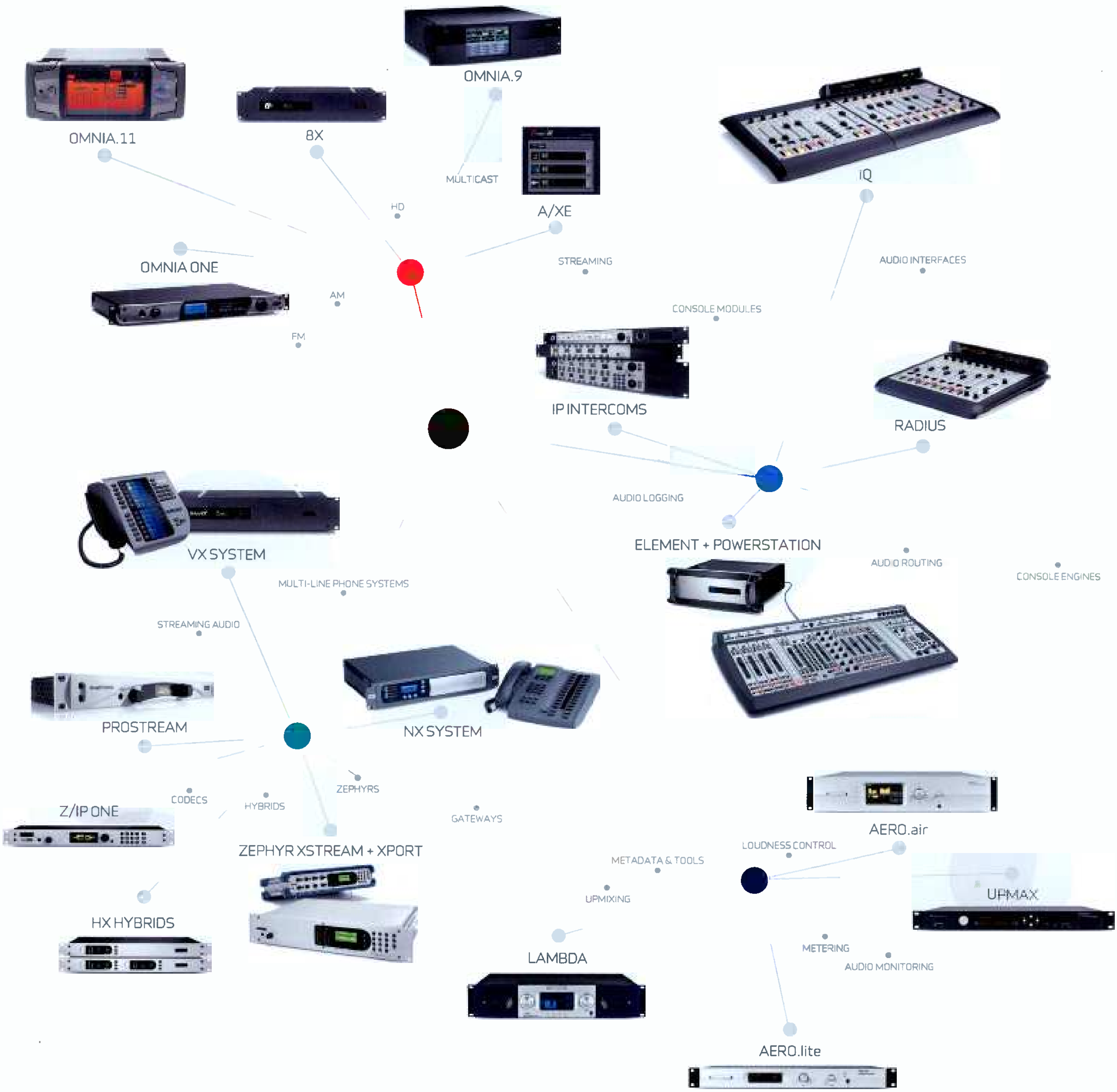
NOW!

TELOS | OMNIA | AXIA LINEAR ACOUSTIC

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



VOL.8 2012-13

28 YEARS

TELOS OMNIA AXIA LINEAR ACOUSTIC

1984

Steve Church invents Telos 10, the world's first DSP adaptive telephone hybrid, and first DSP based product for radio broadcast.

1985

Telos Systems Founded.

1986

Frank Foti designs Vigilante audio processor at WHTZ, New York.

1989

Telos ONE Hybrid.

1991

Cutting Edge Unity 2000 audio processor.

1992

Cutting Edge merges with Telos Systems.

Telos develops Digital Dynamic Equalization adaptive hybrid EQ.

1993

Telos Zephyr introduced; combines MPEG Layer-3 and ISDN for CD-quality remotes.

1996

Telos introduces MP3 for real-time webcasting.

1997

Cutting Edge debuts Omnia.fm Digital Audio Processor.

Telos Audioactive Encoder pioneers hardware-based MP3 streaming.

1998

Cutting Edge becomes Omnia Audio and introduces the Omnia.am and Omnia.net.

1999

Console designer Michael "Catfish" Dosch joins Telos.

Telos introduces hybrids and multi-line talkshow systems for ISDN phone lines.

Omnia introduces Omnia.3

2000

Telos Series 2101 debuts; world's first whole-plant talkshow system.

Omnia.6-fm premieres, world's first 96 kHz/24-bit broadcast audio processor.

2001

SmartSurface networkable control surface shown at NAB.

Zephyr Xstream pioneers low-delay MPEG AAC-LD coding.

Omnia.6FM introduced.

2002

Omnia.5 audio processor for FM and AM.

Telos introduces ProFiler automated program archiving.

Zephyr Xport becomes the first codec to use advanced aacPlus audio coding.

2003

Livewire Audio-Over-Ethernet technology unveiled.

Omnia.6-EX debuts with simultaneous FM, HD Radio™, DAB processing.

Omnia/Crown develop Processing Card for Crown Transmitters.

Axia division of Telos is launched.

2004

Omnia-5EX debuts, first HD Radio processing for AM.

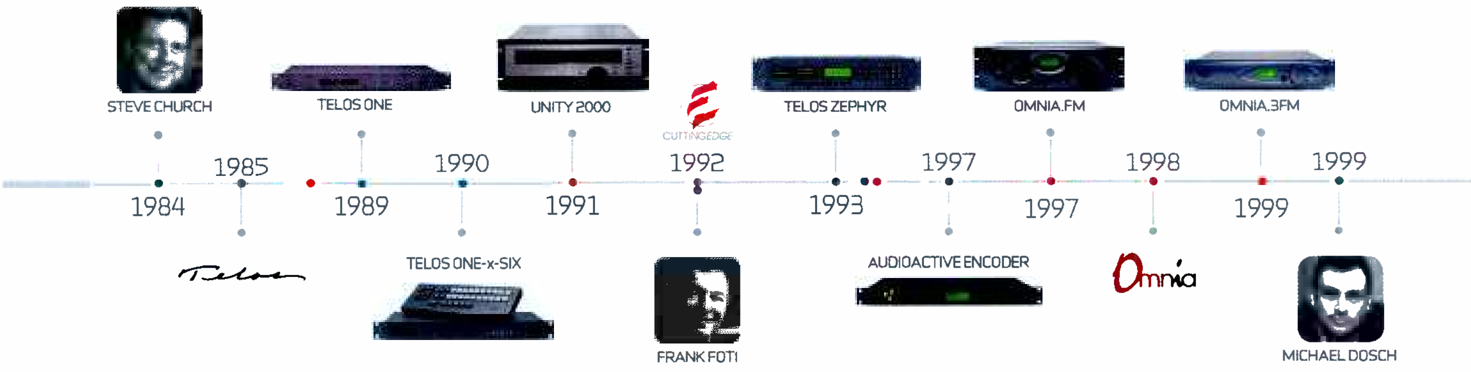
First Livewire-connected studios built.

2005

#1 Station in Los Angeles upgrades to Omnia.

Zephyr surpasses the 14,000 mark in sales.

Axia introduces Element and 100th Axia control surface ships.



2006

- Telos 10 telephone hybrid, the ISDN codec and Axia Livewire tech voted among most influential innovations of broadcasting's first 100 years by readers of Radio magazine.
- Axia iProFiler software introduces "purely networked" audio logging.

2007

- Omnia supplied technology for the first FM-HD surround broadcast in the world over WZLX, Boston.
- Telos debuts Zephyr/IP, or "Z/IP", for use on public Internet data links, and Nx12 POTS/ISDN Talkshow System.

2008

- Linear Acoustic merges with Telos Systems, bringing former Dolby product manager Tim Carroll's extensive television audio background to the mix.
- Linear Acoustic provides upmixing products and technical services for NBC's coverage of the 2008 Olympic Summer Games in Beijing.
- 1000th Axia console placed in service.

2009

- Omnia A/XE software combines Omnia audio processing with streaming encoder.
- Axia simplifies IP-Audio with world's first zero-configuration AoIP switch in PowerStation console engines.
- Steve Church and Skip Pizzi co-write Audio Over IP reference book for Focal Press.
- New Nautel transmitter line includes Livewire IP-Audio interface.

