processing. Or, choose the PowerStation integrated console engine instead.

Each Element fader module includes an overbridge-mounted information panel. Status Symbols give information at a glance about phone lines, talkback activity and more.

Say goodbye to mix-minus hassles. Element automatically generates backfeeds for all sources that need them. Status Symbols tell you when they're active.

Premium-quality 100mm conductive-plastic faders are silky-smooth, but built to last. All Element modules can be ordered with motorized faders for remote control operation or integration with audio delivery systems.

Film-Legendable Switch Module with fixed-function buttons is available in 5-button or 10-button sizes. Backlight colors are changeable, too. Use PathfinderPC software to assign buttons custom routing functions, audio device controls and more.

Integrated, custom-molded finger guards help ensure error-free operation. Element features LEDs in all lighted buttons.

Telos Call Controller module with built-in faders lets you control advanced Telos phone systems right from the board. Status Symbols icons inform operators of line and caller status with just a glance.

Element intercom modules work with Axia IP Intercom system to provide easy inter-studio communications right from the console. Makes it easy for talent to put broadcast-quality intercom audio on the air, too. Available in 10- and 20-station modules with OLED or film-cap displays.

PowerStation is the integrated console engine that works with Element. PowerStation is an all-in-one titan that makes it easier than ever to install IP-Audio studios. Inside that ruggedly handsome case you'll find a super-powered DSP mixing engine, husky ready-for-anything power supply, plenty of digital, analog and mic I/O, EQ, voice processing — and even a custom, built-for-broadcast Ethernet switch with Gigabit connectivity.

We use superior-quality parts to ensure long, reliable service. Long-life conductive-plastic faders, aviation-quality switches, and rear-screened polycarbonate surfaces that won't chip, crack, peel or lose their markings are just a few of the things clients have found to love about Element consoles.

Need to conduct an interview instead of playing music? Press the Show Profiles key to load a saved console "snapshot" instantly.

Full-featured Monitor/Navigation module features independent source selection for headphone, control room and studio monitors. Speed keys and context-sensitive SoftKnobs work with the on-screen display, enabling board-ops to quickly customize console options. Also available: Monitor + 2-Fader module combines monitor and nav functions with two faders — perfect for news or voice-over studios, dubbing stations, or anywhere space is at a premium.

Push the Record Mode button to instantly reconfigure the board for recording phone bits, interviews, etc.

Numeric keypad lets you dial phones without ever leaving the board.

4-Fader modules accommodate any source from the IP-Audio network — mics, line sources, computer playout systems or anything else. Special SET and HOLD keys provide dedicated on-fader control of Telos broadcast telephone hybrids.

Programmable SmartSwitch modules with LCD displays built into the control buttons are available in 10-button and 5-button sizes. Use PathfinderPC's Panel Designer feature to create conditional multi-salvo router events that launch with a touch. LCD text and backlight color can change dynamically.

High-impact Lexan overlays with color and printing on the back, where it can't rub off. We don't just stick the Lexan to the top of the module like some folks do: our overlays are inlaid on the milled aluminum module faces to keep the edges from cracking and peeling — expensive to make, but worth it. For extra protection, there are custom bezels around faders, switches and buttons to guard those edges, too. Element modules will look great for years, and are available in your choice of Bronze or Silver.

# ELEMENT MODULES

## BUILD YOUR ELEMENT

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!

### 4-FADER

The 4-Fader module is the heart of any Element. Use it for any source: line, mic, hybrid, phone or codec source.

**FADER OPTIONS:**
STANDARD
MOTORIZED

BRONZE
SILVER

### CALL CONTROLLER

The Call Controller module has two faders plus integrated line switching controls with Status Symbols for controlling advanced Telos broadcast phone systems.

**FADER OPTIONS:**
STANDARD
MOTORIZED

BRONZE
SILVER

### STANDARD MONITOR

Space-saving 2-Fader + Monitor module has two faders in addition to numeric entry/dial pad, basic Monitor/Headphone controls and Soft-Knob overbridge control set with alphanumeric display.

**FADER OPTIONS:**
STANDARD
MOTORIZED

BRONZE
SILVER

### EXPERT MONITOR

Monitor/Navigation module has deluxe monitor, headphone and preview controls. Numeric entry/dial-pad can be used with Element phone modules; convenient profanity delay controls can be linked to your external delay unit.

BRONZE
SILVER

## PRODUCTION

The Production module gives your talent direct access to frequently-used production options, such as Send bus levels, EQ settings, and panning tools. Includes overbridge selector panel with alphanumeric source display.

BRONZE
SILVER

JOHN PENOVICH, RADIO FREE ASIA

"Axia's Show Profiles are great for us, because with Elements equipped with motorized faders we can create set-ups that are as simple to use as possible."

## FILM-CAP SWITCH

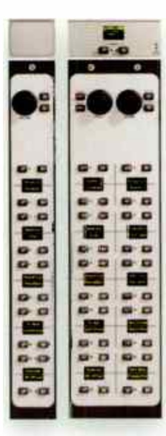
Film-cap switch modules with 5 or 10 buttons give talent access to often-used machine-control or GPIO-triggered routing commands. LED button backlights can be individually changed to any of 8 colors.

BRONZE
SILVER

## SMARTSWITCH

5 and 10-button SmartSwitch modules feature backlit LCD displays; functions, colors and text can change dynamically in response to user input. Use Axia PathfinderPC software to program SmartSwitches with custom salvos, machine-logic commands or other complex routing operations.

BRONZE
SILVER



## 10- AND 20-STATION OLED INTERCOM

10 and 20-station OLED Intercom modules, part of the Axia IP Intercom system, put broadcast intercom controls right in the console. High-resolution OLED displays indicate preset stations; presets and GPIO functions are programmed with a standard Web browser.

SILVER
BRONZE

## 10-STATION FILM-CAP INTERCOM

10-Station Filmcap intercom module has 10 LED-lit film-cap buttons for economical on-console IP Intercom integration. Station presets and GPIO functions are programmed using any standard Web browser.

SILVER
BRONZE

# POWERSTATION

## POWERFUL. EASY. BULLETPROOF

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 best-selling Element 2.0 mixing console. PowerStation is over-engineered to Axia standards, every part chosen for its ability to give constant, uninterrupted service. PowerStation combines four separate devices – a DSP mixing engine, a console CPU and power supply, audio I/O, GPIO and a custom, Axia-designed Ethernet switch – into a self-contained console engine that's engineered to ensure years of reliable, trouble-free service.

There are no compromises: PowerStation uses only best-of-the-best components, like studio-grade mic preamps and 24-bit, 256x oversampling A/D converters, a rigid, EM-tight chassis, an ultra-reliable DSP platform (not a common PC motherboard) and a hardened power supply designed for unfailing service, even in the harshest environments.

PowerStation Main is where you start. Inside is a bulletproof mixing engine capable of powering Element 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!

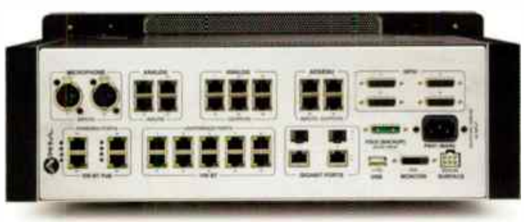
To beef up your PowerStation studio even further, there's PowerStation Aux. Connect it to a PowerStation main to instantly double mic, analog, AES and GPIO ports, and add a redundant backup power supply with auto-switchover. Most redundant supplies protect only the console, but with PowerStation, the 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.

# POWERSTATION

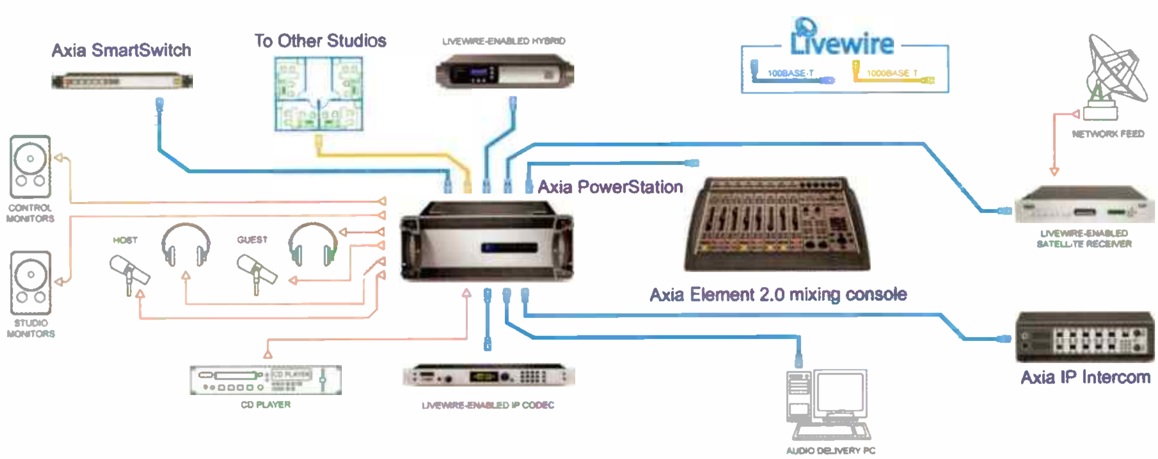
## THE STANDALONE CONSOLE THAT NETWORKS

It's easy to build studios with PowerStation. With 32 built-in I/O ports, DSP mixing engine, console CPU, bulletproof power supply and network switch all in a fanless 4RU enclosure, PowerStation and Element consoles combine to make a networked broadcast console that doesn't need a network. It's completely self-contained.



Of course, PowerStation works flawlessly as part of a large network, but if you unplug its network cable, it's completely unaffected – your programming continues on without interruption.

Not only does PowerStation make it easy to build stand-alone, independent studios, it also makes it easier than ever to network them together. Simple Networking allows up to four PowerStations to daisy-chain without the need for a separate switch, although you can add one or more to expand your Axia network. Because Axia networks are based on standard Ethernet, they're naturally scalable — up to as many as 10,000 stereo streams, all with individual control logic and program-associated data.



STEVE GROSZ, KVIT-FM

"We had researched IP audio consoles. . .and narrowed the field to three vendors, including Axia. As we went around and visited the manufacturers, it became clear that Axia really represented the latest technology and the future, and the other two looked more like '90s state of the art."

# iQ

## THE SMARTER IP CONSOLE

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 stand-alone studio — but that also connects quickly and easily to a larger studio network.

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

RADIO EUROPA 2, BRATISLAVA, SLOVAKIA

the ability to meter two, three, or all four buses at once. There are 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. iQ's OLED fader displays show you the source that's assigned to each channel strip, but they're smarter than that: they 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. Since iQ only requires a single cable to connect control surface to mixing engine, even cable access holes can be small and unobtrusive. And iQ lets you choose between free-standing 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

## CHOOSE YOUR OPTIONS

You won't need to cut up your studio furniture for iQ — it's designed to sit right on the countertop. Just one slim CANBus cable connects the surface to the QOR.32.

iQ meters use bright, high-resolution OLED displays for instant readability, with VU or PPM metering options.

Meters, clock and timer are all on the meter bridge - no external monitor needed.

Options knobs give fast access to source selection, pan and balance and other frequently-used controls. Turn them to make an adjustment, push them for even more options.

One-touch Record Mode button makes it easy for operators to quickly record phone calls or other off-air bits for later playback.

Studio Monitor controls include Talk To keys for board-op communications with in-studio guests, or remote codecs and phone callers using Talk To Backfeeds function.

Light bulbs? Not here. Bright, long-life LEDs illuminate all iQ buttons.

Monitor and headphone choices include all four Program buses plus two external audio sources.

Show Profiles let you save frequently-used console configurations for instant operator recall.

Like its big brother, Element, iQ generates an automatic mix-minus for every source that needs one.

The Telos iQ6 broadcast phone system works with Telos VSet phones, or with the hybrid controls built into your iQ's Telco expansion frame. Connect it to the QOR.32 console engine with a CAT-6 cable, plug in your phone lines, and start taking calls.

What's better than great-looking? Long-lasting. iQ's anodized machined-aluminum surfaces are laser-etched. No painted markings to rub off or fade.

The 10 programmable keys on each side of the 6-Fader+User Buttons expansion frame can be mapped to GPIO functions for control of audio devices or other logic commands.

Like all Axia consoles, iQ is built tough. These 100mm. conductive-plastic faders feel smooth as silk, but they're built to handle the punishment 24/7 operation dishes out.

iQ's high-resolution OLED fader displays normally show the source that's assigned to each channel strip, but change to give additional information when operators choose new audio sources, adjust pan & balance, or perform other tasks. They can also work with the Soft Keys just below to trigger GPIO commands, step automation events and more.

Full-featured Telco controls include Next, Xfer, Block and Hold keys.

6-Fader Telco expansion frame has built-in 6-line controller that works seamlessly with the Telos iQ6 phone system. The familiar twin hybrid controls use OLED displays with exclusive Telos Status Symbols for instant caller information; dial pad lets operator dial out directly from the console.

Desktop-mounted iQ frames don't need a furniture cutout. Remove the end moldings to attach frames together, or keep them separate - the choice is yours.

# iQ

## CHOOSE YOUR iQ

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



### iQ MAIN FRAME

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

### iQ 8-FADER EXPANSION FRAME

Double the size of your iQ instantly! Plugs into the QOR.32 integrated console engine to add eight more faders to your iQ Main Frame. Like all iQ frames, it comes equipped with Axia's rugged anodized machined-aluminum surface, conductive-plastic faders, aircraft-quality switches and LED button lighting.

### iQ 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 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.

### iQ 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 our new Telos iQ6 six-line telephone hybrid (more details opposite). 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.

### QOR.32

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 or Audio Nodes. QOR.32 is convection-cooled for utterly silent, fan-free operation.

### 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.

# iQ6 TELCO GATEWAY

## IQ MAKES PHONES SMARTER, TOO

Up to now, integrating phones with your on-air console could be time-consuming. iQ6 (the telco gateway from Telos that's designed for Axia iQ consoles) makes it simple: one CAT-5 connection to your QOR.32 console engine, and hookup's done.

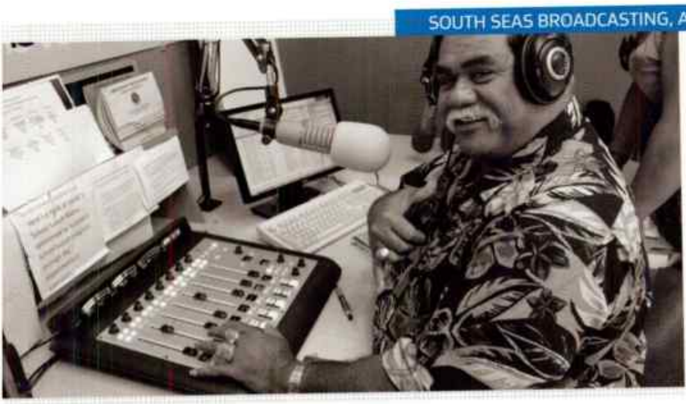
iQ6 works with the iQ 6-Fader Telco expansion frame to give operators seamless, on-console control of incoming lines and callers. Take calls, dial out, step through pre-screened callers without ever taking your hands or eyes off the board. Familiar Telos Status Symbols icons let board ops know what's what with just a glance. No need to construct a mix-minus, since iQ consoles do that for you automatically; just take a caller to air and they're fed a customized mix-minus without any need for operator intervention.

If you prefer to use iQ6 with a traditional phone-type controller, you can. Telos VSet6 six-line phonesets, with big color screens and animated icons, make fast work of pre-screening incoming calls. Or, use XScreen Lite software from Broadcast Bionics (provided free with every iQ6) for a full-featured software screening environment.

iQ6 works with POTS or ISDN phone lines, and comes equipped with two high-performance Telos DSP hybrids with Digital Dynamic EQ — the same advanced hybrids found in our Nx phone systems.

Naturally, since it's part of the iQ console system, iQ6 saves you money and time. It plugs right into one of the Livewire ports on your QOR.32 console engine — no need to solder discrete I/O cabling and connectors — your installation time is drastically reduced. Just plug in your telco lines, connect iQ6 to your Axia network with a CAT-5 cable, and after some fast Web-based setup, your talent will be taking great-sounding calls to air.

SOUTH SEAS BROADCASTING, AMERICAN SAMOA



ANDY STEVEN, 60 NORTH RADIO

"We chose Axia Radius and the entire technical team immediately fell in love with it!...Our on-air talent have all been impressed with the flexibility and setup of the Axia Radius Console - it really is so easy to configure."

# RADIUS

## SPEND LESS. GET MORE.



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

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

Radius is the easiest AoIP console ever. Just connect the 8-fader mixing surface to the QOR.16 integrated console engine, plug in your sources and power, and you're ready to make great radio. Because it's so compact, Radius is the perfect standalone console, but Gigabit ports on its QOR.16 integrated console engine let you connect it to other studios too. Radius' network gateway lets you load either 8 or 12 audio sources from anywhere in your Livewire network, while sending either 4 or 8 locally-produced streams (depending upon your configuration choice) 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 consoles costing much more.

No external monitors needed; easy-to-read segmented LED meters are built right in.

Three stereo mixing buses plus a stereo utility bus for plenty of mixing capacity.

Clock and event timer, right where you expect them.

Silky-smooth, side-loading 100mm. conductive-plastic faders beg for your touch.

High-resolution OLED displays show source assignments, pan and balance information, Soft Key actions and more.

Machined aluminum construction. Built tough for 24/7/365 use.

Quickly record phone bits or off-air segments using one-touch Record Mode.

Four Show Profile slots let operators recall frequently-used console snapshots.

Monitor section gives board ops control of control room and studio monitor sources and volume.

# QOR.16

## INTEGRATED CONSOLE ENGINE

Audio I/O, GPIO, console CPU, super-duty power supply, 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. And you can daisy chain as many as four QOR engines without the need for an external Ethernet switch, making installation even more economical.

Plenty of professional, balanced mic, analog, AES and Livewire I/O in a fan-free 2RU chassis.

Built-in Ethernet switch lets you network devices and studios easily.

Rugged, built-in super-duty power supply. No line-lumps or wall-warts on Axia gear.

It doesn't just look cool - it stays cool, thanks to beefy heat-sinks and fan-free design.

# MEET RAQ AND DESQ

## BIG CONSOLES 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.

Meet RAQ and DESQ, two compact, special-purpose IP consoles from Axia. These slim units fit neatly on the corner of a desk, or in a 4RU rack space; anywhere you need a little bit of mixer. They may be compact in stature, but they're big on features and performance. RAQ and DESQ both have "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 they easily out-class mixers that take up much more room.


# RAQ

## RACK-MOUNT IP CONSOLE

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

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

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

# DESQ

## COMPACT DESKTOP IP CONSOLE

DESQ is a six-fader console too, but in a form-factor that lets it fit just about anywhere there's a few inches of spare space: DESQ is only 16 inches (39.9 cm) square. It's built Axia-tough, with a machined-aluminum work surface that takes the rough stuff jocks can dish out. Our familiar 100 mm. side-loading 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 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.

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.

# QOR.16

## INTEGRATED CONSOLE ENGINE

RAQ and DESQ are powered by our QOR.16 integrated console engine. It's fanless and silent, so you can deploy it anywhere — even next to the mic. A single cable connects it to your RAQ or DESQ mixer. It's got everything: two Mic inputs, eight analog inputs and four analog outputs, an AES/EBU input and output, four machine-logic GPIO ports, and eight Livewire ports, four with PoE to power studio accessories like Telos VSet phones and Axia xNode audio interfaces.

Although they're perfectly suited to standalone operation, RAQ and DESQ are IP-Audio consoles too, ready to connect to a Livewire network. That's thanks to the zero-configuration Ethernet switch with Gigabit that's built in to every QOR.16 (and every other Axia integrated console engine, too).

And here's the kicker: one QOR.16 can power two RAQ or DESQ mixers — or one of each! Despite all these features, RAQ and DESQ are 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 broad-casting, or as personal mixers. What else could you use them for? The possibilities are endless...

**TOM WIEBE, GOLDEN WEST RADIO**

"In the bad old days, we used to spend weeks living in hotels, working long days pulling multi-conductor cable, wiring punch blocks and soldering Christmas trees. Today, we're able to do a complete Axia install in three days with one or two engineers."

# AXIA ACCESSORY PANELS

## CONTROL, WHERE YOU WANT IT.

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

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.

(For all Axia consoles.)

### PRODUCER MIC

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.

(For all Axia consoles.)

### HEADPHONE SELECTOR

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.

(Element consoles only.)

### MIC CONTROL/ HEADPHONE SELECTOR

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.

(Element consoles only.)


### FIVE-KEY BUTTON

Five-key Button Panel can be placed wherever remote control of contact closures or routing commands is desired. Film-legendable keys contain LED backlights with individual color settings, and work with Pathfinder routing controllers to put fingertip control right where it's needed.

(Element consoles only.)

### FOUR-KEY SMARTSWITCH

Four-key SmartSwitch has illuminated, dynamic LCD keys that can change text based on conditional logic macros you construct in Pathfinder using simple drop-down tools.

(Element consoles only.)

# AXIA ROUTING CONTROLLERS

## FINGERTIP CONTROL JUST WHERE YOU NEED IT.

Axia control panels, used with Axia Pathfinder routing control tools, let you put routing power anywhere — in a studio turret, a TOC control panel or an equipment rack. These accessory control panels work with Axia's Pathfinder software and Pathfinder Core 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 (in rack-mount, cabinet-mount and Element console drop-in versions) with 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.

A family of three Routing Control Panels (X1, X2 and XY) allow convenient on-the-fly routing of networked sources from anywhere in your facility. X1 and X2 controllers let you route any networked audio source to a pre-selected fixed output; XY controller lets you dynamically route any source to any output of your choosing.

The Axia Router Selector Node enables fast, easy selection of any source on your IP- Audio network. There are eight convenient front-panel pushbuttons you can map to frequently-used sources, while the "tuning knob" and display screen allow browsing your entire network; perfect for news booths or dubbing stations where only one active feed is required. The Router Selector also provides a selectable amplified feed to headphones as well as analog/digital inputs and outputs. Great for allowing non-technical folks to easily move audio from external sources (like field recorders) into the Axia network.

8-BUTTON OLED SOFTSWITCH BUTTON PANEL

9 AND 17-BUTTON SMARTSWITCH PANELS

5, 10 AND 15-BUTTON FILMCAP PANELS

ROUTER SELECTOR NODE

X1, X2 AND XY ROUTER CONTROL PANELS

CABINET MOUNT BUTTON CONTROLLERS

# IP INTERCOM

## GO AHEAD: TALK AMONGST YOURSELVES.

IP Intercom comes in several rack-mount and desktop styles, plus drop-in modules for Axia Element consoles. Mix and match to build your customized intercom system!

Imagine a digital intercom system with no central matrix. Actually, don't bother — we've built one. Axia IP Intercom saves on cost, space, and installation time, and eliminates special plug-in cards altogether. It's real plug and play that works every time — even when you need to add a station, or reconfigure the ones you've got.

Everybody knows the advantages of IP and Ethernet – low cost, easy installation and maintenance, efficient infrastructure. Thanks to its efficient Ethernet backbone, installing IP Intercom is a simple single-click connection. Of course it's easily scalable: plug as many stations into your switch as you want and add on from there. Then start talking! And if you move to a new location, you can just pick up the gear and take it with you — there's no expensive, hard-wired, custom-cable multi-pair infrastructure mess to deal with.

Don't have an Axia studio network? That's OK. You'll still save money and increase efficiency by choosing IP-Intercom; it's a stand-alone system with I/O that will accommodate multiple consoles. But if you do have an Axia system, you'll get seamless console integration that gives your operators benefits other systems can't. For instance, you can take broadcast-quality intercom audio directly to air. And you can feed IFB audio directly to intercom callers.

IP Intercom gives you unlimited full-bandwidth access to any studio, news or sports venue, office, hallway, broom closet or wherever. Talk and listen to individuals or groups hands-free, with no echo or feedback — IP Intercom features exclusive AEC advanced echo cancellation from Fraunhofer Labs (the inventors of MP3), so there's never any open-mic feedback during conversations. Ever.

IP Intercom system is completely digital. Other intercom systems try to make you think they're digital by piping their analog signals over CAT-5 cables, but the last thing you need during a breaking story or transmitter failure is hum and buzz getting between you and the guy you need to talk to. With IP Intercom, there isn't any.

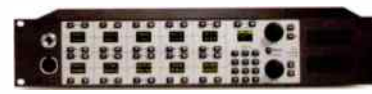
So you've gotta be a genius to use it, right? Actually, anyone with an index finger can operate this system with ease. The web interface makes setup simple. Sharp, high-contrast OLED displays are easy to read from anywhere in the room. And our clever callback feature makes sure you'll never miss a call, no matter what you're doing.

## INTERCOM+ By Broadcast Bionics

Expands Axia Intercoms to two or more geographically diverse sites over a corporate WAN or even the Public Internet. Intercom+ runs as a Service on a Windows PC providing reliable gateway connectivity with Livewire-connected Intercoms nearby or far away. Speak freely with colleagues at remote Axia studios and enjoy convenient cueing and collaboration. Intercom+ even maps your enterprise network and allows you to manage all your intercom units from a single location and simple GUI.

## IC.20

The IC.20 twenty-station intercom panel has 20 presets presented on high-resolution OLED displays for quick contact with frequently-called stations. There's a keypad and display for fast access to stations system-wide, plus group talk and auto-answer functions.



## IC.10X EXPANDER

The IC.10X Expander pairs with the IC.20 panels. It adds 10 station presets (each with a sharp 10-character OLED display, and talk and listen keys) to your existing IC.20 for a total of 30 station presets. A single Ethernet connection is all that's needed for hookup.

## IC.10

IC.10 has 10 station presets with OLED displays. Like its big brother, the IC.20, it has a built-in speaker, front and rear-panel mic connections, 4-pin locking headset jack, analog I/O presented on both XLR and RJ-45 connectors, and a GPIO connection for speaker mute/dim and external line-status tallies.

## IC.1

The IC.1 ten-station intercom panel has ten LED-backlit film-cap buttons that can easily be labeled with station names. Buttons are programmed using the built-in Web interface and any browser.

## IC.20 DESKTOP

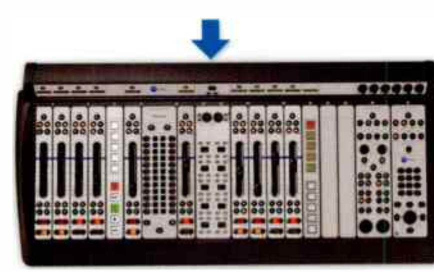
Desktop version of IC.20 puts intercom access on any studio surface. Perfect for producers' or screeners' positions, operations centers, etc.

## IC.1D

IC.1D has 20 preset stations presented on LED-backlit button caps; an economical way to add intercom function to any space.

## ELEMENT MODULES

Add IP Intercom directly to your Axia Element mixing console! Several drop-in modules make it easy to quickly take broadcast-quality intercom audio to air, or record off-the-air. See Page 67 for more details.



## SOFTCOM

Axia Softcom Intercom for Windows makes any networked PC a part of your IP Intercom system! The easy user interface mimics the IC-20 control panel, with preset locations for 20 frequently-called stations. Auto-answer and hands-free functions are supported, and a drop-down station finder gives instant access to stations not pre-programmed.

# AXIA ROUTING
## POWER AND FLEXIBILITY AT YOUR FINGERTIPS

IP-Audio is much more than just an easy way to connect audio gear using Ethernet. An IP-Audio network provides sophisticated, high-speed links between every device in your studio. Once everything's connected, it's a cinch to use the data routing capabilities of the network to send audio anywhere you want it. Change air studios at the touch of a button... automatically switch satellite feeds to air... even listen for dead air and automatically switch to backup audio.

With Axia, you can make use of a wide variety of hardware and software-based tools designed to help you easily harness this power. After all, what good is power without control?

# PATHFINDER
## ROUTING AUTOMATION

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.

### 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.

TROY MAJESKA, FAR EAST BROADCASTING COMPANY

"Axia's tech support [is] truly the best in the industry! Convenient times (24/7), able to log in remotely, multiple knowledgeable staff members, and eagerness to help. You set the bar, and every install I do will have Axia at its core because of that."

## 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 requirements. PathfinderPRO supports server "clustering" – running simultaneously on two connected, yet independent computers – for the ultimate in redundancy and security.

PathfinderPRO all of Axia devices and supports as many end-user connections as your CPU can handle. PathfinderPRO can directly control console VMix virtual mixers, Element 2.0 motorized faders, Show Profile changes, and more. 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.

## PATHFINDER CORE PRO

The new Pathfinder Core PRO network routing appliance builds on years of routing experience to deliver the power of Pathfinder software in a dedicated hardware device — no Windows server required. You program Pathfinder Core PRO via the Web browser on your PC, using the intuitive graphical interface provided by its built-in webserver. Pathfinder Core PRO gives you peace of mind by moving your critical routing controls from PCs to purpose-built hardware. It also gives you freedom — freedom from concerns about software compatibility, automatic OS patches, and computer hardware limitations.

Pathfinder Core PRO is fast, efficient, and simple to use. Just connect the 2RU appliance to your network with a CAT-5 cable, assign an IP address, and Pathfinder goes to work, automatically detecting your Axia audio sources, destinations and GPIO ports. In just a short time, you'll be ready to build routing commands as sophisticated (or simple) as you can imagine. Using Boolean logic, you can construct powerful new Core Events command sets to initiate anything from a single route change to system-wide, cacscading scene changes.

Pathfinder Core PRO is fanless, with dual-redundant, field-replaceable Telecom-grade power supplies and embedded Linux platforms to ensure the ultimate in reliability. And, for the ultimate in distributed redundancy, two or more Pathfinder Core PRO appliances may be "clustered" to provide automatic, distributed backup of your vital routing functions.

# STUDIO SOFTWARE FROM AXIA

## CONTROL, CUSTOMIZE, CONNECT

### AXIA iPROFILER

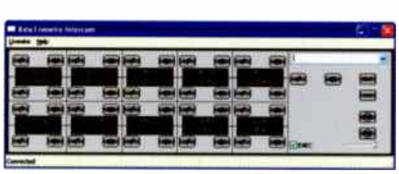
Axia's networked version of the popular Telos ProFiler logging software lets you simultaneously capture up to 24 stereo audio channels to time-stamped MP3 audio logs directly from your Axia IP-Audio network — no audio cards required.

iProFiler is an all-in-one program audio logger, aircheck skimmer, and remote listening application. The iProFiler suite includes software that lets you record, manage and play back archived audio files on any standard Windows PC. Modes include logging (continuous archival storage), skimming (records only when talent mic is open), reverse skimming (records only when talent mic is closed), and SmartSkimming (low-bitrate logging switches to a higher bitrate when talent is on-mic for best quality captures).