2010

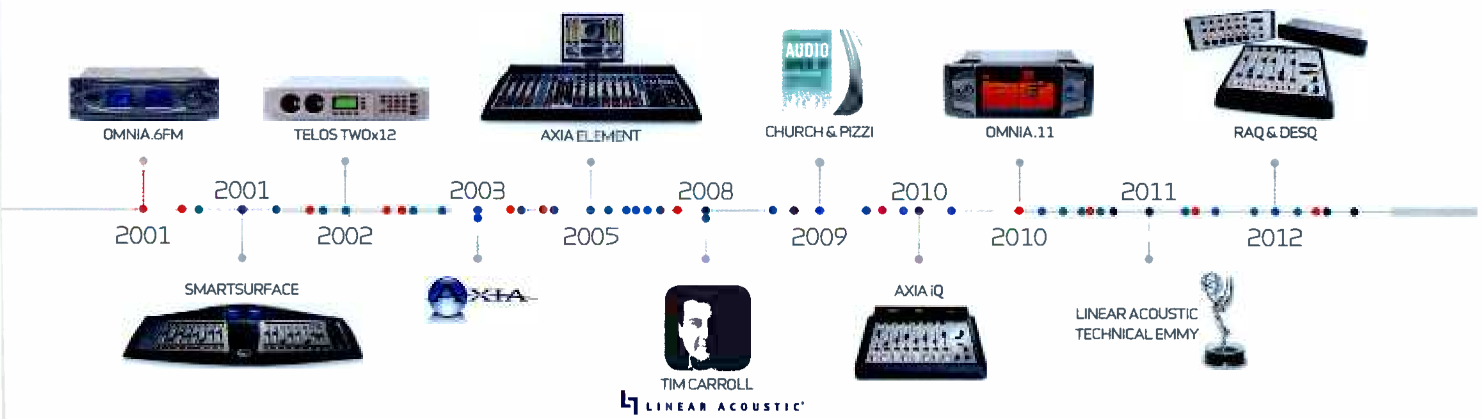
- Telos announces VX, the world's first VoIP phone system designed for broadcasters.
- Telos Z/IP ONE IP codec debuts.
- Steve Church receives NAB Radio Engineering Achievement Award.
- Omnia.11 begins shipping to radio on November 11.
- Axia introduces IP Intercom, world's first broadcast intercom for AoIP network. Axia iQ console debuts.
- Linear Acoustic provides 24 AERO.qc units and technical services for NBC's coverage of the 2010 Olympic Winter Games in Vancouver.

2011

- Linear Acoustic recognized by the National Academy of Television Arts and Sciences with a 2010 Technical EMMY® award for audio/metadata loudness control technology.
- 7000th Omnia ONE ships.
- Omnia F/XE released, specially engineered for file-based audio processing and encoding.
- Axia debuts OpenAoIP.com and the Livewire Limitless License to encourage equipment interoperability.
- Telos debuts Hx1 & Hx2 POTS hybrids; 1,500 are sold within a year.

2012

- Telos ProSTREAM hardware Internet processor / encoder is introduced.
- Omnia.11 on-air in all Top 10 US markets.
- Axia RAQ and DESQ compact AoIP mixers introduced.
- Linear Acoustic provides products and technical services for NBC's coverage of the 2012 Olympic Summer Games in London.
- Linear Acoustic partners with Dolby to develop next generation of DTV audio products.



NOW!

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AXIA

LINEAR ACOUSTIC


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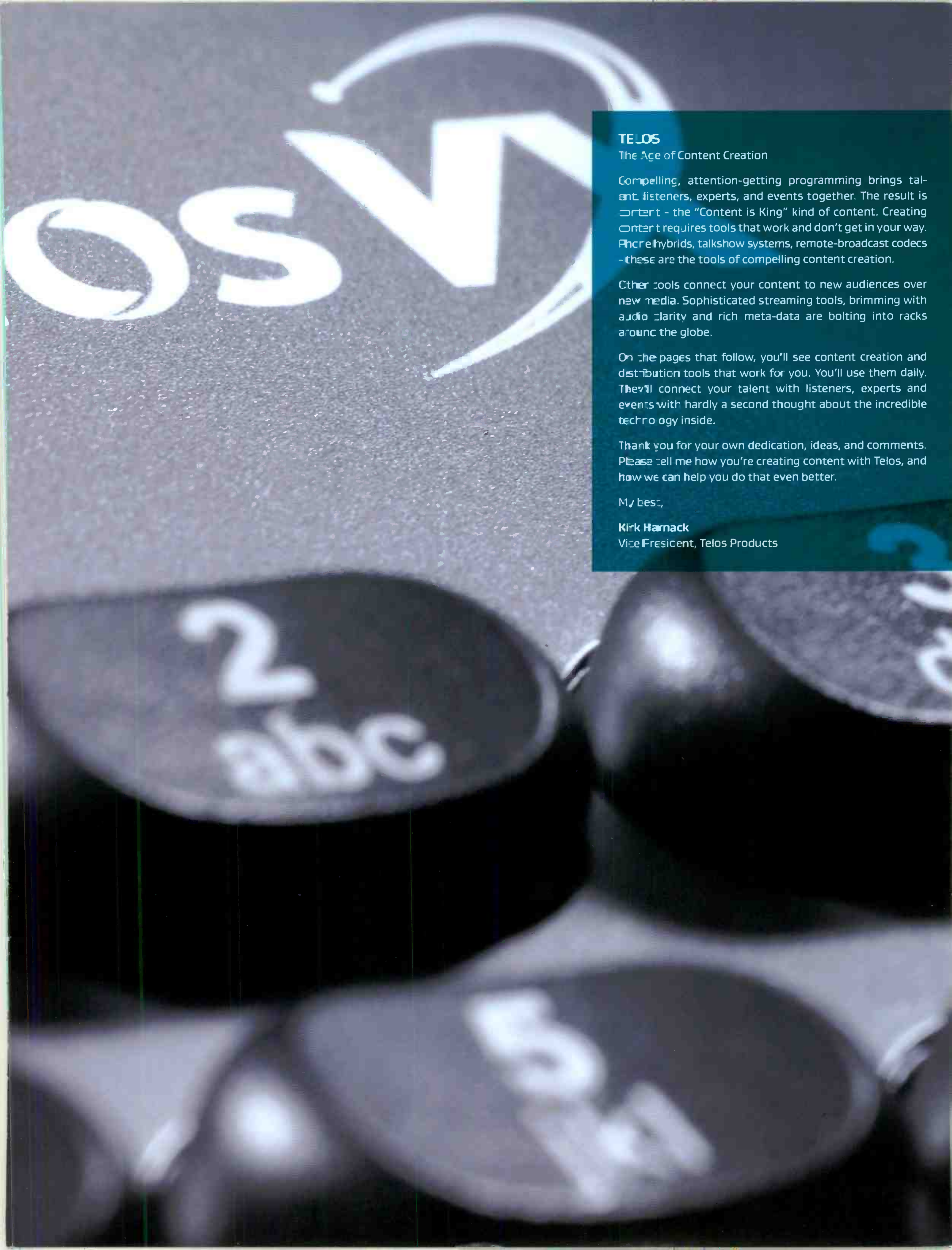


TELOS

The Age of Content Creation

Compelling, attention-getting programming brings talent, listeners, experts, and events together. The result is content - the "Content is King" kind of content. Creating content requires tools that work and don't get in your way. Phone hybrids, talkshow systems, remote-broadcast codecs - these are the tools of compelling content creation.

Other tools connect your content to new audiences over new media. Sophisticated streaming tools, brimming with audio clarity and rich meta-data are bolting into racks around the globe.

On the pages that follow, you'll see content creation and distribution tools that work for you. You'll use them daily. They'll connect your talent with listeners, experts and events with hardly a second thought about the incredible technology inside.

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

My best,

Kirk Harnack
Vice President, Telos Products

FOUR STEPS TO THE BEST PHONE CALLS EVER

GETTING THE MOST FROM NEW PHONE TECHNOLOGY

One hundred thirty-six years ago, Alexander Graham Bell spoke the famous sentence "Mr. Watson—Come here—I want to see you". Chances are good that you're still using that same technology today when you put phone callers on-air.

Broadcasters have used Plain Old Telephone Service ("POTS") for some or all of the talk path from listener/callers or interviewees to the studio. We've used POTS for breaking news reports as well as sports coverage and for remote or "outside" broadcasts. As the least-common-denominator technology, POTS is still the technical glue that connects newer digital services together, often at the broadcast studios. And, while ISDN gave us a digital audio path to the local telco switch, we've had no chance to upgrade the audio codecs employed in ISDN telephony, as ISDN was a direct, functional replacement for analog POTS. Moreover, ISDN - like POTS - is a circuit-switched technology with the attendant limitations and expenses that become anomalous in today's packet-switched, Internet-centric age.

As limited in quality as POTS may be today, it's still better than Mr. Bell's early work with analog telephony. The same is true with Voice over IP ("VoIP"). Early efforts suffered from a nascent, slower Internet, competing and incompatible standards, and very low bitrate codecs that didn't sound good. Today VoIP has grown up to be the world-class telephony player. And VoIP over Session Initiation Protocol ("SIP") is the standard with which old telcos, new providers, and end users are all connecting. SIP is the protocol that we broadcast engineers will be getting familiar with, too.

WHY VoIP FOR BROADCAST?

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.

VoIP is a natural for broadcasters as well, interconnecting the phone system with audio interfaces, phone sets, console controllers, and PCs running screening software by way of efficient, low-cost Ethernet. Using VoIP, you can finally share phone lines among multiple studios and route caller audio anywhere in your facility, easily and instantly. Got a hot talk-show 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. A VoIP talkshow system can even connect with your business office's VoIP PBX to allow easy call transfers.

But VoIP from Telos isn't just business-class VoIP; it's tailored to the requirements of broadcasters. Every incoming line has its own fifth-generation Telos 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 truly consistent caller audio levels. New Acoustic Echo Cancellation from FhG removes feedback and echo in open-speaker studio situations. And should 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. The quality of nearly all calls is improved, too, thanks to less transcoding and no 4-wire to 2-wire (digital to POTS) transitions.

WHAT ABOUT THOSE "FOUR STEPS TO THE BEST PHONE CALLS EVER?" HERE THEY ARE:

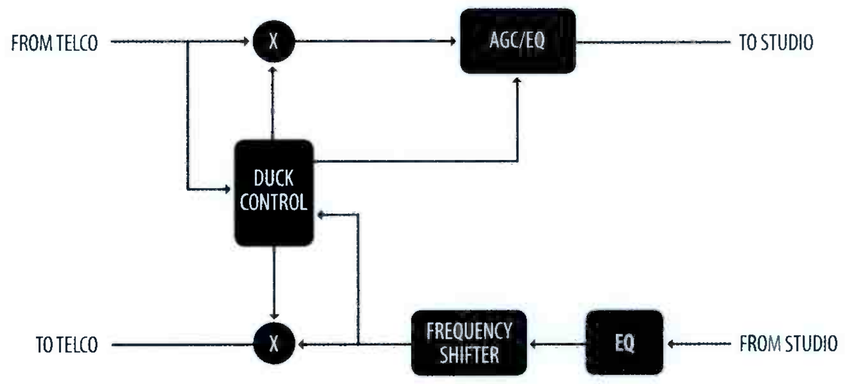
1. FLUSH THE POTS.