Production Directors and program producers love the integrated audio browser that lets them tag segments and export WAV files for editing. And you can listen to "live" audio over IP as it's being logged — perfect for program consultants or group PDs.

### iPLAY

Axia iPlay PC software allows any Windows PC to listen to streamed audio directly from your Axia network. Give PC monitoring capabilities to PDs, GMs and sales staff using their existing computers. Choose from a list of all available audio, or use the eight user-programmable preset buttons for quick access to favorite channels; on-screen level display meters auditioned audio.



### SOFTCOM

Make any networked PC with a sound card part of your Axia IP Intercom system. Axia SoftCom software works just like our rack-mounted Intercom hardware panels, with preset locations for 20 frequently-called stations. Auto-answer and hands-free functions are provided; an easy drop-down station finder gives instant access to any station in your facility. Convenient site licensing lets you run SoftCom on any – or every! – PC in your network.

## SOFTSURFACE

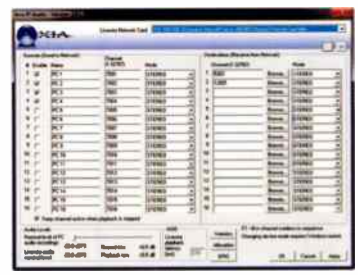
New SoftSurface Virtual Console software for Windows gives you powerful real-time control of your Axia Element mixing console from home, office, or anywhere an Internet connection is available. Take direct remote control of your Element, or, pair SoftSurface with an Axia StudioEngine to create a "virtual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

## iPROBE

Axia iProbe is an intelligent network maintenance and diagnostics suite that consolidates managing, updating, and remote-controlling your Axia system into one easy-to-use software application. There's a powerful Auto-Documentation feature that queries and documents configuration settings for every networked Axia device — great for administering large networks. The Organizer tool lets you perform many advanced tasks, such as gathering Axia Audio Nodes 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 — all from the comfort of your home or office.

## IP-AUDIO DRIVER

The Axia IP-Audio Driver lets you send and record as many as 24 stereo channels of PC audio directly to Axia networks via Ethernet — no sound cards needed. It 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. Ask your favorite delivery-system provider if their software is Livewire capable. A single-stream version is also available for use with production workstations or individual PCs.



# xNODES

## WORLD'S MOST ADVANCED IP-AUDIO INTERFACES

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.

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 easy to deploy, too. They're fanless, so you can tuck one anywhere you need I/O. They're compact: two xNodes fit side-by-side 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.

Finally, xNodes have amazing audio specs. They operate at a network sampling rate of 48 kHz and have high-resolution 32-bit floating-point SRC chips with an astonishing dynamic range. Coupled to studio-grade sample rate converters, xNodes produce a "sweeter," more natural audio quality — clients routinely tell us of noticeable sonic improvements after the installation of their Axia network.

### MICROPHONE xNODE

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, to suit your connection preference.

### ANALOG xNODE

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.

### AES/EBU xNODE

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

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.

# xSWITCH

## THE NETWORK SWITCH BUILT FOR IP-AUDIO

IP-Audio is easy, right? Simple as plugging in an Ethernet cable. Except for one thing: programming the switch. Meet xSwitch, the world's first network switch custom-built for Livewire IP-Audio applications.

With xSwitch, there's zero configuration – the network experts at Axia have done it for you. Press a single front-panel button (or use the built-in webserver) to enter an IP address, and xSwitch is ready. No config files to upload, no command lines to type in – just connect and start routing audio.

Like their xNode brothers, xSwitch is compact (two can fit side-by-side in 1RU) and fanless (those big heat sinks on the front don't just look cool), so you can deploy them anywhere, in even the tightest locations. Each one has eight 10/100 Livewire ports (four with PoE – Power over Ethernet) and two Gigabit ports with SFP that provide redundant copper or fiber connections with auto-switching.

With all this, xSwitch is the perfect way to add network capacity nearly anywhere. Use it as a studio edge switch, at transmitter sites, to reduce the number of needed core switch ports, or cluster an xSwitch with up to eight xNodes to create a "supernode" for use in studio clusters.

# LIVEWIRE: CONNECTED

## MORE PARTNERS MEANS A MORE VALUABLE NETWORK

In 2003, Axia introduced Livewire, a real-time AoIP networking system using Ethernet for uncompressed digital pro audio. Today, Livewire is the broadcast industry's de facto standard for networking studio equipment. More than 3000 radio studios are equipped with Axia AoIP networks, with over 25,000 Livewire-equipped devices in service.

We believe that broadcasting's future is networked. Enabling broadcast devices to talk intelligently with one another opens up efficient new ways for talent to operate. Networking removes complexity from the broadcast plant, just as it enhances the way equipment works together. Imagine hooking up a multi-line talkshow system to your console with just one cable – and then commanding that system with controls built right into the board. You don't have to imagine: Axia Livewire lets you do that right now, today. Operating more efficiently, decreasing costs of maintenance and installation – that's what IP-Audio networking is all about.

Plenty of broadcast companies share our networked vision, and have become Axia Partners, producing software and hardware that "speaks Livewire" natively. But we won't be happy until everything in the studio is networked. So, we've decided to make Livewire a gift to the broadcast community: we've introduced the Livewire Limitless License (or L3, for short). Under L3, anyone who wants to put Livewire into their products can do so without the cost of ongoing licensing fees. We've published our designs, opened our source code and even shared our specification documents and development information with our partners. Even easier – manufacturers can use the little interface card shown here to add Livewire to their products just by plugging it into their designs. Companies like Studer, Orban, MAYAH and AEV have already jumped on board, with more to come.

LIVEWIRE® INTERFACE CARD

## PARTNER PRODUCTS

Here are just some of the products from Axia partners that connect directly to Livewire networks:

MICROMEDIA MADI NODE


AUDIOSCIENCE 6585
LIVEWIRE SOUND CARD

INTERNATIONAL DATACASTING
SFX4104 EXP SATELLITE RECEIVER

FRAUNHOFER CONTENTSERVER

SOUND4 IP PCI
AUDIO PROCESSING CARD

PARAVEL SYSTEMS IROUTE

NAUTEL VS1 FM TRANSMITTER

# LIVEWIRE AND RAVENNA, NOW CONNECTING.

## WHY THE TWO LEADING IP-AUDIO PROTOCOLS ARE WORKING TOGETHER.

At NAB 2012, we raised quite a few eyebrows with the announcement that Livewire, the originator of IP-Audio networking for broadcast, and RAVENNA, the IP-Audio protocol from ALC NetworX, had decided to become partners.

Many folks have asked us why we've partnered. After all, Livewire introduced the idea of IP-Audio to broadcasters in 2003, and it's been very successful! There are more than 45 Livewire partners, like Nautel, AudioScience, Telos and RCS. Over 4,500 consoles with Livewire are installed, and more than 50,000 Livewire-equipped devices in the field.

In 2007, ALC NetworX, launched by the well-known German console maker Lawo, announced RAVENNA. It, too, gained prominence and established a network of partners anchored by respected European manufacturers such as Genelec, Neumann, AEQ and AETA.

It didn't take long for Livewire and RAVENNA to realize that what both of us were doing was very similar. Both are standards-based open technology platforms. Both share core philosophies and a desire for broadcasters to have equipment that goes together easily, works seamlessly, and increases efficiency. Both have established clients and partner networks. Wouldn't it be cool to make all those networked devices work together?

Now that we've agreed to this IP Audio standard, with a huge installed base of products and combined partner networks, the question that emerges is: What about competing protocols? We're sure you're going to hear from some other console company who will tell you that they've got the protocol that's going to be the standard in the future. Well, one or two devices does not a network make. Ask them how many partners they have whose products work on their system today. If they still sound confident when you ask that question, ask them: Aren't you the same guys who said IP-Audio would never work?

For IP-Audio to continue to grow more valuable, there must be an increasing number of networked broadcast devices. RAVENNA and Livewire are the two leading protocols in broadcasting. Together, we have 60 technology partners whose devices interconnect. Our collaboration paves the way for products from both groups to inter operate, giving broadcasters more choices.

Livewire products continue to work as they always have. And if you want to add RAVENNA devices to your Livewire network, a firmware upgrade for existing Livewire devices will add a RAVENNA-compatible mode to allow Livewire and RAVENNA devices to interoperate.

When will compatibility be phased in to Livewire devices? We've already begun! The first RAVENNA-compliant devices – Axia xNodes – are here now. We expect to add more new RAVENNA-compliant devices over the next few years. And we will implement RAVENNA-compatible mode in all Livewire devices.

How is Livewire different from RAVENNA? RAVENNA uses a different clocking scheme than Livewire and it requires a different hardware interface design. You must have an IEEE-1588 clock master for the RAVENNA devices. And you must have a RAVENNA/Livewire device on the network that can synchronize to the 1588 clock and then offer a synchronized clock to the Livewire devices. Axia's new xNodes can provide both functions. One xNode placed on a Livewire network can provide the clock to RAVENNA devices and also to Livewire devices on the same network.

Think of this new RAVENNA broadcast protocol as Livewire 2.0. We worked together with RAVENNA engineers to shape this new protocol. It contains the best ideas of RAVENNA and Livewire, together. And we will continue Livewire support on future products. Livewire and RAVENNA will co-exist on the same network and interoperate in RAVENNA-compatible mode.

Axia's philosophy has always been "do what's best for the client." And it makes no sense for console companies to try to beat each other with "my protocol is better than yours" arguments. Nobody wins these, least of all the broadcaster. What's truly best is for everyone to work with the same protocol! For example, consider AES-3. Many products are compatible, including consoles from competing companies. The products are differentiated in other ways, but they all use AES-3. You never have to wonder if plugging Device A into Device B is going to work.

IP-Audio needs to be the same. Which is why Livewire and RAVENNA have decided to work together. And also why we are both members of the AES X192 project, established to define an International IP-Audio interoperability standard. This means that you needn't be concerned about future incompatibilities — your Livewire network will be standards-compliant from the start.

So how does all of this benefit you, the Axia fanatic? You have access to a large and ever-growing group of compatible products. More devices that network will simplify your system, reduce costs and increase flexibility. And if you have an existing Livewire system, and you are not planning on adding any RAVENNA products, there is no reason to change anything — your Livewire system will operate as it always has, giving you bulletproof reliability and performance 24/7/365.

# EVOLVING DIGITAL AUDIO NETWORKS

## TAKING THE 'STING' OUT


### ABSTRACT

Digital audio networking is here to stay. It is the state of the art, and it is the future. But with which standard? Today there are multiple competing systems, and work on a compatibility standard is underway. Knowing a digital audio networking standard is the future that will be, what should equipment vendors and users do in the meantime to get 'over the hump' of these next few years? This paper will examine this problem by breaking down where some of the real design issues are. Given the difference between the systems, what is the most that could need to be changed? How can an equipment manufacturer today designing to use one particular digital audio network interface potentially leave the door open to the future standard even though that is not yet settled? Broad categories of equipment design issues will be considered, like: network capability, synchronization, stability and design of local clocking hardware, stream formats, memory for buffering, mips for protocol parsing, real time requirements on hardware and software.


### GOAL STATEMENT

How can you design products today that will not need too extensive modifications to interoperate with different digital audio networks, and the future X192 AES standard whatever that settles to? There are signposts already pointing the way to the end goal, if you know how to see them.

We are at that point in the history of digital audio networks that will be looked back on as just before standardization happened.

The answer is to break down the implementation of an audio network interface design, and examine each part, and see how each part may vary. We will find that there is not too much variation, and with some careful planning and selection of design features, it will be possible to support many of the existing audio networks and be ready for X192.

### BOTTOM UP APPROACH

There are two ways to break down and describe a system: top down or bottom up. There are many white papers [1], books [2] and current and pending standards documents [3] that describe the design of audio network from generally the top down. To draw a contrasting approach, I will describe a bottom up interface design. At the end, we will find that there are a collection of 'pieces of the puzzle' that when put together, form the end goal of a more future proof design.

### PARTS OF AN AUDIO NETWORK INTERFACE DESIGN

- Timebase recovery (synchronization)
- Audio packet handling
- Audio data buffering
- Buffer synchronization
- Serial to parallel, parallel to serial
- Connection management, discovery and control

### TIMEBASE RECOVERY

Timebase recovery is the essential process of having all devices on the audio network use a common sample rate and common knowledge of sample phase, which in the case of packet buffered blocks, where the block boundaries (or 'frames' ) are at the same time. Examples of timebase sync are IEEE-1588 and the earlier Livewire sync method, which are similar in function, but vary in some details. All these methods use the mechanism of "clock packets" which contain a timestamp of the master clock at the moment they were sent, and are accurately timestamped with the local clock counter at the time of arrival.

The first thing to note is that accurate timebase recovery is directly related to, and essential for, low latency (low delay) of the audio going over the network. This is because the latency of an audio network is not the time of flight of the packets over the wire or packets though the switches (although over very long distances the speed of light does eventually dominate), the latency is determined by the amount of buffering used, and the minimum amount of buffering that can be used is determined by the time inaccuracy, the wander and jitter of the transfer of the data in time, which is all coordinated by the timebase as recovered at each device relative to each other. So less accurate timebase recovery implementation would require more buffering, and would be only able to serve a higher latency audio network, but an accurate timebase recovery can serve both low and high latency audio network designs.

If you want a very low latency audio network (for example the 750 microsecond "Livestream" mode of Livewire), this corresponds to an audio sample block of 12 samples per packet at 48Khz sample rate, and an audio frame / packet rate of 4Khz (250 microseconds). To do this in practice, we use a timebase recovery accuracy of +/- 5 microseconds, which is about ¼ of a sample. This very tight timebase accuracy is so the device can meet the schedule of taking input audio packets, performing required DSP, and transmit the output packets before the deadline of the next 250 microsecond frame. Any error in the local timebase relative to any other device takes away time that you have available to do processing of the audio. You can pipeline the audio frames' input / process / output sequence at the expense of added latency, but timebase error still threatens the ability to know the moment of time when all the input packets are present at the shifted execution time, and threatens making output transmissions by the frame deadline.

It is possible to make use of a less accurate local timebase, even a purely software operating system task scheduler, as for example in purely software IP audio drivers for Windows / Mac OS / Linux are available to replace hardware sound cards. But these software-only drivers cannot generate the lowest latency type of streams. Example: A typical OS task execution time accuracy is +/- 10 milliseconds (or more), with some of the better more modern OS's down to a few milliseconds. These require larger buffers at the receiving end to cover the transmit timing error, and this buffering causes the audio latency, in this the example just given, on the order of tens of milliseconds.

To reiterate: an audio network interface product with a more accurate timebase can always interface with and easily meet the wider frame deadline of a higher latency audio network with bigger buffers. But the converse is not true, a device with less accurate timebase cannot interface with a lower latency audio network.

**Puzzle Piece #1: Design for high timebase accuracy, which will not limit you from interfacing to any audio network (high or low latency).**

The more accurate the time stamps are, the better, and less time measurement error is introduced into the timebase recovery process. This is analogous to less noise input to a time recovery Phase Lock Loop (PLL). The timestamping can be done in software (part of the packet receive interrupt), but the interrupt response latency then becomes timing error. (You may not want a high priority interrupt per packet, anyway. There can be many, many packets, hundreds of thousands of packets per second coming to your device in a low latency multichannel audio network system.)