Work with your Incumbent phone provider or a Competitive provider to convert your existing POTS lines to SIP Trunks. This can be the best way to bring telephone calls into your facility. However, in many regions, the old Incumbent telco is less prepared to bring you SIP Trunks than a Competitive carrier is. Don't hesitate to check out SIP services from competing carriers. Some competitors will offer the physical connection, too, such as a new T1 path or an alternate copper or fiber IP connection to your facility. Other Competitive carriers require that you supply your own IP connectivity. Consider getting a dedicated data connection for your SIP Trunk service and using MPLS (a type of Quality of Service protocol) or at least Packet Prioritization through your router. What if you just can't convert to SIP from the phone company? No worries, you can still take the next three steps toward better phone audio by using a "gateway" device. Gateways convert POTS (or T1/E1 or ISDN) to VoIP over SIP; exactly what you need to upgrade in steps 2 through 4.

2. USE SIP-NATIVE TELEPHONE HYBRIDS.

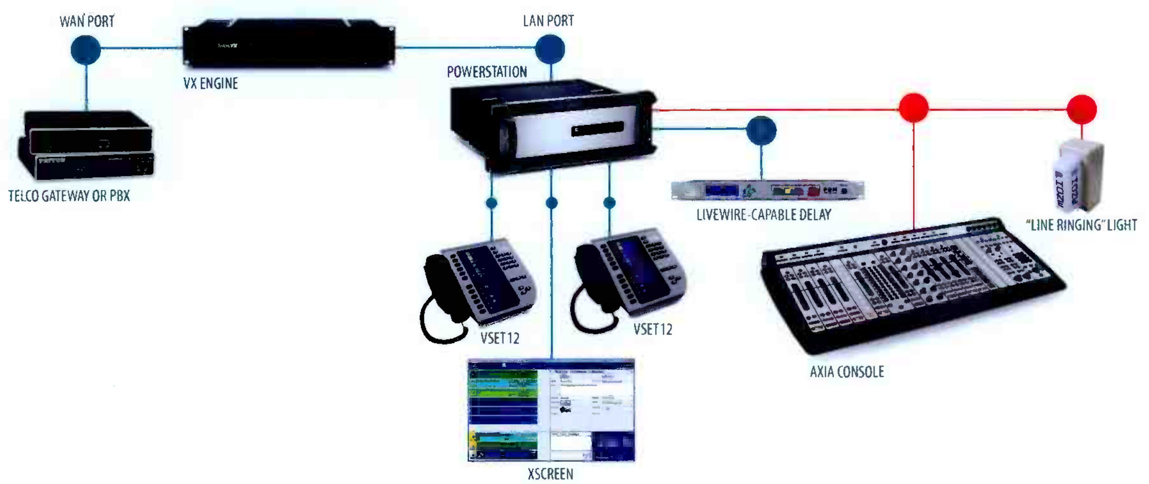
Many broadcasters bring SIP or ISDN into their facility, but then convert these connections to the lowest-common-denominator, POTS, for connecting to their older on-air telephone hybrids. This dual conversion - to POTS and back to 4-wire (inside the hybrid) - only compromises quality. The Telos VX Engine is chock-full of SIP-native phone hybrids. Indeed, instead of switching multiple phone lines to a couple of hybrids, the VX Engine terminates each SIP call with its own, dedicated hybrid for unmatched clarity and superior conferencing.

FOUR STEPS TO THE BEST PHONE CALLS EVER



3. GIVE EACH CALLER AN AUDIO PROCESSOR.

Most phone hybrids have either no audio processor or just a basic audio limiter or compressor - not enough when caller audio is so variable in quality. If you want your callers to sound consistently great, with similar loudness and frequency response call-to-call, then audio processing designed specifically for callers is what you want. The Telos VX delivers 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. Wide-band acoustic echo cancellation from Fraunhofer IIS completely eliminates open-speaker feedback.



4. GO NETWORKED AND GET FLEXIBLE.

Great sounding phone calls on every show implies flexibility in operation. It's no good to have the best new equipment stuck in one studio when you have to move to another temporarily. That's why using a networked on-air talkshow system is critical to producing great-sounding callers every day. A networked on-air talkshow system lets you move from studio to studio, keeping exactly the same show structure, including your Warmline and Hotline. No matter which studio you're in today, you get the same clear, consistent caller audio. A networked talkshow system affords full supervision of all line appearances, so it's easy to share desired lines across some or all studios, and even with a business PBX. Call screening is flexible, too, with no special or separate wiring needed; you get full call screening capabilities anywhere on the network.

MORE TIPS FROM THE FIELD

For the last couple of years, Joe Talbot, Product Manager for Telos, has been helping dozens of broadcasters execute these four steps. From single stations to worldwide networks, Joe and Telos users are bringing the absolute best telephone quality possible to their programming.

Together they're demonstrating that it's possible to dramatically improve audio quality and operational flexibility, while reducing costs and the number of components in the studio on-air phone system.

WHEN CONSIDERING SIP TECHNOLOGY JOE SUGGESTS CONSIDERING THESE QUESTIONS:

- How can we keep as many phone lines as possible in the four-wire domain?
- How can we make the system as reliable as possible while still consolidating delivery facilities?
- How can we best position ourselves for the future of telephony, which will be 100% IP?
- How can we identify the best service provider choices in our area and for our situation?
- How can we make these dramatic changes without adding complexity or adversely changing our users' experience?
- What are our risks and what is our fallback plan? How much backup do we need?

Thinking about office and studio phone integration, we've learned that engineers should consider more than "is this PBX SIP capable" when shopping. The overall "openness" of the system and manufacturer, the licensing and maintenance costs should be considered when choosing a PBX. In several locations, we've found that it would be relatively simple to connect the VX directly to the PBX — but then found out that the arbitrary licensing costs to the PBX vendor would be several thousand dollars.

We're finding that SIP providers will often assume that you want the G.729 compressed codec, not understanding that audio quality is a primary concern that you're willing to pay for. Never lose sight of why you are making changes in the first place! Generally speaking, you'll want your SIP provider to deliver call using the G.711 codec as a minimum.

Consider how your lines are delivered and how many you actually need. You will get fewer channels (17) on a T1 IP connection than a TDM T1 connection (23). If capacity is an important issue, your best choice is to have SIP trunks delivered in some other (non T1) fashion if possible, or simply use a SIP gateway, bringing in TDM on a PRI from the carrier, then converting to SIP. That scenario is all digital and four-wire.

We've learned that the Incumbent (ILEC) phone companies often are usually not the best choices for SIP service at this time. They can be mired in old technology, plus can suffer from a "not invented here" attitude. Competitive (CLEC) phone companies are often into SIP technology with "both feet" and are committed to delivering excellent quality and reliability.

My best,

Kirk Harnack
VP, Telos Products

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.

VX connects to traditional POTS and ISDN telephone lines via standard Telco gateways. But it can also connect to VoIP-based PBX systems and modern SIP Trunking services to take advantage of low-cost Internet-delivered phone services. Using standard Ethernet as its data backbone, VX significantly eases the cost of phone system installation, maintenance and cabling, while making it easier than ever for talent to take control of their phone system. VX is truly the future of broadcast phones.

With Telos VX, you get the flexibility and low cost of modern telephone SIP networking joined with power digital signal and audio processing. You can move and share lines between studios at the touch of a button. 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. To make the most of this networked environment, we've built VX around the VoIP standard.

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.

WHY VoIP FOR BROADCAST?

VoIP is a natural for broadcasters, interconnecting the phone system CPU with audio interfaces, phone sets, console controllers, and PCs running screening software by way of efficient, low-cost Ethernet. Using VoIP, 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 allow easy call transfers.

But it's not just VoIP — It's VoIP from Telos. Every incoming line has its own fifth-generation Telos 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 should 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.

VX uses Ethernet as its network backbone, a powerful yet simple way to share phone lines among studios and connect system components. VX plugs right into Axia IP-Audio networks, connecting multiple channels of audio and control via a single Ethernet RJ-45. If you don't have an IP-Audio network yet, that's OK; VX works with Telos audio interfaces to 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 nearly 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 (software or hardware) is possible. » Modular, scalable system can be easily expanded to manage a network of up to 20 studios, each with a dedicated Program-On-Hold input – truly a "whole-plant" solution for on-air phones. » Up to 16 hybrids, with as many as 48 active calls (up to 4 per hybrid), may be placed on-air concurrently. » Each call receives a dedicated hybrid for unmatched clarity and superior conferencing. » Native Livewire integration: one connection integrates caller audio, program-on-hold, mix-minus and logic directly into Axia AoIP consoles and networks. » Connect VX to any radio console or other broadcast equipment using available Analog, AES/EBU and GPIO interfaces. Audio interfaces feature 48 kHz sampling rate and studio-grade 24-bit A/D converters with 256x oversampling. » Powerful dynamic line management enables instant reallocation of call-in lines to studios where demand is greatest. » 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 rackmount 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 as normal. 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.

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 desired.

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

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 configured 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.

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.

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.

VX + By Broadcast Bionics

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. VX+ adds enhanced functionality to VX including outgoing announcement and voice mail capability. VX+ runs as a Service on a Windows PC.

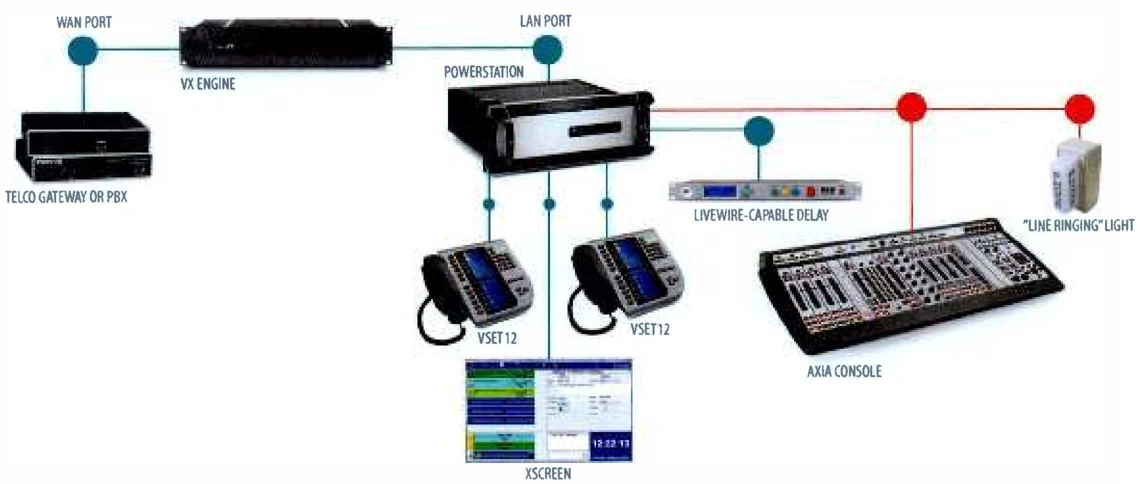
PAINLESS HOOKUP

READY. SET. BROADCAST.