The solution is to use an FPGA to monitor the incoming packets, and either timestamp all of them or have a recognition filter that finds the clock packets by pattern matching and performs the time stamp, and makes this time stamp information available to software which will be running a software PLL timebase recovery algorithm.

IEEE-1588 enabled Ethernet hardware contains this same kind of timestamping mechanism. Although with the present state of the art, you must pay close attention to the hardware features that are implemented in the chip you choose, and the features that the PTP (Precision Time Protocol) stack expects. These must match. Also depending on the assumptions made, an existing PTP stack implementation may or may not produce for you a usable hardware clock that you can run the sample clock of analog/AES converters from (if you are interfacing).

**Puzzle Piece #2: Use an FPGA based incoming packet timestamping module. Stamp all packets, or be able to have its recognition filter configured to accommodate any of the sync clock packet formats.**

Sync recovery is essentially a PLL that steers a local programmable oscillator, in order to match the external clock master. This sync method using relatively infrequent clock packets means that you don't get a very large amount of sync 'steering' information. In PLL terms, this means the bandwidth of the PLL feedback loop is low. When your PLL feedback bandwidth is low you benefit from the most stable local oscillator that you are using. A PLL must steer itself to overcome 3 things: to track the remote master clock, to overcome the time noise (clock packet time jitter) and to compensate for the drift of the local oscillator.

Hitching a ride from the digital cell phone industry it seems, a class of inexpensive TCXO (Temperature compensated crystal oscillator) of about +/- 2ppm are available for very reasonable cost around $2. The net system performance this enables is worth this cost, to have a stable local timebase reference as compared to the difficulties created by the standard variety +/- 20 to 50ppm crystals. A clock generator design that give simultaneously both stability, low jitter, and a wide range of high resolution programmability, which will be a solution for any audio network solution at any sample rate, is the Direct Digital Synthesis (DDS) method. This can be a chip, or implemented in FPGA.

**Note:** if you are to have analog audio converters in your product, or have AES digital local inputs and outputs with sample rate converters (SRC's) to external AES sample rates, you use this master DDS generated timebase as your master overclock for the converters and the sample clock to the 'inside' of the AES SRC's.

Of course, the very nature of having an audio network allows you to eliminate analog or digital interfaces from the product design, if you so desire. In this case, of a pure audio network device, you still have the need of an accurate local timebase to generate the essential moment in time (an interrupt to the processor) to know when to take the audio packets that came in, perform signal processing, and generate audio packets out to the network. Any error in the moment of time that this process is triggered, represents being possibly late delivering output audio packets and in a low latency system, could underflow the small buffers of the next device in the audio path. The existing crystal associated with the CPU may drift too much, and the scheduling time resolution of the existing CPU resources may be too coarse, so that the accurate TCXO plus DDS may still be desired.

**Puzzle Piece #3: Use a local clock oscillator design consisting of a stable TCXO with a DDS architecture.**

The remaining part of audio network timebase recovery is essentially a "software PLL" that uses the differences between the received timestamps of the clock packets and the master clock time value carried inside the clock packet, to carefully speed up or slow down your steerable local clock (TXCO + DDS). The details of how this is done vary for different audio network sync schemes, but the point here is that this should be software. While it may be possible to implement more of the PLL control algorithm in hardware ASIC or FPGA, the observation is that the rate of clock packets is comparatively low (1 packets per second for IEEE-1588 up to 30 pps for Livewire), so there is no real need not to handle this algorithmic complexity in software. In fact, the standard IEEE-1588 Precision Time Protocol (PTP) stack can do much of this work for you. The catch is the assumptions made by the PTP stack as to how the local timebase is generated and steered. A standard PTP stack may only be assuming that it is steering the kernel task scheduling mechanism, not a physical clock. But this can be made to work with the right modifications to the PTP hardware driver.

**Puzzle Piece #4: Use a software PLL for timebase recovery.**

### AUDIO PACKET HANDLING

The assumption is being made that the actual network interface (Ethernet) is an embedded interface in a general purpose CPU. Ethernet interfaces themselves are highly evolved (multiple generations of design technology since the 1980's), very efficient using fast block DMA offloading practically all load from the CPU, and very low cost, practically zero cost, being included in the core platform of the CPU. For economical audio network interface design, we want to take advantage of all of the above factors and use the CPU embedded Ethernet interface, and avoid adding as much external special function hardware as possible.

Multichannel uncompressed audio involves a relatively large amount of data throughput. If the data word and bit format of the audio samples and sample blocks do not exactly match the internal format of your processor, or if the audio channels are not packed or interleaved in a format compatible with your processor, this may represent a very significant CPU MIPS load, to unpack, possibly bit shuffle, and repack the audio network packet format.

A CPU is very efficient moving blocks of data, either using a single cycle repeated 'movem' type opcode, or with DMA hardware assistance. FPGA's are very efficient at byte and bit manipulation, packing and unpacking.

So the solution is to let the CPU be limited to doing block moves of audio data to and from the payloads of audio packets. The audio data block moves go through a high bandwidth interface to and from an FPGA. The FPGA does all of the bit level manipulation as needed, and channel interleaving, adapting to or from the details of the audio network data format.

**Puzzle Piece #5: Let the host CPU do only block moves of audio data, the CPU never touches audio samples individually. Use an FPGA to adapt / convert / repack or construct the detailed audio network packet payload format.**

Must there be an FPGA?

No, but consider the following: If your product has analog or AES local I/O, it is going to need multichannel parallel to serial / serial to parallel conversion, which is best performed in an FPGA. The timestamping mechanism described above wants to live in an FPGA (unless you are strictly using IEEE-1588 enabled hardware for IEEE-1588 sync only). The high accuracy timebase requires logic. This is a tradeoff between more cost for more CPU cycles as opposed the FPGA cost. If your design needs an FPGA for any of the other reasons, then partition the audio sample bit and packing functionality into the FPGA as well.

### AUDIO DATA BUFFERING

Buffering may be done in the host CPU's memory and in the FPGA. Or alternatively all buffering in the FPGA. Remember for low latency audio, the buffers are small, so not so much FPGA RAM is needed for buffering. For high latency larger buffers, CPU host memory is more economical, holding audio in blocks in the format of the audio network packet payload.

**Puzzle Piece #6: Audio data buffering can be in the host CPU memory (as packet payload formatted blocks), in the FPGA, both, or only in FPGA.**

### BUFFER SYNCHRONIZATION

Buffer synchronization means knowing where in the buffer to put the latest arriving input data, and from where in the buffer to take out audio data to be sent at present instant of time, as a function of absolute time and /or the immediate relative time to network frames.

Buffer sync can be tricky. Pointers have to be initialized, the data transfer process has to be initiated at just the correct time, then increment the pointers exactly to match the data flow. Finally the buffer pointer logic has to be able to recover from abnormal conditions such as missing packets, momentary network disconnect or malfunction, and transient conditions of timebase recovery (either initially or transfer of clock mastership events). If you want to give the acid test to a audio network product design, in the middle of audio streaming, unplug and replug the network cable a couple of times in quick succession, and see if the audio recovers completely and cleanly in a relatively short time (a fraction of a second, ideally). If not, this is a sign of buffer synchronization logic problems.

While it is possible to implement buffer sync logic in FPGA, the number of special cases point to a software implementation, that guides or command FPGA buffer pointers as needed.

It is to be noted here that there can be relatively major differences between different audio networks in this seemingly straightforward topic of buffer sync logic. The differences stem from the fundamental policy defined by the audio network, of when the samples of a given audio stream are to be played out. One policy is to begin to play out the audio stream as soon as possible, with as little buffering as possible as determined by the timebase accuracy of the receiving device. This has the advantages of simplicity of definition, and the ability to automatically adapt to long distance transmission where the time of flight of the packets is not negligible, but has the disadvantage that the absolute time that a given audio sample is used (or output) at the destination is not precisely defined. In practice this latter disadvantage is unimportant if the latency is low, and audio streams closely related to one another in phase (i.e. stereo pairs or surround channel sets) share the same packets in the same stream.

An alternate audio network policy is to play out the audio stream samples according to the correct absolute time as defined by the timebase information in the audio packets themselves. This has the advantage of precise definition of the use or output time of all audio samples and channels, regardless of how they are grouped together in shared packets or not.

Precise differences in these two policies, amounts to different audio buffer synchronization logic. For an audio network interface design to not be limited to one policy or the other, and to be able to handle the more complex special cases, the audio buffer logic should be in software.

**Puzzle Piece #7: Implement audio buffer synchronization logic in software.**

### SERIAL / PARALLEL INTERFACING

If the audio network device contains local analog or digital AES interfaces, or contains DSP functions that are best served their input/output data on multichannel serial ports, multichannel serialization is most straightforward to implement in FPGA logic.

Note that at all of these serial interfaces the timebase is still defined by the audio network. In other words, the bit clocks and frame signals are derived from the master DDS, which is steered by the software PLL to come into sync with the audio network master clock. A local interface, or DSP device cannot be the master clock for any of the interfaces which end up going to an audio network. An audio network requires all interfaces to be synchronous to its timebase.

If an interface or a device is fundamentally on its own sample rate or timebase, and cannot be made to use the audio network sample rate and timebase, audio sample rate converters (SRCs) will have to be used. Every SRC has two clock inputs, one for each side of the converter. In the case of an audio network, one side of the SRC will be connected to the audio network interface logic, and be a slave to the network interface logic master clock. The other side of the SRC can face the outside, non-network interface, and be a slave to that interface's master clock.

**Puzzle Piece #8: Use sample rate converters (SRCs) to interface all local audio interfaces or devices that cannot be slaved to the overall audio network sample rate and timebase.**

### CONNECTION MANAGEMENT, DISCOVERY AND CONTROL

An audio network accomplishes the connection and transport of many thousands of channels of audio in an efficient and timely manner. Imagine a giant, low cost, easy to automate, patch panel. But as soon as you have a complex operation, using thousands of channels of audio, and you care about what the end result sounds like, you are going to have to practically manage all of those channels.

### SOME BRIEF DEFINITIONS:

Connection management is the process used to make audio routes from point to point, once the user decides which input channel of audio is desired to appear at each output. The different connection management methods may range from using network broadcast addressing (no management at all, flood the entire network), to multicast (freely transmit all channels, receiver selects which channel), to unicast (arrange each transmit / receive pair on a one to one basis).

Discovery is finding out what audio channels are present, and what material they contain. This can range from a fixed prearranged order, on the fly data gathering, or a central registration and database lookup scheme.

Control is initiating or ending the audio connections if the endpoints need to be turned on and off, optional gain controls, initiating channel changes, etc.

The good news is that an audio network, being based on a data network, easily handles all of these required management tasks without any additional cables, wires, or interfaces. The bandwidth for connection management, discovery and control is a small fraction of the network bandwidth consumed by the audio, so with proper use of Quality of Service (QOS), can coexist. The bad news is that these methods are farther in the future from standardization than the audio part of the audio network itself.

**Puzzle Piece #9: Be prepared to be flexible with software implementations of connection management, discovery and control.**

## CONCLUSION

By breaking down the implementation of an audio network interface design into these flexible parts as we have examined here, they can be seen to be capable of not needing too extensive modifications to interoperate with many of the existing audio digital audio networks, and the future X192 AES standard whatever that settles to. By managing the design risk, products can be brought to market today without having to wait for future standards to settle, and with confidence that the future standards will be able to be supported.

## APPENDIX

### CASE EXAMPLE: WHAT IT TOOK TO EVOLVE LIVEWIRE EQUIPMENT TO BE COMPLIANT WITH RAVENNA

Livewire is a digital audio network system from Telos Systems, established over 10 years and in use in over 3000 radio studios worldwide. In the spring of 2012, it was announced that Livewire devices would embrace and support Ravenna, a contemporary digital audio network, because the value for all present and future Livewire customers is increased with interoperability. Support is also given by for the AES X192 initiative to define a interoperable digital audio network standard. Examining the details of what it took to make Livewire equipment able to fully support Ravenna, is informative and instructive to the subject of this paper.

Livewire and Ravenna both used a common IETF RTP (Real Time Protocol) L24 audio stream packet format, which defines all of the contents of the audio packets. What could be wrong? There were three main hurdles:

**Hurdle #1:** Ravenna relied on a specific use of the source timestamp in the audio packet, Livewire did not.

**Solution:** 1a) For compatibility with existing Livewire devices, Ravenna defined a new 'Livewire compatible relaxed mode', ignoring this value.

1b) The software driver in newer designed Livewire products will fill in the source timestamp value, to be fully Ravenna compliant.

**Hurdle #2:** Ravenna uses RTCP messages in the audio multicast channel to control audio streams. Livewire assumes streams are 'always on' and does not use nor expect RTCP packets.

**Solution:** 2) a software update for Livewire devices to safely filter out and ignore the extra RTCP packets.

**Hurdle #3:** Different sync methods. Ravenna uses IEEE-1588, Livewire uses a pre-1588 sync method.

**Solution:** 3a) As long as both systems externally slaved to the same master timebase, both are in sync with each other.

3b) Newer designed Axia products now can use IEEE-1588 directly, consisting of modifications to the FPGA clock packet timestamp filter, and modifications to the timebase recovery synchronization software PLL routine.

**Summary:** The required modifications to make the Livewire interface equipment also compatible with Ravenna were relatively minor, able to be changes in FPGA design and software drivers. This bodes well for being able to adapt to future standards like AES X192, when it settles.

## REFERENCES

[1] Various authors. White papers at www.axiaaudio.com/tech

[2] Church, Steve G. (2009). Audio Over IP: Building Pro AoIP Systems with Livewire. Focal Press.

[3] Audio Engineering Society, X192 High-performance streaming audio over IP interoperability. www.x192.org

**Greg Shay**
Chief Science Officer
The Telos Alliance

AMERICAN PUBLIC MEDIA | MINNESOTA PUBLIC RADIO

# MPR AND SCPR:

## TRACKING AXIA FROM MINNEAPOLIS TO LOS ANGELES

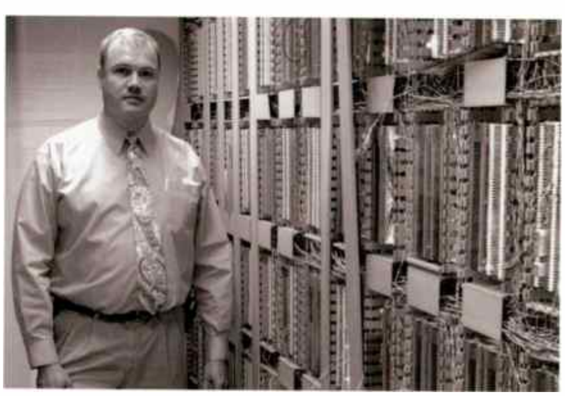
In the 10 years since Axia debuted to stunned broadcasters, Livewire IP-Audio has become the industry standard for AoIP broadcast audio infrastructure. But it wasn't always that way: every story has a beginning, and the pioneering work of Minnesota Public Radio (MPR), and in particular, Tom Nelson, Director of Engineering & Facilities Maintenance, American Public Media Group (APMG) – parent company of Minnesota Public Radio, Classical South Florida and Southern California Public Radio – parallels the growth of Axia Audio.

Minnesota Public Radio's Tom Nelson at one of MPR's many Axia Element-equipped studios

### BEGINNINGS: CONSOLES & ROUTERS

The story begins in 2004, when MPR and APMG were set to double the size of their facilities in the Minneapolis-St. Paul headquarters. The studios had originally been built with analog broadcast consoles. Early in the process, a group was formed to select a control surface for the first 20 studios. The major TDM console manufacturers made presentations, of course; then Axia personnel demonstrated a prototype Element console, and everything changed.