AXIA INSTALLATION

The Telos VX is a "facility wide" on-air telephone system. That means multiple studios, multiple stations, multiple shows minimal hardware requirements. With VX, there's no need for 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.

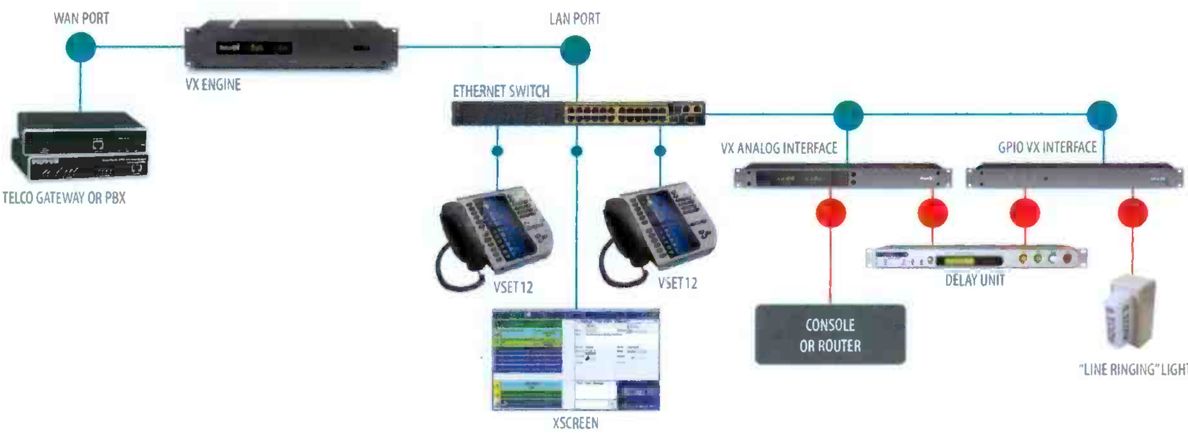
Telco is delivered via IP through a POTS, ISDN or T1 gateway device, a SIP PBX, or a dedicated IP circuit using SIP Trunking. Got an Axia Livewire AoIP studio network? Telos VX will plug right in. Audio inputs and outputs are Livewire real-time audio channels and travel over your existing Axia system 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.



RADIO ONE, WASHINGTON DC

NON-AXIA INSTALLATION

Don't have IP-Audio networking yet? Not to 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 Talkshow 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. But there's more: Nx systems feature caller audio sweetening by Omnia, special echo cancellation to tame tricky VoIP and cellular calls, and anti-feedback routines to tackle the acoustic feedback that plagues open speaker applications.

Telos Nx6 and Nx12 systems can be ordered to work with your choice of POTS or ISDN (BRI) phone lines, and work with a variety of control surfaces, including the Telos Desktop Director, Call Controller and Console Director drop-in module. Of course, there's also an Ethernet connection for use with call screening software (and one-click connection to Axia IP-Audio networks). With Nx talkshow systems, talent and producers both benefit from unique Telos features that help make shows run smoother, faster and more error-free, such as our exclusive Status Symbols visual call management icons that clearly show line and caller status.

Nx6 works with up to 6 telephone lines and Nx12 with up to 12 lines, and each have four hybrids for extra flexibility in fast-paced talk environments. They both feature a useful "dual studio mode" that allows a single system to power phones for two studios simultaneously, 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. Function keys on Desktop Director and Call Controller devices let you command GPIO-style outputs for push-button command of profanity delay systems and recorders.

Nx6 and Nx12 work flawlessly 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 a 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, and with adjustable caller ducking to help your hosts keep control of the conversation. Finally, a sophisticated pitch shifter and studio adaptation routines help keep feedback from appearing when taking calls with open speakers.

But wait - there's more. 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. Speaking of Ethernet, Nx systems ease setup and administration with their built-in Web servers; just hook a computer to the network to perform configuration and remote monitoring functions. 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 your premiere choice to make, take and screen calls with your Nx phone system. These sophisticated, yet easy-to-use phonesets make fast-paced production a snap. And caller management has never been simpler, thanks to intuitive Telos Status Symbols — clear, easy to read graphical icons that convey line and caller status at a glance.

Desktop Director helps you screen calls quickly and efficiently using deluxe features like the built-in handset, speakerphone or optional headset. Hosts receive immediate information about line availability, on-hold and ready-for-air queue status from Status Symbol icons. An extended version for use with Nx12 can control all four hybrids individually.

CALL CONTROLLER

The Telos Call Controller is a simplified, cost-effective option for call screening and on-air control. The Call Controller connects to the Nx6 or Nx12 in the same way as the Desktop Director. It uses an external, user provided, telephone for call screening and studio telephone operation. Simply connect the Call Controller to your Nx system, then plug in any compatible analog telephone and you're on your way. Like the Desktop Director, Call Controllers have large buttons and intuitive Status Symbols to help talent keep track of line and hybrid status.

CALL CONTROLLER FOR ELEMENT CONSOLES

These drop-in modules for Telos Element 2.0 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; coupled with Element's built-in dialpad, talent can make, take and bring callers to air without ever taking their eyes off the board.

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 Telco gateway for Axia's new iQ console. 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 puts audio, hybrid control and backfeed for six phone lines onto one skinny CAT-5 cable. Setup is simple: plug it into your Axia network, do some fast web-based configuration, and your 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.

JASON WISNIESKI, iQ6 PROJECT MANAGER

"The iQ6 realizes the Livewire dream - one connector for all your studio audio and control - and brings it to telephony. It really brings home the benefits of running a Livewire-based studio. I love that it's powerful as its own system, and very flexible when paired with an iQ Console System."

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 that 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 line 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.

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. Forget tape decks and expensive logging hardware — ProFiler runs on any Windows PC to produce time-stamped audio archives you can listen to on standard MP3 players.

Each ProFiler package includes a Telos audio card with pro-level balanced audio inputs and buffered parallel closure 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 cards 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.

ON THE ROAD WITH TELOS VX

REAL WORLD BROADCAST VoIP

We've been busy as beavers lately, shipping Telos VX broadcast VoIP phone systems to clients around the world. They're now running in radio and television stations and connected to PBX's, dozens of telephone service providers, and many different types of hardware.

The Telos VX is a "whole-plant" broadcast telephone interface that delivers the best quality telephone audio possible today. It helps you make the most of lines and trunks, making them available in any studio at any time.

I get to see radio stations everywhere!
Naturally, I'm accustomed to only the finest accommodations.

One of the things that I get to do is go to stations and broadcaster get-togethers like SBE meetings, NAB and the like, where I see a lot of old friends and always make more. I get plenty of questions, sure, but I also get to hear a lot of great ideas from clients about how they use the VX system. It's the best part of my job. We've learned a lot over the past year from our friends and colleagues, and even phone companies (if you can believe that!).

Some of the best VX installations are where the office and studio phones are connected together. This means that you don't need multiple phones in the studio, and you can do things like transfer calls from the front desk directly to the studio. It's great to have just a single, powerful VSet telephone in your studios. The other thing is, with this kind of 100% digital "four-wire" line connection, the audio quality is really, really impressive.

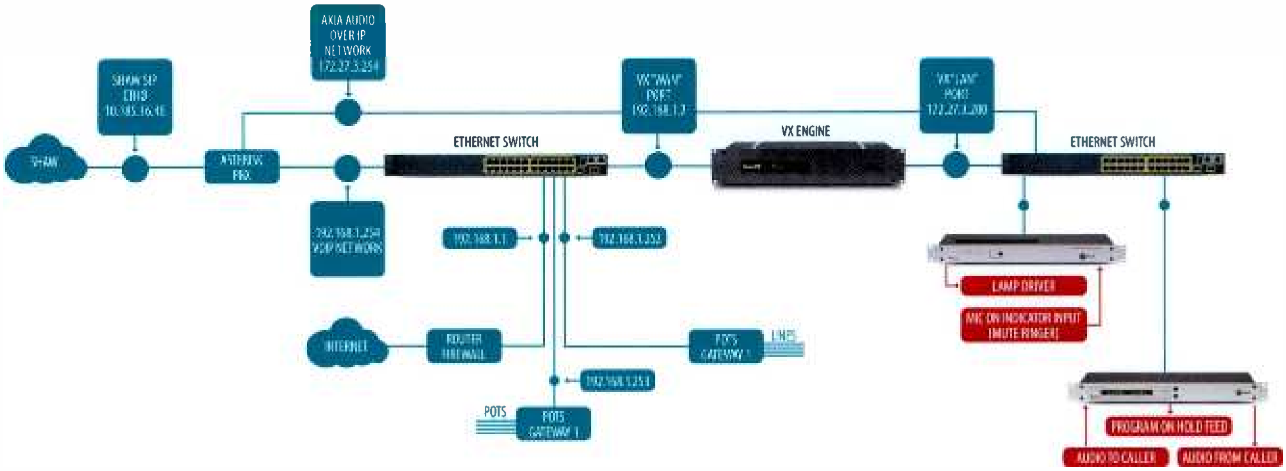
Our clients have installed and tested the VX with many of the most common PBXs out there. As I write this, there are VX systems connected directly to PBX products from Avaya, Cisco, NEC, Mitel, Nortel and open source Asterisk. Now, not every model from these manufacturers supports SIP (Session Initiation Protocol), even though it's a standard. Some of them charge extra licensing per extension or for SIP capability. Some support only SIP trunks. But VX is amazingly flexible, and our crack Telos Support team can help you hook up VX to just about whatever you happen to have.

Even if you just want to keep your POTS service because your PBX is stuck in the '80s, we've been working with Patton Electronics to make using their advanced SIP gateway products easy to buy and implement. You always have a direct SIP upgrade path. We've got you covered.