"Right away, the group noticed the tremendous cost savings in wiring, punch blocks and infrastructure when a TDM install was compared with an IP based facility," remembers Nelson. "The money allocated for studios could suddenly go a lot further."



One of the walls of wiring at MPR, replaced by IP-Audio

Neat high-density Ethernet and Axia DSP mixing engines power MPR studios

While networked mixing consoles are commonplace today, Nelson points out this was not the case in 2004. "Back then, analog systems were being phased out, and TDM consoles and routers were the standard replacement. It was a huge leap of faith to go forward with an AoIP system, but looking back, we made the right investment."

Labor costs were reduced significantly, Nelson recalls. "At the time, there were no systems integrators with Axia experience. But then we realized that we didn't need one. We just wrote out specs for running CAT-6 and fiber back to a Cisco switch, and had the work done by voice and data contractors who were already working on-site." He adds that for smaller installations, it is a simple matter for engineers to simply run their own CAT-6 cables. The hardest part of such a large installation was programming the Cisco switch properly.

MPR found that Livewire simplified installations in other ways as well. Connections for automation PCs, for example, could be done directly with AudioScience sound cards designed for use in radio broadcast automation systems; these use the Axia Livewire protocol to record and play audio streams over a standard Ethernet network. In a similar way, audio from remote workstations can be brought into the system via the Axia IP-Audio Driver, which allows PCs to stream audio directly to the network from their NICs and eliminating sound cards altogether.

### GROWING CONNECTIONS

MPR devised a plan to allow the gradual phase-out of legacy analog consoles by migrating audio sources into Axia audio nodes and using PathfinderPC routing software. "At the time of its completion, MPR had the largest install of Axia Element consoles anywhere," recalls Nelson.

Talent preps for air behind a dual-frame Element console at MPR

As Livewire has been formalized over the years, many partners have been added. More than 60 of the industry's top audio hardware and software vendors have partnered with Axia to integrate Livewire functionality into their newest offerings.

The MPR studios were renovated with Axia Element consoles, and their existing analog intercom system is following suit, being replaced by the new Axia IP Intercom. As with the Element control surfaces, MPR Broadcasting Engineering partnered with Axia R&D to help develop the features and functionality of the world's first broadcast intercom system with an Ethernet backbone. IP Intercom is a breakthrough technology not only because of its ease of installation and broadcast-quality audio, but also because it allows the program audio streams to be joined seamlessly with communications channels. "We found this to be particularly useful in linking IFB from the producer's position with remote guests on the phone, ISDN, or via IP codecs," notes Nelson.

Large consoles and clean, uncluttered sight lines typify MPR studios

### HEADING FOR THE COAST

89.3 KPCC

As MPR's Axia network continued to expand and grow, their Los Angeles affiliate, Southern California Public Radio (SCPR), was beginning to contemplate building new program origination studios for their Los Angeles affiliate, SCPR, and their local FM, KPCC.

KPCC FM started life as a college station licensed to Pasadena City College and since 2000 it has been operated and managed by SCPR. The transition involved a complete revamp of the station's programming, including the addition of a large local news department. KPCC quickly outgrew its cramped quarters on the lower level of the main library building, and made the decision to develop a new off-campus facility. SCPR Technical Operations Director Doug Johnson began thinking about the requirements for a new facility in 2008.

Recalls Johnson:"First, it needed to be large enough to accommodate the station staff, which had grown to about 80 full-time employees, as well as anticipating planned growth. The studio requirements included two large talk show studios with control rooms, a news host booth, production studios, editing booths and a master control studio for monitoring various tech systems and running the station during off hours. There was also a need for a large auditorium suitable for community and station events, as well as live or taped broadcast events."

New SCPR / KPCC facilities in Pasadena, California

Early in the search Johnson and his staff located a two-story building with 30,000 square feet near downtown Pasadena, and convenient to three major freeways — always an important consideration in Los Angeles — and the project planning began in earnest.

### CHOOSING AXIA - AGAIN

Of course, one of the biggest decisions facing Johnson and his team was the selection of control surfaces. As MPR's engineering staff did, Johnson's crew narrowed their choices to four and visited stations using those systems. Again, Axia was selected, based in part to the success of their MPR colleagues' $50 million construction project using Axia — and the fact that their know-how and expertise would plug right into the SCPR project.

As the project progressed, more decisions on equipment purchases were finalized. "We already owned and used a robust ENCO Systems play-to-air system and were satisfied with its operation and technical support," recalls Johnson. "We upgraded to their new single RU workstations and added several more workstations for the larger facility." ENCO is also an Axia Software partner, offering play-out systems that directly record from and playback to Axia networks using the Livewire IP-Audio Driver.

SCPR Master Control features a massive 28-position Element console

Architectural responsibilities for the project were divided. Design for the building itself was done by the architectural firm of Chu/Gooding in Los Angeles; studios were designed by the Russ Berger Design Group of Dallas. Pasadena has a strict architectural and historical review board requiring all new construction to adhere to the town's architectural heritage and image. Having a local firm made it much easier to interact with the city and guide the station through the permitting process. The next technical challenge was identifying and hiring a systems integrator to pull it all together. "There are several good integrators in the Los Angeles area, but few [at the time] had experience with the Axia installs," notes Johnson. "Since Axia is assembled more like a voice and data network, traditional integration approaches do not work."

In the end, Johnson chose to work with Tom Nelson and his technical staff from APMG given their deep experience with Axia products and recent completion of a similar project; they were also familiar with SCPR's personnel and organizational structure.

SCPR Production Studio B waits for its next producer to arrive

## ON THE AIR

One of the biggest operational challenges in any studio move is how to transition various signal paths to the new location without service interruption. "We settled on a temporary T1 connection between our new facility and the existing transmitter location," explains Johnson. "This allowed us to transition to the new building methodically. We began with early-evening test broadcasts from the new studios. Once we were confident we could broadcast from the new location, we began to increase broadcasts from the new building over a two-week period."

The next month, SCPR made the permanent change to the new facility. Over one weekend the Pasadena-based staff transitioned to the new location. A week later, the Los Angeles staff moved in.

"At first, we were nervous about being a trendsetter in Los Angeles," recalls Johnson. "But APMG did all the installation and backup. Axia has been rock solid for the past two years."

One of the edit bays for KPCC features a 12-position Element console

SCPR makes excellent use of a capability found in all Axia mixing consoles: the Show Profiles feature, which allows users to save console "snapshots" and then recall them later. Show Profiles can automatically load audio sources, reassign mic faders, select monitor feeds, even automatically apply voice EQ and dynamics on consoles so equipped. "We have one person whose job is on-site training," says Johnson, and new hires get one week of instruction in all facets of the organization's complex broadcast operations. The technical trainer also makes sure each of SCPR's 16 studios work Identically, with the same presets and Show Profiles. This position also acts as tech support for SCPR's four news bureaus. "This has proven to be a very successful strategy," adds Johnson.

Looking ahead, Johnson is anxious to upgrade the SCPR phone system. Currently Telos Series 2101 systems are in use, but migration to the new Telos VX Broadcast VoIP talkshow system is planned for the near future.

### AXIA IN THE FUTURE

The operations of APMG continue to expand and flourish, and Axia will be part of those operations. Tom Nelson is currently tackling more APMG Los Angeles facilities, which will include three new studio control room pairs, and five production rooms. All will be equipped with 12- and 24-fader Element consoles, and Telos Nx12 broadcast phone systems.


In the heart of MPR's TOC, all systems are "OK"

Axia is the only American broadcast manufacturer with a research center located in Riga, Latvia; a key factor that is not lost on Nelson, as Riga has long been a noted telecom and IT technology development hub in Eastern Europe. "It makes no sense for broadcasters to reinvent the work that Cisco is doing in R & D," says Nelson; "Axia has focused its research in the areas that it's good at, and has the maximum impact." The center also has more immediate benefits for customers. "There have been times when we needed some customized [routing] code for a project, and the solution has literally arrived overnight in the form of an e-mail attachment from the center in Latvia."

No broadcast plant is static these days, and Nelson is already planning for the expansion of APMG facilities and replacement of gear well into the future. "Looking ahead, I see construction for the next five or ten years. An AoIP infrastructure is so reliable and easy to install, I see us being involved with Axia projects for the next ten years."

**Tom Vernon**
Reporter-at-large
The Telos Alliance

Aero.2000 - System - System - Display Options
DRC - Program 1
Input
A
Multiband
Final Limiters
Output
DRC
Loudness 1
F
M
S
Program
Pgm 1
-24
ITU
Readout
Memory Slot
Fullscreen Mo
1: DRC - Progra
2: Loudness
3: Off
4: Off
5: Off
6: Off

## LINEAR ACOUSTIC

Looking back over the long history of television there have been countless landmark technical achievements and changes along the way, not the least of which has been the shifting role that sound has played in the viewer experience.

Not long ago, television audio was considered "good" by its mere presence. It was delivered into family rooms by a mono speaker inside a very large, very dense wood-grain cabinet. And there was no remote control. But, there was a human operator at the station manually adjusting audio so that everything sounded smooth and pleasing.

Today, high definition services deliver exceptional video and audio to our homes, possibly being reproduced by a magical combination of glass and light and 5.1 or 7.1 (or more?) speakers and the results bring tears of joy to our eyes. At most houses though, the audio is more than likely being forced out of a tiny, underpowered, poorly performing "speaker" in a way-too-thin cabinet. Same tears, not so much joy. Sadly, the operator at the station has long since retired.

Viewers expect dynamic, high-quality, multichannel audio that is consistent regardless of how or where it is consumed, despite the challenging and pesky details of enabling that to happen. The bar has been raised.

Linear Acoustic has grabbed that bar and is running with it. We are known in the industry for uncompromising and practical loudness management, upmixing, downmixing, encoding, decoding, metering and monitoring solutions. All of our products address the real needs of real broadcasters. However, these products and technologies are the result of a much larger passion.

We also spend countless hours researching new ideas, listening to the results, working closely with broadcasters and the creative community to test things in the real world – all to push the state of the art forward and help define the future of television audio. While others are content with being reactive and releasing "me too", we know that we must constantly innovate ways of bringing the most exciting and engaging audio experience possible to television viewers. Whether it is local news, live sports, a documentary, dramatic programming, or a full-length Hollywood movie watched in a home theater, on a large television, a small television, a portable device or all of these simultaneously – our goal is to make sure the home viewer, Hollywood, and your station are all satisfied.

Every now and then, it's great to pause for a second to look back and see how far we have come. Winning a technical Emmy in 2010 for our real time loudness and metadata management was exciting and it was okay to enjoy the moment. But taking on the responsibility for helping to create a better future often means heading back to the lab, heading out to the field, always looking forward and thoroughly enjoying a job we take very seriously.

**Christina Carroll**
Vice President and Executive Director
Linear Acoustic

# AERO.2000™

## AUDIO/LOUDNESS MANAGER

Loudness Control, Upmixing, AES and SDI I/O, Compensating Video Delay, plus Optional Dolby® Encoding and Decoding and Nielsen Watermarking.


AERO.2000 is audio purity for digital television. A very inclusive extension of the popular AERO.air™, it combines air-proven loudness control, audio processing, encoding/decoding, unmatched upmixing capabilities, and extensive I/O features.

Factory presets ensure consistent, reliable signal transmission, while access to individual controls provides fine-tuning for experienced users. Adjust AERO.2000 for wide-band multi-stage processing, multiband multi-stage processing, or anywhere in between.

AERO.2000 accepts 5.1-channel and two-channel audio via included AES or HD/SD-SDI inputs. Audio is then processed by ITU-compliant AEROMAX® loudness and dynamic range control, including patented Linear Acoustic Intelligent Dynamics™ hybrid metadata processing. The result is smooth audio with appropriate dynamics without affecting the original content.

Industry standard two-channel to 5.1-channel automatic upmixing is provided by the "Hollywood approved" UPMAX® II algorithm resulting in a consistent and captivating surround-field, perfectly downmix compatible for all stereo viewers.

A fully processed selectable LtRt surround or LoRo stereo downmix of the main program audio is provided at all times for legacy stereo distribution paths or simple local monitoring. A 6.3mm (1/4") high-level headphone connector and VGA output for multi-screen display complete the package.

Extensive standard I/O includes up to 16 channels of AES inputs and outputs, auto-sensing HD/SD-SDI I/O, with included compensating video delay, enables de-embedding and re-embedding up to 16 channels of audio plus SMPTE 2020 (VANC) metadata. Embedded channels can be routed through or around processing. Encoded signals can be de-embedded and re-embedded.

A powerful front panel headphone connector is provided for local monitoring, while a bright, colorful display, large rotary encoder, and four control keys provide simple menu navigation and adjustment, with remotely control via parallel TTL inputs and status outputs.

Gigabit Ethernet enables TCP control by automation systems and allows full adjustment and display of all parameters, including return audio for remote monitoring, through a TCP/IP Remote Application.

The Linear Acoustic AERO.2000 contains dual redundant power supplies and hard relay bypass of the digital audio, SDI and metadata signals for mission critical applications.

Options such as Dolby® encoding and decoding and Nielsen watermarking can be field-enabled via purchased software keys.

**PROCESSING STRUCTURE:** 16 total audio channels in two instances. **First instance:** 5.1+2 with stereo local/voiceover input, downmix output, and dual-path upmixing. **Second instance:** Dual stereo (2+2) with stereo local/voiceover input and auto SAP/DVS.

**OPTIONS:** » **NIELSEN OPTIONS:** Generates revenue critical NAES II and Nielsen Watermarks audience measurement codes. AERO.2000 precisely inserts these signals for maximum code recovery – after audio decoding and processing and before transmission encoding. » **DOLBY DECODING OPTIONS:** Allows reference quality decoding of Dolby Digital (AC-3) and/or Dolby E content plus metadata from any AES or SDI input signal. » **DOLBY ENCODING OPTION:** Dolby Digital (AC-3) and Dolby Digital Plus .

# AERO.1000™

## AUDIO/LOUDNESS PLATFORM

High Density Metadata-Based Loudness Control with Dolby® Coding, Processing, Upmixing, ITU-R BS.1770 Metering, TCP/IP Remote Control, and Redundant PSU

» Fully-Featured TCP/IP Remote Control Application
» HTTP and Web Server included


AERO.1000™ is a fresh, revolutionary approach to balancing 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.

» Up to 8 AEROMAX® audio engines including UPMAX® » Linear Acoustic Intelligent Dynamics™ hybrid metadata processing » Up to 8 Dolby decoders and 8 Dolby Encoders » Utility ITU-R BS.1770 loudness meters » 3GHz HD/SD-SDI I/O with compensating video delay » Up to 16 channels of AES I/O with reference input » Stereo +4dBu Analogue I/O » Front panel headphone output » TCP/ IP remote control and HTTP server » Redundant PSU » Up to 8 Nielsen watermark encoders » DVB-ASI I/O (optional)

### VIEWERS ARE LISTENING. METERS ARE METERING.

Now that the industry is focused on loudness, solutions are rampant. Unfortunately, sound quality is mostly forgotten in favor of meter satisfaction. Linear Acoustic has innovated and ardently supports the approach of getting loudness matched to target upstream using metering and/or file-based scaling tools. Leave the final polishing provided by dynamic range control (DRC) as a part of transmission.

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 - it 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.

Applied by default in all consumer decoders, metadata provides DRC that can be disabled in higher-end systems. How does it sound? Exactly like the high-quality audio control always provided by Linear Acoustic: Exceptional. The difference is that the audio can remain untouched. Or not. Broadcasters can choose permanent control where necessary, leaving reversible control for high quality trusted programming.

Handling up to 64 channels of audio, encoded or baseband, AES, SDI or DVB-ASI, AERO.1000 offers extremely high density. Performing control as a hybrid between single-ended and metadata processing, AERO.1000 preserves quality. Designed and assembled in the USA, the lightweight and rugged AERO.1000 is a solid investment in performance and lexibility. As new processes are discovered, AERO.1000 will be our go-to platform for delivering them to the industry.

A bright yellow OLED display and integrated rotary navigation cluster provide straightforward menu navigation and function adjustment. Failover bypass relays on all I/O maintains signal continuity. Dual auto-ranging power supplies for redundant worldwide compatibility.

# LINEAR ACOUSTIC® INTELLIGENT DYNAMICS™

## SATISFYING VIEWERS WITHOUT PERMANENTLY IMPACTING AUDIO QUALITY

As the world makes genuine progress towards aligning the average loudness of commercials, interstitials and programs, the dynamic range of programs also stands to benefit. This is a good thing for action adventure movies played in rooms with home theatre systems, but is challenging in a typical living room with the kids playing, the dishwasher running and maybe the baby trying to sleep in the next room. The problem is made worse by television sets with ever-shrinking speakers that strain to reproduce audio.

The junction or boundary between commercials, interstitials, and programs is also an issue. Everyone is familiar with a commercial that ends up feeling too loud because of the emotional (and quiet) scene that preceded it. Everything measures correctly, but it just sounds wrong!

### AUTOMATIC GAIN CONTROL

The solution to all of these issues is to manage the dynamic range, or the difference between the softest and loudest parts of an audio segment. This can be done with an Automatic Gain Control (AGC), and is nothing new. AGC is relatively simple to accomplish. A detector rectifies audio and via time constants produces a control signal that is used to vary the gain of the audio signal.

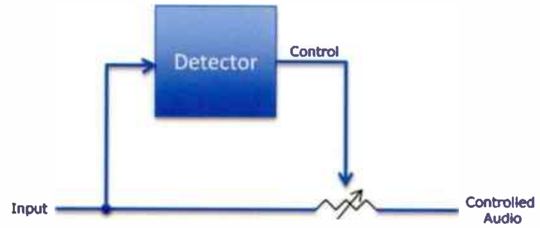



Figure 1 – Simple Automatic Gain Control (AGC)

This is the basic function of most current audio processors: increase the gain for quiet audio and reduce the gain for the loud parts. Unfortunately, while this will fix the boundary issues it will also take the life out of the intentionally dynamic action scenes. Everything gets a little something whether it is needed or not. It is permanent. And everyone gets the same thing.

Separating the detection and application of gain control is not a new idea, but until the advent of audio coding, it was not practical. The data rate of the audio is high (sample rate x resolution x channels), about 1Mbps per channel. The data rate of even a simple 8-bit wideband control signal is also high (sample rate x resolution), likely 384kbps. Sophisticated multiband control could end up taking more data than the audio signal itself!

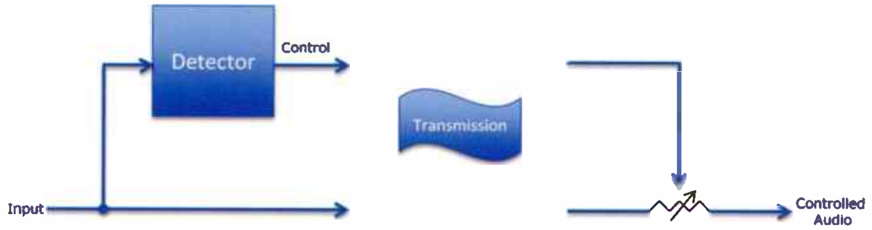



Figure 2 – Automatic Gain Control (AGC) with detector and gain control split across transmission, impractical due to the data rate of the audio and control signals.

### METADATA DRC

The most popular multichannel audio transmission formats, Dolby Digital (AC-3) and Dolby Digital Plus, utilize this technique of separating audio and control data, or Metadata and one part of this metadata is dynamic range control or DRC – essentially an AGC split across transmission. However, in this case the data rate of the audio signal and the control data is reduced to make the system practical.

Audio coding, or bit rate reduction, uses time and frequency domain techniques to represent audio above the threshold of human hearing. In the frequency domain, audio is organized into blocks and so control signals must also be organized into blocks. This results in control signals that require less data but are also a somewhat coarser representation of gain control.

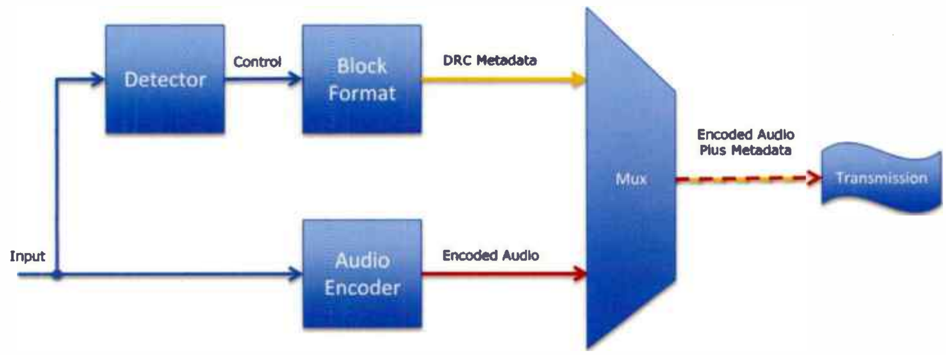



Figure 3 – AGC implemented as part of low bitrate audio encoding to control dynamic range (DRC).

Decoders reconstruct the audio and use the Dynamic Range Control (DRC) portion of the data to modify the dynamic range of the decoded audio. Thanks to windowing, the effects of the control signals are smoothed, resulting in the potential for accurate control that sounds good. Importantly, the audio remains intact and by merely scaling or defeating the application of the control data, the original dynamic range can be restored as can be seen below.

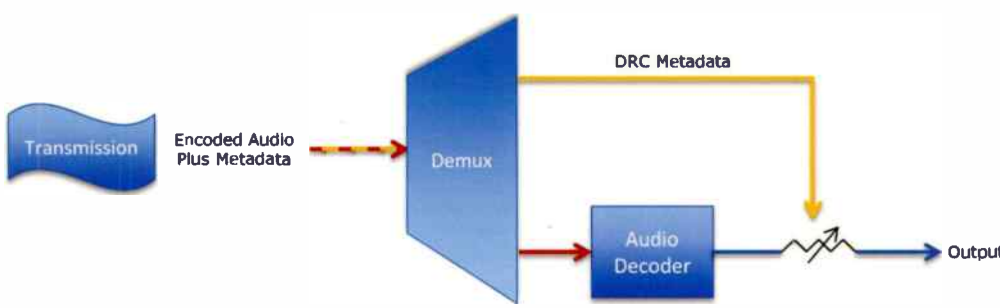



Figure 4 – Dynamic Range Control metadata is applied to the decoded audio by default, but can be scaled or bypassed.

Metadata DRC has traditionally been generated by the Dolby Digital encoder using one the built-in profiles such as Film Standard or Music Light. The problem is that these are one-size fits all and tend to be more appropriate for DVD mastering than broadcast audio processing. The effects can be acceptable if other metadata parameters are correct, but can quickly result in too little control if everything is not perfect.

**THE FIRST PART**

To solve this part of the problem, Linear Acoustic has developed a patented technology which generates new metadata dynamic range control via traditional processing techniques, resulting in a hybrid dynamic range control that can be permanent, reversible, or anywhere in between. It does not rely on any upstream or local metadata in order to function properly, thus like a permanent single-ended processor, it will always produce predictable results.

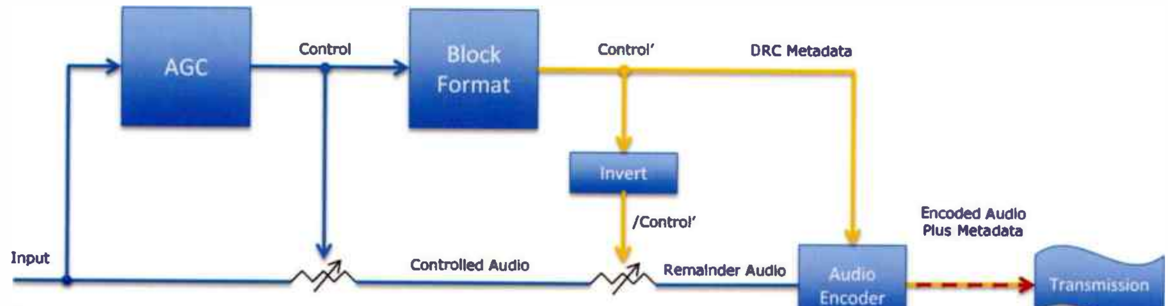



Figure 5 – Linear Acoustic hybrid metadata DRC generation

Importantly, for the typical consumer, there is no audible difference between advanced dynamic range control being generated and permanently applied by a broadcast processor prior to encoding, or a hybrid system where the same controls are generated but are passed within existing metadata DRC and applied by the very large universe of installed decoders.

This is only part of the puzzle though. Fully reversible (i.e. 100% metadata) control is only appropriate for programs that obey the rules regarding loudness. After all, this is dynamic range control. Certainly, automation could be used to trigger presets that would apply processing with varying degrees of permanence. Dayparting, or schedules in the processor itself could do much the same thing. However, these systems are ripe for mistakes or abuse.

### THE OTHER PART

A new technology from Dolby measures loudness and quantifies other important details about an audio signal. Called Dolby Evolution data (EVO for short), it is signed to enable fast authentication and is carried along with the audio via several methods, including as a metadata extension and also within multichannel PCM audio.

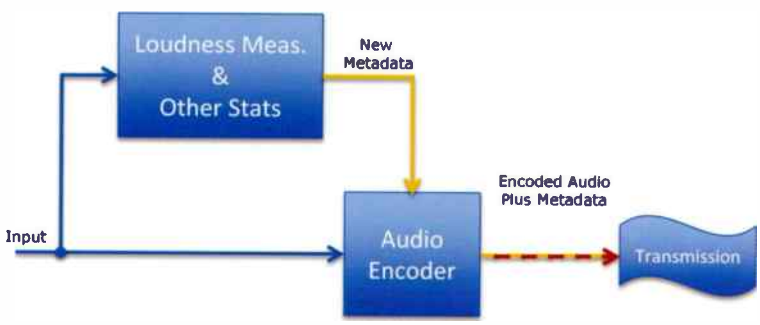



Figure 6 – Dolby loudness data insertion into encoded bitstream.

Part of a larger systems approach called Dolby Intelligent Loudness™, each stage of the chain from production onwards benefits from the presence of EVO data. Not only is it useful for instantly displaying previously measured loudness, increasing accuracy and removing the complexity of downstream metering and spot checking, it can also be used to automatically control processing.

LINEAR ACOUSTIC | TECHNOLOGY

**INTELLIGENT DYNAMICS**

Combining EVO data with a dynamic range control system that can vary the percentage of permanent processing results in a system called Linear Acoustic Intelligent Dynamics™ which finally enables programming to be processed to the degree and permanence dictated by the programming itself. Content can automatically control the processing and mitigate the need for further downstream changes that could irreparably change it.

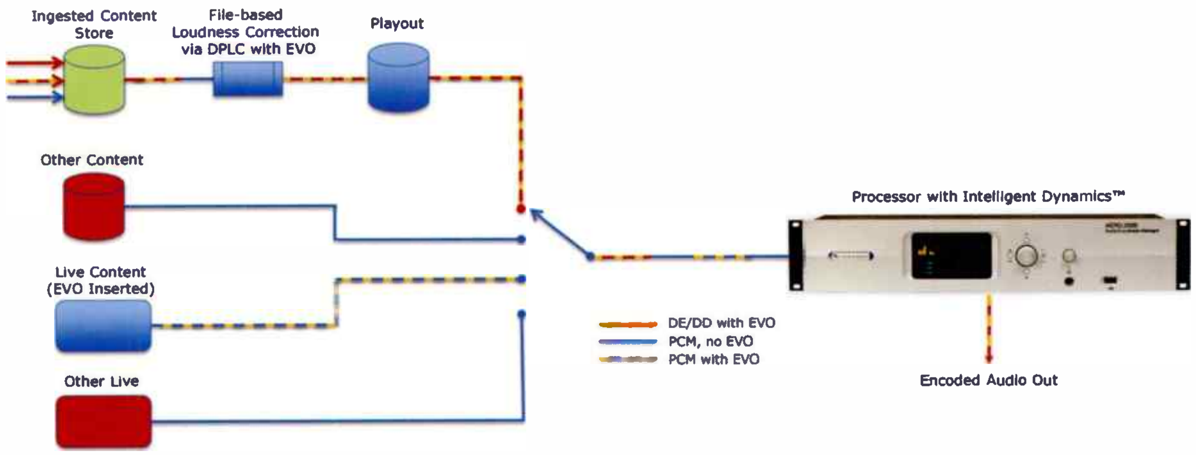



Figure 7 – Signal flow with showing Dolby EVO data accompanying audio through a simplified broadcast chain. This data can be used to display loudness and other statistics and a processor like the Linear Acoustic AERO.1000 and AERO.2000 can vary the reversibility of the DRC applied to the audio.

Using the same Dolby Digital and Dolby Digital Plus decoders which contain standardized and tested metadata implementations, consumers can enjoy the benefits of audio that can be tailored to their liking. From the default of controlled dynamic range for noisy environments to full dynamic range for experiencing the full impact of well-produced programming, Dolby Intelligent Loudness and Linear Acoustic Intelligent Dynamics deliver the best of both worlds.

**Tim Carroll**
CTO
The Telos Alliance

# AERO.one
## AUDIO/LOUDNESS MANAGER

Audio/Loudness Manager for Stereo, 5.1 and 16 Channel Audio HD/SD-SDI I/O and optional Dolby® Digital (AC-3)/Dolby Digital Plus Encoding


AERO.one is proven loudness control and upmixing for digital television.

Built in loudness control, metadata control, and optional transmission encoding make AERO.one the ideal choice for stations that want to provide a seamless, optimum quality experience for their audience. Now your viewers can be protected from loudness shifts and loss of surround sound in a simple, cost effective, compact, and feature rich manner suitable for main or backup transmission paths.

Like other processors in the Linear Acoustic AERO family, the AERO.one accepts up to eight pairs of PCM audio (4 pairs via AES, up to eight pairs via SDI) to handle 5.1 + 2 channel audio programming. The unit can apply adaptive wideband and multiband, multistage ITU compliant loudness control and upmixing to the audio, with or without metadata guidance, to tame loudness and image shifts while preserving more of the original content than previously possible.

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.

Upmixing and processing modes can be controlled by a combination of GPI contact closures and applied metadata.

Downmixed versions of the main programs are available at all times. These signals can be either a stereo LoRo downmix or an industry standard LtRt surround encoded mix.

Metadata can be applied, if available, via the VANC space of an applied HD-SDI signal or from a standard serial input to control of upmixing and processing functions. Extensive protection is provided to prevent audible effects of incorrect or missing metadata.

A highly-visible LED display and simple navigation cluster provide easy function adjustment. Relay bypass of all signals for trouble-free operation in transmission critical environments.