You can even use VX to get rid of that wall of old auto-answer telco couplers. The VX can be programmed to answer a line, or several lines in a hunt group, and feed great quality audio to those lines and/or receive audio from them.

A VSet phone and a computer are all that's needed to record broadcast-quality remote drops. Tampa's Cox cluster uses smartphones with G.722 to achieve "HD Voice" remotes.

Block diagram of VX installation at Corus, Winnipeg, shows how VX integrates



COX MEDIA

One of the coolest VX installations I've seen was down in sunny Tampa, Florida, where our friends at the Cox stations use their VX, not just for phone calls, but also for remotes! Once they got connected to their IP Phone service provider, PAETEC, they set up their VX system so that their street team's iPhones and Android phones could call right into any control room, and go live instantly, using the wideband G.722 codec. They found free apps at the iPhone store, and made a few simple setting tweaks so that all the street team had to do was dial the studio's 4 digit extension number! These remotes are fast, easy and toll-free. No special setup is needed at either end to make the call; it's completed over the 3G or 4G network, or using available WiFi. The system will even "talk" to other vendors' codecs, without complication.

CORUS COMMUNICATIONS

Up in balmy Winnipeg, Manitoba, Canada, our friends at Corus Communications get phone service from Shaw Cable. Shaw did a great job of providing SIP trunking, an example of another excellent option for studio phone service. Depending on your location, there are often more service provider options than you think, and they're worth considering seriously. The Corus stations also have stations coming from their Mitel PBX and lines from MTS, the local utility telephone company.

Take a look at the diagram above, and you'll see that Corus installed an Asterisk PBX to add voice mail, call detail recording, Network Time Protocol (NTP), and VPN access for maintenance. In the ultimate act of recycling, they made it out of an old spare PC, and just kept coming up with more uses for it.

Patton SmartNode VoIP gateway.

Using our partner Patton Electronics' powerful gateway products is another way to bring lines directly into your VX from legacy providers, or equipment. Patton has ISDN PRI, BRI and POTS gateway devices available, and we work closely with them to make your installation easy with standard configurations, or customization, as desired.

Telos Partner companies Broadcast Bionics and Neogroupe have created special software products that extend and enhance VX capabilities. Bionics' PhoneBOX is a database driven software suite that makes screening and airing phone calls easier. With an emphasis on operator work flow, Caller ID and ANI information are used to sort, filter and track callers, ensuring that the best callers get on the air.

Joe Mauk at Peak Broadcasting's KMJ in Fresno, California liked what he saw in both VX and PhoneBOX, and understood the system's potential. He recognized that a combination of the two would be a particularly good fit for his six-station cluster, which included not one, but two talk stations.

Talent at KMJ uses VX, controlled with PhoneBOX software, to take calls to air.

Do VoIP phones work with analog consoles?
Screener's position at KMJ says the answer is "yes."

His first step was to figure out how best to get lines into his system. The stations were using a large 1A2 key system with a mix of Telco POTS lines and PBX extensions. His goal, as always, was to get the lines delivered in some 4-wire fashion (best quality = least cross-talk). The direct-inside-dial PBX extensions posed no problem, as they were 4-wire from the phone company, delivered on a Primary Rate Interface ISDN circuit. There was some 2-wire at the PBX extension line card — but that didn't matter, as we wouldn't be using those circuits.

Joe needed to use a PRI gateway device or an Asterisk PBX to accept traffic from KMJ's Eon Millennium PBX, and then convert it to SIP to keep the traffic 4-wire all the way. Ultimately, Asterisk was chosen because Joe also wanted to use the G.722 7 khz "wideband" codec, which the VX system supports, for remote broadcasts and live news reports. The routing flexibility that Asterisk provides would give him many other options as well.

The remaining problem was that there were still some old, copper-loop POTS numbers being used with the old 1A2 system. Joe decided that the numbers could be ported over to one of the local-exchange carrier PRIs that feeds his office PBX as DID numbers. The on-air calls then come into the station on the 4-wire Bearer channels of the PRI, thence routed to the Asterisk PBX for use with the VX (and, simultaneously, to analog PBX stations with the old 1A2 gear). The result: complete call routing flexibility, Caller ID support, and a simple fallback solution in case of trouble.

The number port went as expected. The Millenium PBX, the Asterisk PBX and the VX were configured, and a few test calls made. Then, Broadcast Bionics set up PhoneBOX via remote-configuration. The system works flawlessly!

One other thing: KMJ has a wireless Internet provider in the building. We used this to our advantage, adding some off-site VoIP extensions and a couple of trunks via the wireless provider, with great success. Peak Broadcasting now effectively has a backup provider for local service in case the PRI lines, or their PBX, goes down. And they're also able to use the VX's G.722 codec support for remote broadcasts, and for high-quality news actualities from reporters in the field using iPhones.

These are just a few examples of the power and flexibility of VX. Ask any VX client, and they'll likely rave about its ease of use and cost-effectiveness. They'll probably also tell you that one day, all broadcast phones will be VoIP-based. Luckily, you don't have to wait for "one day." With VX, that future is here, now!

Joe Talbot
Product Manager
Telos Systems

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 pioneered the concept of the ISDN codec in the first place! 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 models with built-in mixers and Phantom microphone power help reduce equipment inventory and setup time. Each Zephyr Xstream model has a 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. And 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 three Zephyr Xstream models with capabilities tailored to fit your needs: The standard rack-mount Xstream, the Xstream MX rackmount with built-in DSP mixer, and the portable Xstream MXP, a ruggedized portable version with mixer that's ready for the road.

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. » MPEG AAC (Advanced Audio Coding). The new standard for audio coding permits true CD-quality stereo transmission with a connection speed of just 128 kbps. Xstream includes exclusive Error Concealment technology to prevent the occasional network glitch from being heard. » Low-Delay MPEG AAC-LD coding. Using AAC-LD, you'll enjoy crystal-clear audio quality with 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 is enabled to allow reception of independent audio streams arriving from two distant ISDN lines – great for bilingual broadcasts. » Hand-in-glove operation with companion Zephyr Xport portable codecs to facilitate reception of 15kHz audio using a POTS field connection. » A V.35/X21 option to allow connection to serial synchronous data equipment, for use with dedicated lines, Switched 56 circuits or satellite services. » An Auto Receive mode that quickly determines the correct coding algorithm for incoming audio streams.

ZEPHYR XPORT

THE "GO ANYWHERE" ZEPHYR

When you're going on the road, you want a codec that won't weigh you down. At just seven pounds (3.18 kg), the small, light Zephyr Xport is the perfect companion. At the heart of Zephyr Xport is a custom DSP-based modem, optimized for maximum performance with audio codecs. Exclusive Telos technology lets Xport use a standard analog phone line (or digital phone line, with the optional ISDN interface) to connect with any Zephyr Xstream ISDN codec; a full-featured mixer with mic and line inputs (and selectable audio processing by Omnia) completes the package.

A Zephyr Xport in your remote kit makes your studio's Zephyr Xstream a "universal codec," since you can use either POTS or ISDN to connect with your studio. You save the cost, studio rack space, training time and telephone lines needed to support dedicated POTS and ISDN codecs — not to mention console audio inputs and mix-minus outputs.

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. You get the most reliable connections and the best audio, too.

FEATURES AT A GLANCE

» Superior sound: aacPlus™ coding gives you the best-quality audio of any POTS codec, even at low bit rates. » Optional ISDN capability. You can plug Xport into any POTS outlet and dial your studio's Zephyr Xstream; an easy upgrade lets you plug into digital phone lines as well for ultimate flexibility no matter where you're broadcasting from. Two Xports can even share a single ISDN line! » Super friendly operation: Xport is the easiest-to-use Zephyr ever. Anyone can make it play. » Ultra mobility: Xport is light-weight, portable, durable. You can tuck it into a flight bag! » 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.

Z/IP ONE

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

Broadband Internet is everywhere, which makes it ideal for live remotes. Unfortunately, public IP links are also notoriously erratic. You might be lucky enough to get a good connection. . . but even if you do, 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 newest member of the Zephyr family, 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 phone data services — even from connections behind NATs and firewalls.

Telos collaborated with Fraunhofer (the developers of MP3) to develop a unique coding 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. It delivers reliable audio despite varying network conditions, and without the need to fiddle with settings or codecs.

Z/IP ONE 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. But if congestion starts to occur, Z/IP ONE automatically lowers bit rate and increases buffer length to keep audio flowing at maximum quality.

Another way Z/IP ONE extracts excellent quality from even not-so-excellent IP connections lies in its use of a new codec based on low delay AAC: Advanced Audio Coding-Enhanced Low Delay (AAC-ELD), which gives excellent fidelity at low bitrates with nearly inaudible loss concealment and very little delay. (You have your choice of standard high-performance codecs too, including AAC-HE, AAC-LD, MPEG4 AAC-LC, MPEG2 AAC-LC, G.711, G.722 and even linear PCM.)

Z/IP ONE's front panel is friendly and simple to use. Naturally, there's a built-in Webserver too, for remote control and easy configuration using any Web browser. Our exclusive 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 convenient XLR ins and outs, a Livewire port for IP-Audio, WAN jack for connection to "the outside world", and even a parallel port for GPIO contact closures. All in a compact, 1RU package.

Z/IP ONE. The convenience of the Internet — the sound of Telos.

FEATURES AT A GLANCE

» Z/IP ONE is wireless capable and can connect to IP networks via Wi-Fi, EVDO, and UMTS. » 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. » Dual IP ports for separate streaming and control. » Easy browser setup via built-in Web server. » Push mode for one-way network connectivity such as satellite broadcasts. » Multiple push mode, push to multiple destinations. » Sophisticated NAT traversal support. » Convenient directory server, no need to know another device's IP address. » Transparent RS-232 channel for audio side channel or metadata, e.g., RDS. » 8-bit parallel GPIO port for signaling and control. » Slim 1RU form factor fit is equally at home in a studio rack, remote kit or road case.

iPORT

LIVEWIRE TO IP MPEG GATEWAY

Talk about "high-density" equipment: the 2RU Telos iPort saves you money and rack space by housing eight stereo MPEG codecs in one small device. A pair of 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. You can use a QoS-enabled IP link between two studios with Livewire networks, put an iPort at each end, and 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 the Livewire AoIP standard for 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. Not using Livewire yet? No problem – just pair the Telos iPort with an Axia analog or digital audio node to make a standalone high-density audio codec package.

How do iPort streams sound? Fantastic. For years, Telos has had a tight working relationship with Fraunhofer IIS, the inventor of MP3 and co-inventor of AAC. The encoding algorithms inside iPort are genuine FhG, not some no-name knockoffs. A full range of state-of-the-art codec types and bitrates are supported; the highest-possible quality implementations, running on 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 iPort on a single industrial motherboard, rather than the usual "multiple DSP cards in a frame" approach. Together with the Livewire-only audio interface, the iPort costs a fraction of legacy card-frame designs.

THE IP WAY TO HEAR FROM THERE

WHY IP AND BROADCAST PHONES MAKE THE PERFECT PAIR

TELEPHONY: it's the technology for electronic transmission of voice, fax, or other information between distant parties. Indeed, broadcasters are all about connecting remote sounds, ideas, and voices with our audiences. Telephony is elemental to broadcasters.

We receive or gather - TV folks say "ingest" - audio programming from satellites, telephone callers, remote talent via codecs, and downloaded audio files. We assemble this audio and mix with local content - talent, commercials, traffic, weather - then we broadcast the resulting program audio to our listeners.

Keeping with the telephony theme, it's becoming clear that IP telephony is displacing the traditional connections with which we grew up. And broadcasters using IP telephony in its many forms are noticing something - it frequently sounds better than the POTS or ISDN service it's replacing. This improved audio quality is possible, and now even practical, along several key paths in a broadcaster's operation. Even better, IP technology is generally less expensive than the equipment and services it replaces.

Telephone calls carried in whole or in part by IP sound better than those carried by traditional transports - all other things being equal. Improvements come by dint of better quality phone instruments, "4-wire" connections from end to end, endpoints negotiating best codec usage, and even from intelligent and powerful audio processing in an IP phone system designed for on-air use.

HERE ARE SOME REAL-WORLD EXAMPLES OF IMPROVED AUDIO QUALITY USING IP TELEPHONY:

» A major, worldwide television news network upgraded to a Telos VX IP phone system. The network's audio engineering director reports that all phone calls – even from traditional phones – are sounding clearer and are easily intelligible. Audio levels are brilliantly consistent, too, even from reporters and newsmakers calling in from the other side of the globe.

» At radio stations where a Telos VX IP phone system is installed, radio talkshow hosts, DJs, and call screeners are telling me their callers sound noticeably better - clearer - and with less of that "cell phone distortion". Show hosts spend less time asking a caller to "repeat that, please" and more time in real conversation. Caller audio levels are consistent, too. Board ops hardly touch the phone fader, as the call-to-call level is dependable.

» Calls originating from IP-based phones are beginning to use better-sounding audio codecs. Phone and computer apps now afford wideband voice quality calls using the stalwart mono broadcast codec, G.722. Newer codecs are coming, and will go into Telos IP telephony systems so broadcasters can enjoy even better calls — both from listener/callers and from "remotes" or outside broadcasts. IP telephony lets us use phone apps to communicate with better codecs, giving us great quality voice remotes over convenient cell phones.

» On the high-end of audio quality, a parallel of IP telephony includes dedicated stereo IP codecs and mid to high bit-rate algorithms, such as AAC and its derivatives. Stereo IP codecs are replacing ISDN codecs. They're also replacing other audio connection devices including T1, E1, and RF-based STL systems.

More real-world examples of technical success begin with the glamorous life of professional voice-over artists.

Whether home-based or in a commercial studio, voice-over artists are sending coded audio at far higher bit rates than is possible over ISDN. No long distance call charges, either. VO studios need just a good Internet connection and an IP-audio codec such as the Telos Z/IP ONE. Connecting to a studio in the next town, or to a broadcaster in another country, is as easy as using instant messaging software.

Broadcasters are covering sporting and other location events using IP codecs. Often times a broadcast engineer will have several IP connectivity options, depending on the venue. While a wired Internet connection is nearly always best, there are options for WiFi and cellular 3G, 4G LTE and WiMax across a growing wireless landscape. Dialing and connecting is easy for non-technical talent, and there are never any long distance call charges, even while covering extra innings, overtime periods, or that city council meeting with a never-ending agenda.

We thought IP-audio would make a good backup Studio-Transmitter Link (STL), but now broadcasters are using the Telos Z/IP ONE and Telos iPort at robust bit rates, while relegating their traditional STL systems to backup duty.

Why? Better audio!

If the bandwidth is available, high bitrate AAC or even linear PCM is the way to go. And higher audio sampling rates are typical of IP codecs as compared with digital RF STL systems. A 48 kHz audio sampling rate gives full 20kHz audio end-to-end -perfect for full-time STL, even to HD Radio transmitters, and studio quality audio contribution applications.

IP-Audio is a dream come true for broadcasters networking with other stations to cover local or regional programming. Whether over a corporate WAN or the public Internet, programs of interest over a region – or nationwide – are easily shared and distributed by IP audio codecs.

RADIO ONE, WASHINGTON DC

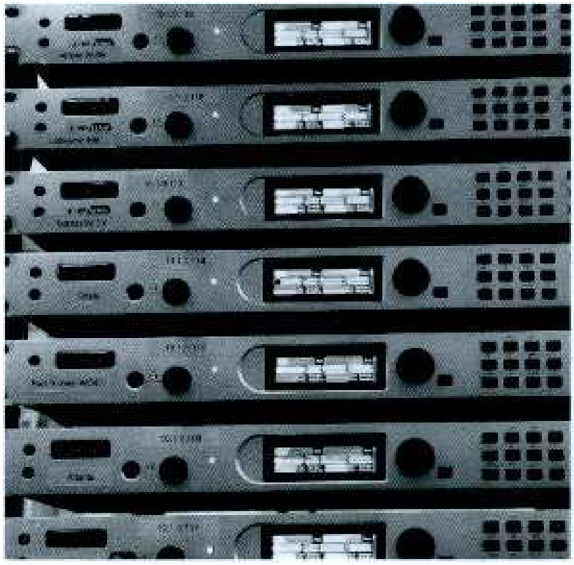
One network in New Zealand, PungaNet, blazed this trail in 2009 by connecting some 25 radio stations in different cities and towns around the country. Their multi-channel system, based on the Telos iPort, allows for one-to-one sharing, one-to-a-few distribution, or one-to-all coverage. Ad hoc networking is easy, too, as stations share talent, programs, and air-time across the network.

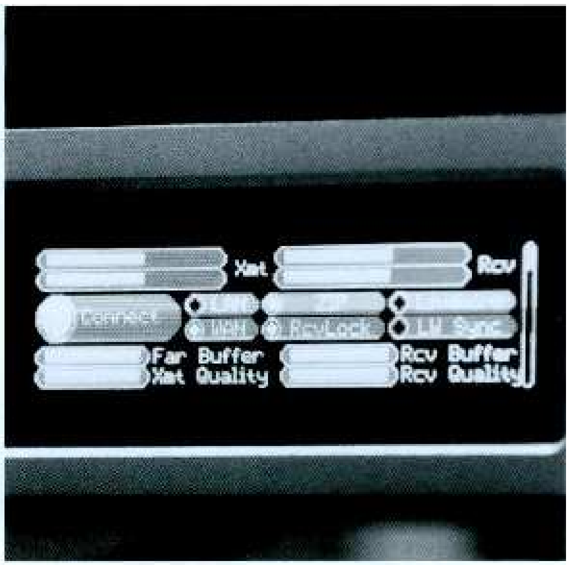
IWI RADIO MAP

A legacy broadcast network in the U.S. is connecting remote talent in New York, Washington, D.C., Los Angeles, and London with editing and production studios over both their corporate WAN and the public Internet. High-quality audio along with IFB, tally lighting, and intercom uses the same IP connectivity, saving money and keeping the connections simple.

A well-known French broadcaster is connecting Paris and Bordeaux for show talent, voice-over, and commercial production. A Telos iPort in each city provides eight bi-directional, low-latency stereo connections, plus GPIO signaling.

A multi-language European broadcaster recently employed IP-audio to keep hundreds of programs on-air, with no interruption while moving studio operations to a new building.





A regional broadcaster in the U.S. is using Telos Z/IP ONE IP codecs exclusively for high-quality, reliable STL duty to eight transmitter sites. Some are close-by while a couple are hundreds of miles away. The AAC coding algorithm at 320 kbps delivers unimpeachable stereo audio to most sites. Lower bit rates using HE-AAC are configured for sites with less IP bandwidth available. A robust IP connection strategy ensures at least two Internet connections at each location, with automatic switching or sharing provided by smartly-configured routers.

There is a theme common to successful IP-audio implementation by all the broadcasters mentioned here; it's a good working knowledge of IP configuration and some basic routing knowledge. While these skills are important for success in IP-audio implementation, they're not difficult to come by.

Indeed, Telos is here to help. As a member of the Telos family, you're entitled to free, personal Tech Support, every day, around the clock, by phone or e-mail. All Telos equipment manuals are online, ready to view or download to your computer or tablet. You also have access to Telos white papers, and our Tech Blog at TelosAlliance.com/blog.

IP technology is here to stay. Literally billions of devices connect with each other every moment of every day using Internet Protocol. Together let's make broadcasting, audio sharing, and connecting with talent and listeners, better, easier, and of the highest quality. The awesome flexibility of IP is right at your fingertips.

My best,

Kirk Harnack
VP, Telos Products
The Telos Alliance

IOAN RUS, PROSTREAM PROJECT MANAGER, TELOS

"ProSTREAM takes the pain out of streaming; as there is no PC to manage, it makes streaming about as easy as you can get. Plus, with the combination of Omnia processing and genuine Fraunhofer AAC and MP3 encoding, ProStream consistently gives you the best audio quality so you can be proud of your streams."

TELOS PROSTREAM

INTERNET STREAMING IN A BOX.

Broadcasters often ask "How do I stream audio to listeners over the Internet?" Up 'til now, the answer too often included mini-jacks, poor-quality sound cards, a PC to maintain, and a collection of software that didn't always play nicely together.