Available options include internal 5.1 channel Dolby Digital (AC-3) and Dolby Digital Plus (E-AC-3) Encoding and SNMP monitoring.

**AERO.ONE (V3)** – 5.1+2 channel loudness control plus dual UPMAX upmixing engines and downmixed output.

# AERO.lite

## AUDIO/LOUDNESS MANAGER


Cost-effective 2-channel CALM and R128 loudness compliance - that's what AERO.lite™ delivers. Solve inconsistent audio loudness with a simple set-and-forget, feature-rich stereo processor that incorporates award-winning loudness control tools and extensive I/O. Perfect for main or backup transmission paths.

Input and output signal levels are displayed alongside processing meters, and the optional ITU-R BS.1770 measured LKFS output loudness value is provided to give instant verification of loudness compliance.

Audio can be extracted from any pair of an applied HD/SD-SDI signal, AES, or balanced analogue inputs and routed to the processing core. Output is provided simultaneously via the front panel 6.3mm (1/4") headphone connector, +4dBu balanced analogue outputs, AES output and for re-embedding into any or all SDI pairs. Analogue or AES inputs can be used as sources for embedding even if not used for processing. Since all 16 channels are available for de-embedding and re-embedding, pair shuffling can be easily accomplished.

Designed and assembled in the USA, the lightweight and rugged 1RU aluminum chassis is durable enough for installation in challenging environments like OB trucks and cramped edit bays. Built on a broadcast quality Linear Acoustic platform, the AERO.lite is professional grade equipment.

A bright yellow OLED display and integrated rotary navigation cluster provide straightforward menu navigation and function adjustment. Failover bypass relays on all I/O maintains signal continuity. Auto-ranging power supply for worldwide compatibility, and sealed, locking 2.5mm DC input connector for available redundant power supply.

# HEARST TELEVISION CHOOSES LINEAR ACOUSTIC®

## 29 STATIONS TO RELY UPON AERO.air™, AERO.2000™ AND LQ-1000™ FOR CALM COMPLIANCE


LQ-1000™ Loudness Quality Monitor (see page 120)


AERO.2000™ Audio/Loudness Manager. Built upon the highly successful AERO.air™ (see page 108)

On December 13th, 2012, the Federal Communications Commission began to enforce the rules outlined in the much-talked-about (and oft misunderstood) CALM Act. This legislation-turned-law was created to address the so-called "loud commercial" problem that startles families off their sofas when blaring ads contrast with quieter programming, sending popcorn – and complaints to television stations – flying.

We talked with Joe Addalia, Director of Technology Projects for Hearst Television, in the fall of 2012 as he prepared to make sure all of his company's stations were providing CALM-compliant audio to those viewers.

You might expect someone whose company owns twenty-nine TV stations in the US, reaches around 18% of the country's households, and programs sixty-two discrete channels to be a little nervous, but to the contrary, he's pretty calm himself.

That mood stems from the choice of the Linear Acoustic® AERO.air™ Transmission Audio Loudness Manager and LQ-1000™ Loudness Quality Monitor as the standard equipment pairing to control and meter loudness at all Hearst stations.

"CALM isn't going away," Addalia pointed out. "We knew we had to address compliance head on. Hearst doesn't take short-cuts, we don't compromise on quality, and we don't look for band-aid solutions just to get by. We wanted to do this right, so I researched all of our options and kept coming back to Linear Acoustic."

It was during that research that Addalia discovered that some of the other projects he was working on – including finding standardized solutions for audience measurement encoding, upmixing, downmixing, and better quality audio processing – could also be accomplished with a single product, which turned out to be the AERO.air.

"On the surface, it's not an inexpensive product," admits Addalia. "But when you look at all that it does, suddenly it's a bargain. We get full control over our audio, which sounds fantastic, a solid downmixed signal, Nielsen encoding located where it's supposed to be in the chain, and CALM-compliant loudness control in a single 2RU box." That type of consolidation is important in Hearst's view. "Stations get into trouble when we make things too complex. AERO.air addresses so much with one box and really simplifies things."

How much does Hearst like AERO.air? "We're putting them on the main HD channel of all of our stations. Once that project is done, our goal is to have identical, fully redundant backup transmission chains for every station, and each of those will all have an AERO.air as well."

In keeping with the company's no-compromise approach, Hearst has also installed a Linear Acoustic LQ-1000 Loudness Quality Monitor in each station. "We use the VGA output of the LQ-1000 to feed a section of the monitor wall in the master control room so the operator can always keep an eye

MASTER CONTROL OPERATIONS AT HEARST AFFILIATE WESH IN ORLANDO, FLORIDA


on loudness," said Richard Monn, Chief Engineer of Hearst's Orlando operation. One of the meter's inputs looks at the primary encoded HD signal full time, while the other is used on a rotating basis for the required spot-checking as well as multicast channels or audio playing back from a server.

Plant-to-plant consistency is very important to Addalia. "If you walk into a Hearst station in Jackson, Mississippi, you'll see the same gear you see when you walk into our station in Boston. Once we find the products that best meet our needs, we like to form a close relationship with the company that makes them and deploy those products across the board. For loudness control and metering, it's Linear Acoustic all the way."

For 2013, the Linear Acoustic AERO.2000™ has become the successor to the popular, venerable AERO.air, offering the same proven performance in an even more flexible platform.

More information about these and other Linear Acoustic products can be found at www.linearacoustic.com

# AERO.mobile™

## AUDIO/LOUDNESS MANAGER


AERO.mobile™ is rich audio clarity for Mobile DTV. Part of convincing an audience that Mobile DTV is an exciting option means providing them with an enjoyable experience. Viewers are listening. AERO.mobile is designed to ensure viewers not only hear and understand content, but are surprised by the clarity.

Mobile DTV must overcome physical constraints – small speakers and environmental issues such as background noise. In addition, program audio can range from mono to 5.1 channels and from faint to screaming loud. These factors combine to impair intelligibility, make viewing tedious and cause viewers to give up.

Traditional audio processing alone cannot enable diverse audio content to be reproduced effectively from mobile and handheld devices. In fact, it can make the situation worse. Simple wideband techniques don't do enough, and multiband systems soften important audio cues if overused – both negatively impact intelligibility.

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 dialogue 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.

Importantly, Mobilizer technology has been carefully designed and tuned to support and enhance systems like Dolby® Mobile which are being introduced for use within mobile receiving devices.

The rugged 1RU AERO.mobile is intended to be installed directly before the mobile audio encoder in either the AES or SDI paths. Bypass relays are provided to ensure continuous service in the unlikely event of failure.

A bright LED display, rotary encoder, and four control keys provide straightforward menu navigation and function adjustment. Dual, redundant, auto-ranging power supplies are available to allow for trouble-free operation worldwide.

# AERO.file™
## AUDIO/LOUDNESS MANAGER

Linear Acoustic® Loudness Control in a file-based domain.


Developed in partnership with Radiant Grid Technologies and available through Wohler Technologies as WohlerLoudness, AERO.file brings proven Linear Acoustic audio technologies such as multipass scaling, AEROMAX® loudness control, and UPMAX® umpmixing and downmixing to the file-based domain.

Designed to be used at ingest, for managing existing libraries, or for conforming content for different playout services, AERO.file allows loudness control and upmixing to be handled where it is often most effective: In the file domain.

Whether anchor-based such as with Dolby® Dialogue Intelligence™ or based upon the overall average with the relative gating methods of EBU R128, AERO.file extracts the audio from a host of popular file wrappers, provides proper scaling, aligns with the anchor or overall average loudness of the content, and re-wraps the audio without disturbing other video or data essences.

AERO.file supports WAV, AIFF, MPEG 1 Layer II, MP3, AAC, ACELP, WMA, AMR and uses SurCode for Dolby Digital, Dolby Digital Plus and Dolby E encoding and decoding.

AERO.file is available from Wohler Technologies as WohlerLoudness and as an option on WohlerCoder, WohlerConverter 3GPU, and WohlerConverter 6GPU products.

# LQ-1000™

## LOUDNESS QUALITY MONITOR

LQ-1000™ gives vibrant clarity to loudness quality metering. Supporting the latest ITU-R loudness measurement standards, LQ-1000 can also include Dolby® Dialogue Intelligence™.

The LQ-1000 difference is in the display. A colorful long-life OLED groups critical loudness parameters like short, medium and long term loudness, loudness history, current peak level, maximum peak level, and the loudness target.

Color is employed 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. The large LKFS loudness number also changes color to better indicate if the number matches the chosen target.

The LQ-1000 includes two sets of meters to simultaneously measure a 5.1-channel program and a 2.0-channel program. The second meter can alternatively display an internally created LoRo or LtRt downmix. The meters can also respond to metadata applied as serial data or from the VANC space of an applied HDSDI signal, showing the effects of dialnorm and coding mode. True peak metering is also provided.

Loudness history is an essential part of useful loudness measurement, especially for long form programming. The LQ-1000 loudness histogram allows loudness trends to be easily seen, and immediately highlights problem sections.

The LQ-1000 now provides logging to a network drive, which allows stations to keep a record of their loudness measurements should they need to examine the data for a particular date and time. Four internal metering logs are generated - 24 hour, 48 hour, 7.5 day, and user controlled – and include readings every second from A, B, and C meters, plus dialnorm metadata or target loudness and acmod metadata. All information is available in the ubiquitous .csv format and accessible via storage to a local USB "thumb drive." The new built-in HTTP server also provides manual access to the logs and the ability to use http commands to automate log retrieval.

Dolby Dialogue Intelligence is available and provides the most accurate estimate of loudness possible in an automatic meter. Pausing integration during non-dialogue sections and reverting to BS.1770-2 over time, loudness can finally be measured with accuracy independent of program dynamic range.

Common functions such as measurement Start, Stop, and Reset are controlled by dedicated front panel buttons - no need to dig through menus. A powerful, high quality 6.3mm (1/4") headphone output is provided.

A VGA output to feed an external monitor is standard. Options include Dolby Digital (AC-3), Dolby Digital Plus (E-AC-3), and Dolby E decoding, and a 7" remote VGA display.

# LQ-1™

## ITU-R BS.1770 LOUDNESS METER

SDI, AES, Analogue Inputs Standard, Dolby® Decoding


LQ-1™ is perfect streamlined metering. When only one single number is all that is necessary, that number must be correct. LQ-1 manages complex I/O, Dolby decoding, and metadata to provide metering accuracy in a compact, cost-effective package.

» ITU-R BS.1770 Compliant Metering » Simple display of signals and loudness » HD/SD-SDI, full 16-channel de-embed » Discrete AES inputs, downmix AES output » Dolby E/D/DD+ decoding » Dolby Dialogue Intelligence™ » Stereo analogue input » Headphone and +4dBu balanced monitor out » Selectable LtRt or LoRo downmix » GPI/O Alarms and Control » DVB-ASI Input » Ethernet for Logging and SNMP (option) » Dual PSU (option) » Fully upgradable future-ready platform

LQ-1 provides all necessary I/O, routing, decoding and metadata tools.

Input signal levels are displayed alongside the loudness or dialnorm target and measured loudness is continuously displayed. Meter running status (start/stop) is shown as well.

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.

A downmix of the input signal is available as stereo LoRo or surround LtRt, and is provided simultaneously via the front panel 1/4" headphone output, +4dBu balanced analog output and an AES output.

Included Dolby Dialogue Intelligence provides the most accurate estimate of loudness possible in an automatic meter. Pausing integration during non-dialogue sections and reverting to BS.1770-2 over time, loudness can finally be measured with accuracy independent of program dynamic range.

Designed and assembled in the USA, the lightweight and rugged 1RU aluminum chassis is durable enough for installation in challenging environments like OB trucks and cramped edit bays.

A bright yellow OLED display and integrated rotary navigation cluster provide straightforward menu navigation and function adjustment. A medical-grade auto-ranging power supply provides trouble-free operation worldwide.

# UPMAX®

## 5.1 CHANNEL UPMIXER


UPMAX® delivers smooth transitions. Listening viewers are aware of programming changes, especially when the image shifts due to cases where stereo programs can only be reproduced from the Left and Right channels of a 5.1 channel program. This is commonly found in situations where it is not possible to switch the Dolby® Digital (AC-3) encoder. UPMAX is the simple, well-proven, cost-effective solution.

Based on the original UPMAX 2251, the 1RU UPMAX offers the most stable and trusted algorithm in use today for both production and unattended upmixing. Output is completely downmix compatible and the downmixed result is nearly indistinguishable from the original two channel input.

The resulting "Surroundfield" can be infinitely adjusted via the Center Channel Width control and the Surround Depth control. This allows programming ranging from simple stereo to LtRt to be appropriately reproduced from a 5.1-channel playback system.

An optional bass enhancement signal for the LFE channel is derived from the Left, Center, and Right channels allowing quick creation of a subwoofer channel without compromising the downmix. Factory presets are included for typical applications such as music and commercials. Further adjustment is simple and new results can be stored as user-defined presets.

UPMAX includes a utility encoder which accepts 5.1 channels and produces a two channel LoRo or LtRt output. This encoder can be independent or it can be fed by the same channels applied to the upmixer.

UPMAX is rugged and perfect for remote OB trucks, post production facilities, network operation centers, local station production, virtually anywhere upmixing is used.

Upmixing can be controlled via the front panel, GPI inputs, or metadata from serial or VANC (SDI) sources applied to the unit. Smooth bypass of 5.1 channel signals is accomplished automatically via the AutoMAX-II algorithm.

A bright LED display, rotary encoder, and four control keys provide straightforward menu navigation and function adjustment. Dual, redundant, auto-ranging power supplies are standard to allow for world-wide operation. Bypass relays are provided for trouble-free operation in transmission critical environments.

# LA-5269

## DOLBY® DIGITAL / PLUS TRANSCODER

Encode, decode, and transcode the most popular multi-channel audio formats used in television broadcasting in one feature-rich, modular package:

**Right out of the box, the LA-5269 can:** Encode to Dolby Digital (AC-3) from PCM, Encode to Dolby Digital Plus (E-AC-3) from PCM or transcode from Dolby Digital (AC-3), Encode to Dolby Pulse from PCM or transcode from Dolby E.

**Optionally, it can:** Decode Dolby E and transcode to Dolby Digital (AC-3), Dolby Digital Plus (E-AC-3), or Dolby Pulse. Encode to Dolby Pulse (AAC and HE AAC V1 and V2).

All coding is provided by a Dolby-manufactured encoder module featuring the latest versions of each codec for superior sound quality.

Because features and codecs can be updated at any time via software, broadcasters pay for only what they need at the time, knowing that they can always update the unit as their needs change.

Metadata is supported via a serial RS-485 connection and can be extracted from the VANC space of an applied HD-SDI signal per SMPTE 2020. Metadata input is frame synchronized and error-corrected to prevent audible disturbances to the encoded bitstreams. External transcoder input is also frame synchronized, allowing Dolby Digital splicing and smooth transitions without the need for external AC-3 frame synchronizers.

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.

# LA-5280

## PROFESSIONAL AUDIO CODEC


The LA-5280 Professional Audio Codec is the next generation of the ubiquitous Dolby® "DP" series of stand-alone products. Reference encoders and decoders matched with metadata and monitoring tools plus ITU loudness metering with Dolby Dialogue Intelligence™ result in a feature-rich, field expandable coding platform.