ProSTREAM makes netcasting a lot easier. There's no PC needed; ProSTREAM is a single box – an appliance – that simplifies getting your audio streaming on the 'Net. How easy is it? Just send audio to your ProSTREAM, make a few setup selections and you'll be streaming your audio perfectly to most any stream server or streaming service for worldwide distribution.

ProSTREAM combines Omnia audio processing (to make Internet-delivered audio sound its best) with the latest and most widely used audio codecs. ProSTREAM even lets you embed stream metadata, in all of the most popular formats.

Broadcasters know that Telos is the codec expert — and ProSTREAM puts all of our expertise into a single, integrated streaming appliance. 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. Encode directly to MP3 or MPEG-AAC streams, and then feed a Shoutcast, Wowza, Icecast, Live365 or Adobe Flash Media server for distribution to your listeners.

To make your audio really sing, ProSTREAM comes with sweetening from Omnia Audio, the world leaders in audio processing for broadcast. The exclusive Omnia SENSUS™ process shapes and optimizes audio especially for bit-reduced encoding, so your streams always sound fresh and alive.

The best part: Telos ProSTREAM is neatly self-contained in a 1RU box. Just slide it into a rack and it's ready to go – no more running your mission-critical audio over crash-prone PC hardware and operating systems.

An intuitive Web interface puts you in total control of your audio streams, with remote control of all functions via a standard browser. But there are convenient front-panel controls too, so you can manage your most-frequently needed functions right from the rack. Adjust audio input levels, define a metadata source, select a processing preset, codec, bitrate, and target media server. There's also a built-in headphone amp with 1/4" jack and volume control for monitoring input or output audio.

ProSTREAM comes standard with studio-grade analog inputs and outputs, which can be changed to AES/EBU with an optional card. On the input side, you can use Livewire IP-Audio as an audio source. And on the output side, ProSTREAM delivers fully processed, unencoded audio as well as encoded audio, giving you 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.

No matter what your audio source or how you stream, ProSTREAM delivers flawlessly optimized audio that sounds terrific. Pure. Free of artifacts. And alive with character.

MORE STREAMING SOLUTIONS FROM TELOS

HARDWARE OR SOFTWARE? THE CHOICE IS YOURS.

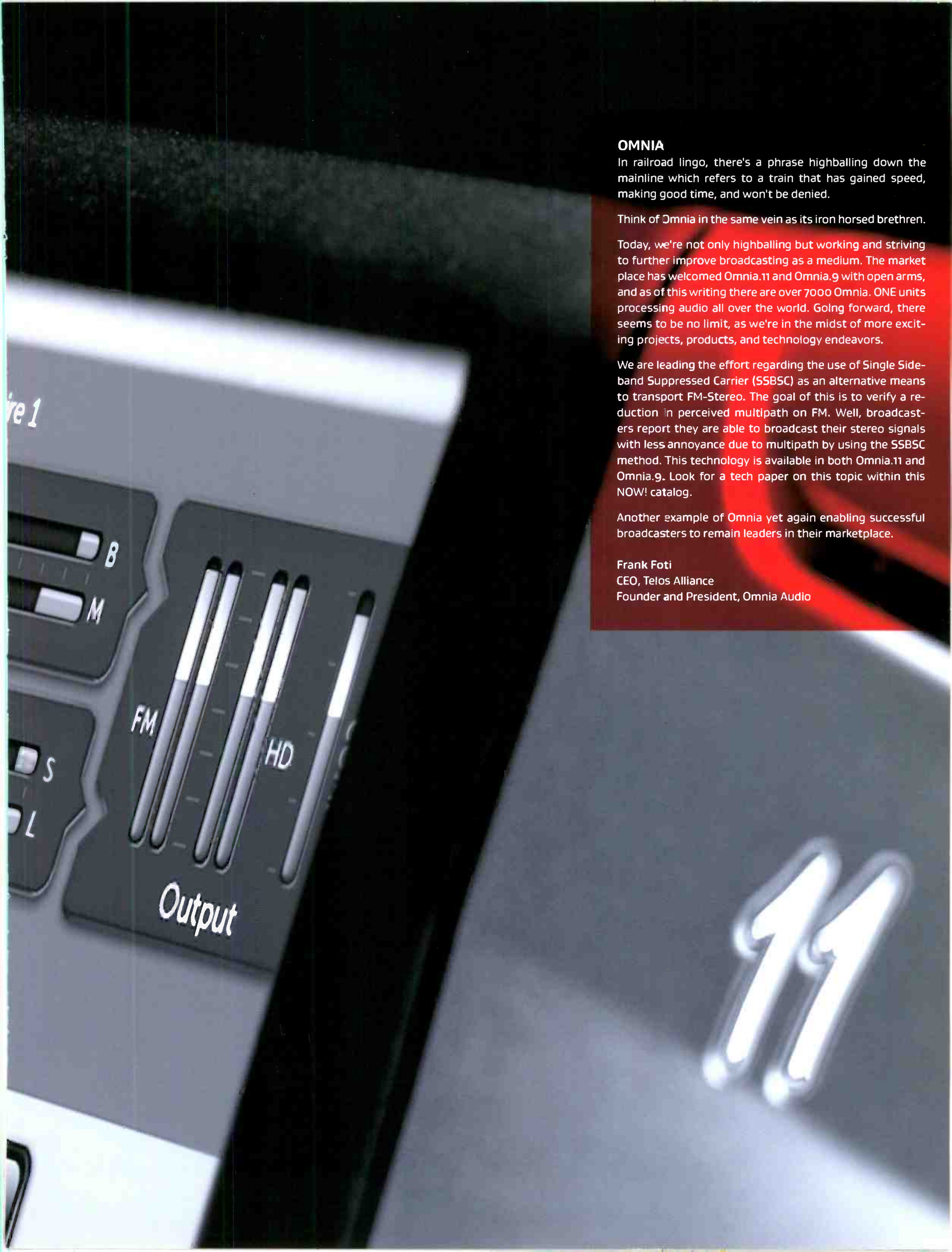
TELOS iPORT

Encode 16 Web streams using one box? It's possible with the iPort MPEG Gateway. iPort contains 16 stereo MPEG-AAC codecs that can be employed to encode your Internet audio. iPort generates standards-based MPEG streams that you can feed to any SHOUTcast, Steamcast, or other SHOUTcast-compatible servers, or to your station's stream replication provider, for distribution to Internet listeners. AAC, AAC-HE, AAC-LD and MP3 formats are all supported, at a wide variety of bit rates. Imagine being able to encode all of your plant's Web audio using a single 2RU hardware appliance! iPort features a built-in Livewire interface for single-cable audio, control and data connection to Axia IP-Audio networks. Don't have an IP-Audio network yet? Just connect iPort via Ethernet to an Axia audio node for standard AES or analog I/O.

OMNIA A/XE

Omnia A/XE is a streaming-based audio processor and encoder which can process audio for a variety of applications, bitrate-reduced and linear. Omnia A/XE is software only, no special cards required. 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. Each program input can be processed and encoded in multiple ways, and sent to multiple servers simultaneously. Processed audio can also be sent to a local sound device for monitoring. Combines genuine Omnia audio processing with the Fraunhofer MP3 and AAC codecs for high quality. High-performance, low memory footprint, native application. 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.


Omnia
192.168.2.199
Fault: None
Config: <none>
Preset: CGSmooth
Pri
Sec
Input
AGC
ST
Wide Band
AGC
Limiters
HD
Analysis
LOUDNESS MONITOR
16 20 25 32 40 50 63 80 100 125
% MODULATION
-14 -12 -10 -8 -6 -4 -2 0 +1 +2 +3
Preset
Wizard
Basic
Lim Thresh
0.0 dB
LO
3 kHz



OMNIA

In railroad lingo, there's a phrase highballing down the mainline which refers to a train that has gained speed, making good time, and won't be denied.

Think of Omnia in the same vein as its iron horsed brethren.

Today, we're not only highballing but working and striving to further improve broadcasting as a medium. The market place has welcomed Omnia.11 and Omnia.9 with open arms, and as of this writing there are over 7000 Omnia. ONE units processing audio all over the world. Going forward, there seems to be no limit, as we're in the midst of more exciting projects, products, and technology endeavors.

We are leading the effort regarding the use of Single Sideband Suppressed Carrier (SSBSC) as an alternative means to transport FM-Stereo. The goal of this is to verify a reduction in perceived multipath on FM. Well, broadcasters report they are able to broadcast their stereo signals with less annoyance due to multipath by using the SSBSC method. This technology is available in both Omnia.11 and Omnia.9. Look for a tech paper on this topic within this NOW! catalog.

Another example of Omnia yet again enabling successful broadcasters to remain leaders in their marketplace.

Frank Foti
CEO, Telos Alliance
Founder and President, Omnia Audio

ART REIS, CHIEF ENGINEER, WPWX CHICAGO

"The overall audio quality is dramatically superior to what he had before. One of the things that the staff noticed right away, was that vocals on songs jump right out better than we have ever heard before."

OMNIA.11

EVERYTHING YOU HEAR IS TRUE

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,

JEAN-PHILIPPE DENAC, PROGRAM DIRECTOR, RFM PARIS, FRANCE

"For my station, I needed to get a sound that is powerful, but unique in that it sets us aside from the competition. Today, RFM has, by far, the best sound of any FM station in Paris, thanks to Omnia.11."

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

ALL YOU CAN IMAGINE. AND MORE.

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 ever offered by Omnia or any competitor anywhere in the world.

There is literally NOTHING else on the market which comes close to the versatility of this processor.

Versatility with no compromise whatsoever on processing power.

The Omnia.9 will simultaneously process FM analog, HD-1 and (optional) HD-2, and HD-3, each independently controlled. Omnia.9 will also simultaneously encode and process the internet streams of FM analog/HD-1, (optional) HD-2 and HD-3 again, each independently controlled.

Omnia.9 features exclusive "Undo" technology: a source declipping algorithm, and program-adaptive multiband expander which removes distortion from source material. This corrects over-processed CDs, so common in today's contemporary music. An onboard, Psychoacoustic Composite Embedder allows 100% audio peaks in stereo (potentially up to 140%), within 100% total modulation. This creates about 3dB extra treble headroom.

Selectable patch points are available for convenient auditioning of the audio signal at any point of the processing chain without affecting listeners. Built-in RDS encoder, dynamically update-able. HTTP push support for automation, such as dynamic RDS and streaming song titles, preset recall. Studio Output with very low latency for talent monitoring.