Encoding and decoding are provided via reference Dolby implementations for highest audio quality. All encoding has matching confidence decoding to ensure proper codec setup and performance. Decoders can also simultaneously monitor external signal inputs such as from a backhaul or off-air IRD. Included Dolby Dialogue Intelligence™ can be selected to provide the most accurate estimate of loudness possible in an automatic meter. Pausing integration during non-dialogue sections and reverting to BS.1770-3 over time, loudness can be measured with accuracy independent of program dynamic range ensuring that the dialnorm metadata parameter can be appropriately set in the encoders.

Audio can be extracted from any pair of an applied HD/SD-SDI signal, AES inputs, and optionally from a selected PID in a DVB-ASI signal. Decoded outputs are provided simultaneously via the AES outputs and for re-embedding into any or all SDI pairs.

Individual channel pairs or a downmixed version of any program are also provided via the front panel 6.3mm (1/4") headphone connector and +4dBU balanced analogue outputs. Metadata emulation is selectable for all outputs.

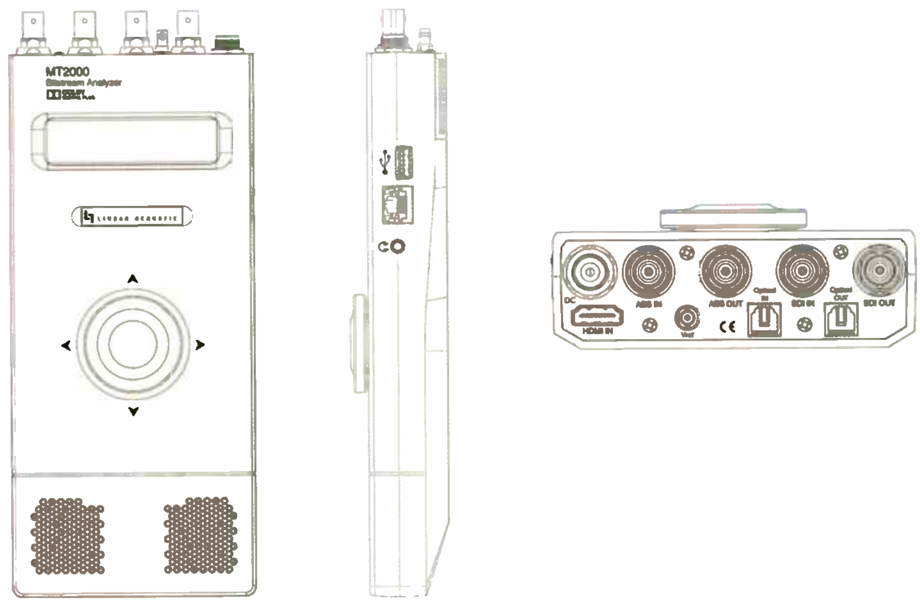
SNMP can be optionally added via a software key to enable remote monitoring of codec performance as well as loudness statistics. Optional factory-installed 8-channel balanced +4dBu analogue output with remote volume control input provides multichannel monitoring functions.

Designed and assembled in the USA, the lightweight and rugged 1RU LA-5280 is a solid investment in performance and flexibility. Fully field-upgradable, all options can be enabled by entering a factory provided key or module.

A bright yellow OLED display and integrated rotary navigation cluster provide straightforward menu navigation and function adjustment. Failover bypass relays on all I/O maintain signal continuity, and dual auto-ranging power supplies enable redundancy and worldwide compatibility.

**OPTIONS:** » Infrastructure version adds Dolby E encoding » Transmission version adds Dolby Digital (AC-3)/Plus and HE AAC (v1 and v2) encoding » Eight-channel +4dBu analogue monitor outputs » SNMP monitoring » DVB-ASI I/O supporting up to four PIDs

# MT2000
## BITSTREAM ANALYZER



The Linear Acoustic MT2000 Bitstream Analyzer is an efficient tool for end-to-end system testing and is the ideal next generation replacement for the original Dolby® DM100.

The MT2000 accepts signals via 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 loudness measurement with Dolby Dialogue Intelligence™.

Logging is provided via the USB connector and/or via SNMP over Ethernet. An extensive set of useful Dolby Digital, Dolby Digital Plus, Dolby E, AAC and HE AAC test bitstreams, including multichannel voice ID, are stored internally, and users can modify the set in the field via software download. The MT2000 can generate the selected bitstream 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.

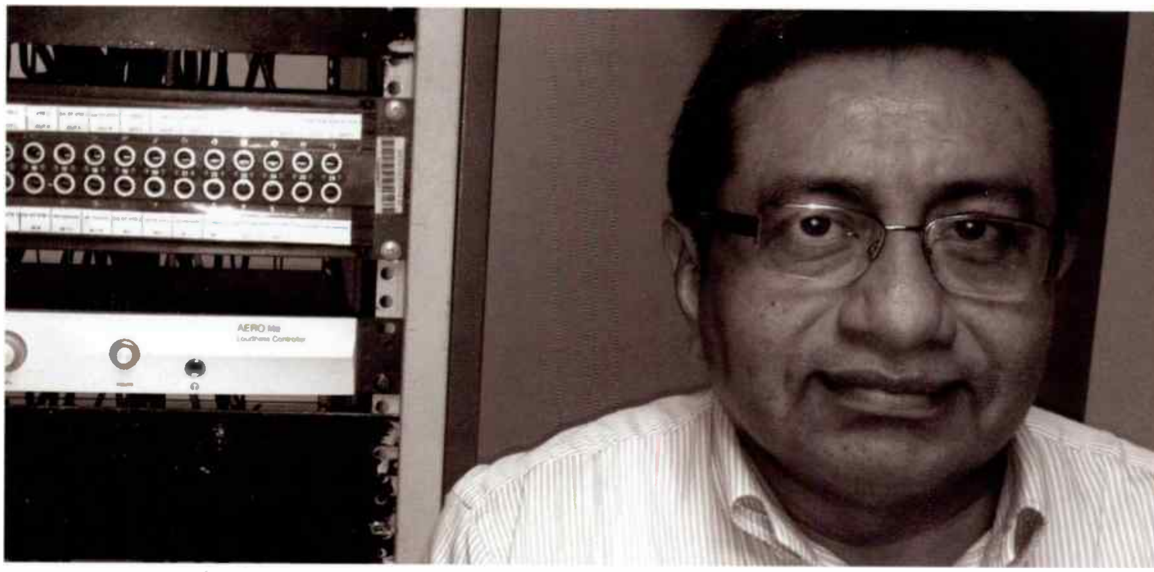
Signals are provided simultaneously via the AES and TOSLINK outputs and can be re-embedded into any or all SDI pairs. Output can be the original input signal, a multichannel PCM decoded version of the input signal, test signals, or, in the case of the SDI output, a combination of all of these. Inputs can be used as sources for embedding even if not used for decoding thus channel shuffling can be easily accomplished.

A bright yellow OLED display and integrated rotary navigation cluster provide straightforward menu navigation and function adjustment. A standard 1/8-inch stereo headphone jack can be switched to monitor any two decoded channels or a downmix of the whole program, while built-in speakers provide a stereo output, useful also for checking mobile audio compatibility. The MT2000 is powered by a field-replaceable NiMH rechargeable battery pack or from its DC charging port via an included universal power supply. The MT2000 ships with a carrying case and adapters.

**OPTIONS:** » SNMP Monitoring » DVB-ASI input » Live Dolby E encoding » HDMI Input.

# ECUAVISA LEADS THE WAY IN ECUADOR

## AUDIO QUALITY AND LOUDNESS CONTROL WITH LINEAR ACOUSTIC®




While regulations regarding television loudness have been implemented in many parts of the world, including the U.S. with 2012's CALM Act, the vast majority of television broadcasters in Latin American countries have not yet been affected by such legislation.

Ecuador's Ecuavisa network was no exception, until it occurred to Engineer Felipe Paucar that because Ecuavisa's programming was also being delivered to U.S. viewers, it needed to be CALM-compliant.

Following an on-site demonstration, Paucar placed the Linear Acoustic AERO.lite™ Audio/Loudness Manager in the transmission path feeding the satellite and cable providers.

"I noticed right away that the fidelity was much higher than it had been before, and we sounded much better than the rest of the stations," said Paucar. "Then we started hearing back from the cable and satellite companies that our loudness levels were perfectly controlled and consistent. That was not the case before AERO.lite. We opted for the built-in loudness meter, and that really made matching the output levels easy". Programming is also distributed to Peru through Telefonica with similar success.

Even though stations in Ecuador don't need to worry about compliance on terrestrial signals at the moment, Paucar is working with Linear Acoustic on choosing the right loudness manager for over-the-air use. "The improvement in audio quality alone is reason enough to put one on the air now. If the day comes when we have to be compliant, we'll be ready."

Linear Acoustic Vice President and Executive Director Christina Carroll says that's the reason audio quality remains the top priority for the company's products. "We've always said that keeping viewers happy is really what compliance is all about. If the audio is pleasing, people don't complain, and compliance is achieved."

For now, Ecuavisa delivers its content with 2-channel stereo audio, but they are about to begin a major technical upgrade and enter the world of high-definition video and multi-channel audio will be part of that project. Says Paucar, "Linear Acoustic made such a difference to our stereo audio that I can't wait to hear what they do with 5.1."

# WHEN BROADCASTERS SPEAK, WE LISTEN

As we were wrapping up the thousands and thousands of pieces of gear that we used during our coverage of the Games in London, one of the things that stood out in my mind was the use of the Linear Acoustic AERO.qc-II. We had one at every venue audio control room and several throughout the Broadcast Center, and they really simplified what we had to do to bring recording done in 3.0, or legacy material originally done in stereo, into a 5.1 presentation. The transitions were automatic, seamless and difficult to discern. In addition, the AERO.qc-II offered a great loudness meter, especially when using the VGA output. Support from the company was well beyond what anyone could have reasonably expected.

**Bob Dixon**
NBC Universal

Having great 5.1 surround sound accompany HD pictures is a necessity with an event of this magnitude. The Linear Acoustic e-squared system was one of the multichannel audio paths used for distributing the programs for broadcast. This is one of the most watched broadcasts in the world, with entertainment moments that are preserved for history, so audio quality and reliability were critical for us.

**Tad Scripter**
Engineer in Charge for the 81st Academy Awards

We are very happy with the AERO.lite units. Everything sounds great, and is in compliance too! These units sound very transparent. Really nice gear.

**Joey Gill**
Chief Engineer
WPSD-TV

KMOV has been using the AERO.air (5.1) for two months and we are very happy with the results. The quality of our audio signal improved noticeably when we placed the unit on the air. The 5.1 channel synthesizing and audio leveling is substantially better than with our previous device. The internal audio/video frame synchronizer function completely cleans up the signal and has corrected a problem with incompatible audio frames on switches. Linear Acoustic did everything possible to ensure that the installation and configuration was correct for our particular needs. I could not be more pleased with the company or the product.

**Walt Nichol**
Director of Technology, Broadcast Media
KMOV-TV St. Louis

Modern digital broadcast audio such as 5.1 surround sound and its metadata have made QC monitoring extremely important to our operations. In our move to a digital environment, we needed an advanced solution that would appropriately adjust metering and playback audio levels throughout the entire broadcast chain. We chose the Linear Acoustic LAMBDA based on an expectation of excellent audio quality, and that is exactly what we see. We've had the units in place for several months, and they have provided exceptional performance across all three of our channels.

**Gene Talley**
Director of Engineering/Operations
WPBT-TV

Like many broadcasters, we experienced a lot of problems with varying audio levels for network and local programming, particularly during playout of news and sports. We needed a way to address this discrepancy in loudness levels and even out audio volume, and the Linear Acoustic AERO.air has proven to be a wonderful solution. We noticed a significant difference immediately upon implementing the processor, and we haven't received any comments about disparities in audio loudness since.

**Brent Robinson, Chief Engineer**
KSL 5-TV

After switching our Comcast channel delivery from analog to digital, we discovered that our audio levels were out of control and we had cracking and popping that we could not resolve, causing viewer complaints every day. We called Linear Acoustic, and they offered to locate and fix the problem for us, leave the equipment in for us to try, and for a very affordable price - a no brainer. Wow! No more complaints from the viewers or the boss - just perfect audio at all times, and in full-time 5.1.

**Jan Strock**
Director of Engineering
WHTM-TV

Our AERO.air was installed and placed into service this past April. The unit integrated seamlessly with our existing equipment, and I've certainly been impressed with the overall quality of the 5.1 surround sound it provides. If that weren't enough, viewer concerns over commercial loudness have been virtually eliminated and we are now prepared as the CALM Act passed into law.

**Moreau Dugas**
Engineering Operation Manager
WSVN-TV

Our viewer complaints concerning audio immediately went to ZERO, and we sound great. Not much more to say besides today digital stations just plain need one.

**Brady Dreasler**
QNI

AERO.max 5.1 is IMPRESSIVE and although I am far passed being able to be impressed, I am with this gadget. The fact that we were able to easily insert our EAS as well was icing on the cake. We put it online with an external Dolby 569 encoder and last night I watched at home with my wife. Wow! The 5.1 is at the output all the time, simulated when local stereo is used, and passed as 5.1 from the network when available. The leveling makes transition between local and network material seamless, and I do not hear (or see) my Onkyo receiver switching surround modes during the breaks either. It has provided WJCT a very uniform and constant off-air sound and fixed dialnorm settings no matter where the material is coming from. FABULOUS gadget.

**Duane Smith**
Director of Technology
WJCT-TV and FM

Our loudness control problems have virtually ceased thanks to the AERO.air (5.1). The LAMBDA is a very powerful (and cool) box. I still need to teach myself how to use it to its fullest potential and hope to add the AC-3 option later this year which will make it a huge addition to my troubleshooting arsenal.

**Prentiss Laird**
Engineering Technical Manager
CBS 42 KEYE

We own two AERO.air (DTV) units and are extremely pleased with their performance. Both units were easy to configure and have provided reliable processing and level control. We also selected these units for the ease of 5.1-to-stereo downmix. We have agreed to supply our cable providers a direct SD feed for several more years. This downmix ability provides us with a single platform solution for both our HD and SD feeds. We monitor all feeds with our LAMBDA monitoring unit. It gives us a good handle on our 5.1 processing and the dialnorm of any feed on our wideband router.

**Jay Nix**
Director of Engineering
KSHB-TV

# THE TELOS ALLIANCE™

Telos Omnia 25 seven Axia LINEAR ACOUSTIC®


**TELOS OMNIA AXIA**
1241 Superior Avenue
Cleveland, OH 44114 USA

**24/7 TECHNICAL SUPPORT**
+1 216 622.0247

**MAIN OFFICE**
+1 216 241.7225

**25-SEVEN**
1085 Commonwealth Ave #360
Boston, MA 02215 USA

**24/7 TECHNICAL SUPPORT**
+1 216.622 0247

**MAIN OFFICE**
+1.216.241 7225

**LINEAR ACOUSTIC**
108 Foxshire Drive
Lancaster, PA 17601 USA

**24/7 TECHNICAL SUPPORT**
+1.216 622 0247

**MAIN OFFICE**
+1.717 735 3611

**TELOS OMNIA AXIA**
Neretas Street 2/1-107
Riga, LV-1004 Latvia

**24/7 TECHNICAL SUPPORT**
+1.216.622.0247

**MAIN OFFICE**
+371 67505580

**LINEAR ACOUSTIC**
Unit 916, Qingniangongshe,
No 104 Chaoyang North Road
Chaoyang District, Beijing, China
北京市朝阳区朝阳北路104号青年工社916室

**24/7 TECHNICAL SUPPORT**
+1 216 622 0247

**MAIN OFFICE**
+86 10.58203666