Another advantage is selectable SSBSC (Single Sideband Suppressed Carrier) technology, standard on Omnia.9. 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

BOB NEWBERRY, CLEAR CHANNEL BIRMINGHAM, ALABAMA MARKET ENGINEERING MANAGER.

"I have never been this excited about an audio processor in quite a while. Words like 'loud' and 'open' might seem mutually exclusive at first but the Omnia.9 achieves this goal splendidly. The fantastic 'de-clipper' that restores the waveforms of badly recorded music is worth the price alone!"

FEATURES AT A GLANCE

» Each processing core is separately fully adjustable and has selectable 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 » Full remote control » On-screen keyboard with several layouts (QWERTY, QWERTZ, AZERTY, Dvorak and ABC sequential) for easy setup and preset name typing » Psychoacoustic distortion-masking composite clipper, enabling a full 3dB of added high-frequency headroom compared to previous processors, for both high fidelity competitively loud FM audio even at 75us pre-emphasis » HTTP push support for automation, such as dynamic RDS and streaming song titles, preset recall » Dayparting (scheduled preset selection) » Composite pass-through (relay bypass) for your backup processor.

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.

MARK WILLIAMES, DIRECTOR OF ENGINEERING FOR MOODY RADIO GROUP

"Several of our stations are processing with the Omnia ONE, which is the processor which I recommend to many of our stations which are not necessarily in big markets, but want to sound like they are."

OMNIA ONE

THE ONE PROCESSOR THAT CAN CHANGE WITH YOUR NEEDS.

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 or Multicast applications. Application can be changed with simple software download, so your investment is ready for present and future utilization. Less than five years after introduction, there are currently over 7000 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.

TOM NELSON, DIRECTOR OF ENGINEERING FOR AMERICAN PUBLIC MEDIA AND MINNESOTA PUBLIC RADIO

"We chose the Omnia ONE Multicast primarily because of how it sounds. It manages low bitrate material beautifully, far better than anything else on the market. That, and Livewire capability makes the Omnia ONE Multicast a no brainer for us."

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.

WPOZ ORLANDO AND OMNIA.11

A WINNING COMBINATION

WPOZ President and CEO Jim Hoge

One of the great success stories of the past year or so has been the phenomenal ratings explosion of WPOZ (http://zradio.org) in Orlando Florida, currently the #1 radio station in the market.

With about 60 signals available in the region, competition is fierce. Therefore, leading stations know how important it is to take nothing for granted, especially when it comes to the overall impact and appeal of their station's audio. Incorrectly processed audio can adversely affect long term listening and jeopardize ratings potential.

As WPOZ President and CEO Jim Hoge points out, an Omnia.11, which they deployed some months ago, dramatically improved the station's dial presence beyond that of any other station in the market at the time:

"When we first fired up the Omnia.11, we were suddenly bigger and brighter than any other FM in the region. The contrast was amazing, absolutely amazing. We quickly established a unique, clear sound that was roaring loud, yet totally distortion free. I have been very impressed."

Mr. Hoge went on to report that the Omnia.11's treatment of the HD-1 has been "equally as impressive. The HD side never sounded anywhere near this good".

So we sat down with Jim Hoge and asked him to expand on his thoughts about the Omnia.11:

WHAT DO YOU LIKE ABOUT THE OMNIA.11 IN TERMS OF AUDIO PERFORMANCE?

The tradeoff with any audio processing is loudness at expense of distortion. The Omnia.11 has broken down this wall where one can dial in the punch and loudness to gather cume yet retain the clarity to keep this new found audience listening longer.

DESCRIBE THE SPECIFIC PROCESSING NEEDS THAT YOU HAVE WHEN PROCESSING THE FORMAT OF WPOZ.

Our genre, AC Christian, has battled poor production for years. The limiter and clipper architecture of the Omnia.11 does help to undo the damage of sloppy recording and poor mastering. However, the biggest problem facing every station in the top markets is PPM encoding. This encoding puts a mid to high frequency buzz in everything. We have found that careful adjusting of the Omnia.11's limiter output mix will alleviate the damage from Arbitron encoding. This is something we have not been able to do with any other processor.

CAN YOU TELL ME HOW OMNIA.11 HAS IMPROVED MUSIC REPRODUCTION AND VOICE?

High-end clarity is the most noticeable improvement. Because of that, we had to go back in and readjust all the mike processing as the talent became much brighter. This is a PD's dream come true (pun intended!)

HOW IS IT AN IMPROVEMENT OVER WHAT YOU WERE PROCESSING WITH BEFORE?

Our previous processor, which is a "latest and greatest" from another company, has been relegated to the "B" chain. No matter which preset we tried, including all the new ones with the new controls, nothing could equal the clarity of the Omnia.11.

HOW DOES WPOZ AUDIO QUALITY AND LOUDNESS STACK UP AGAINST THE OTHER STATIONS IN THE MARKET

Let's see. We installed the Omnia.11's in January 2011 and, since then, have been either number one or two in the ratings monthlies ever since. We also noticed a couple of unbiased, positive comments posted on the Orlando message board on the website radio-info.com.

WOULD YOU RECOMMEND THE OMNIA.11 TO OTHER STATIONS? (NOT COMPETITION, OF COURSE!)

Absolutely, and we have told many friends around the country of our great success with the Omnia.11. And, yes, at least one of our competitors did find out what we are using and installed an Omnia.11 here in Orlando right away!

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.

WOLF KOYGYN, WOLFONTHENET.COM/KHWL

"The Omnia A/XE has breathed new life into our alternative rock streaming service. I bought one of those familiar 'radio station in a box' automation solutions, but the built-in processing was pretty lame. So, I bypassed the processing and encoding and popped in the Omnia A/XE for those portions of the chain. Wow. I just can't get over the improvement in sound quality."

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).*

KENT HATFIELD, VP TECHNOLOGY AND OPERATIONS WXXI ROCHESTER, NY

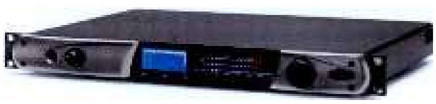
"We were very pleased with the results of the Omnias in the first two streams, so we went with an Omnia ONE for the third. The audio quality is great, and the units have been very reliable and trouble-free. Just plug them in, set them up and go."

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.

GOOM RADIO

74 STREAMS POWERED BY OMNIA ONE MULTICAST AND A/XE.

GOOM Radio is one of the world's most popular streaming services, with 7.2 million regular listeners each week tuning into seventy-four different programming choices from studios in Paris and New York City.

Omnia ONE Multicast and Omnia AX/E are used exclusively to process every one of the fifty-three GOOM programs for France, and the twenty-one GOOM Programs for the US in MP3 and HE AAC stream formats.

GOOM's audio consultant David Perreau explains why Omnia ONE Multicast and Omnia A/XE were the choices across the board:

"The main issues we can have with low bitrates when you broadcast with MP3 or MP4, is the fact that if your processor works more on the density aspect to create loudness, you will hear more artifacts on the program material. Ordinary processors have this problem.

With the Omnia ONE — and its exclusive SENSUS™ algorithm designed especially for lower bitrates — the processor works more on the dynamics aspect to create loudness, to control the output of the audio stage and to increase the punch and the dynamics of our sound. The final result sounds punchy, with no artifacts. We actually use the high quality of our streams as a selling point because they sound so good.

We are always hearing compliments about the quality of our streams from our audience."

SO, WHAT ABOUT SETUP?

Omnia ONE Multicast was very easy to set up. We started with a factory preset and then — in some cases depending on the format of the stream — we adjusted the processor. The only limit is where you want to stop and the loudness level you want to create. When you have a facility like GOOM with so many processors to set up, Omnia ONE is great because storing and uploading the presets is very easy."

David concluded: "I have never had a single problem with any of the Omnia ONEs or Omnia A/XEs that we have at GOOM Paris and GOOM New York City."

The high quality streams of GOOM — powered by Omnia ONE Multicast and Omnia A/XE can be accessed at:

GOOMRADIO.US (NEW YORK)
GOOMRADIO.FR (PARIS)

And don't forget to check out audio consultant David Perreau's site at: **WWW.YAKUDAAUDIO.COM**

DAVID PERREAU, AUDIO CONSULTANT, YAKUDA AUDIO

OMNIA.SG

PUT YOUR STEREO GENERATOR WHERE YOU WANT IT.

Omnia.SG lets you keep the on-air processor at the studio, where it's convenient, while keeping stereo generation at the transmitter, where it belongs.

Designed to compliment your existing audio processing, Omnia.SG is modeled after the same stereo generator used in Omnia audio processors. Omnia.SG provides stereo generation and includes a selectable composite clipper and base-band filter for crisp, clean loudness.

It's the best of both worlds: clear, digital stereo generation and the convenience of locating your processing at the studio.

OMNIA.SG WITH CLOCK SYNC

As an option, Omnia.SG can now be ordered with a built-in GPS clock sync, designed to assist with the synchronization of stations that are on the same frequency, or on closely-spaced adjacent frequencies.

This synchronization is achieved through the use of a GPS receiver with a 10Mhz clock input, easily deployed at each station's transmitter site.

The GPS clock sync is used, for example, when a station puts a booster on the same frequency to fill behind a mountain or tall buildings which block the main transmitter, but is in the primary service area. Unless the stereo encoders are precisely synchronized at both transmitter sites, there could be distortion, noise, and the stereo pilot detector may switch back and forth between the two stations noisily, along with other potentially annoying behavior when a listener is in a spot where both signals are received equally. The Omnia.SG with clock sync option eliminates this problem by synchronizing the sites precisely via GPS.

Omnia.SG users can update to include the GPS clock sync option. Contact Omnia Audio for details.

SSBSC

FM-STEREO TRANSMISSION USING SINGLE SIDEBAND SUPPRESSED CARRIER (SSBSC) MODULATION

ABSTRACT

FM-Stereo transmission, employed in worldwide broadcasting, has been in place since 1961. The system uses double sideband suppressed carrier (DSBSC), within the multiplex baseband signal, as means to transport the stereo sound field to the receiver. This method, while robust and reliable, is prone to the effects of multipath. This paper will discuss an optional method utilizing single sideband suppressed carrier (SSBSC) modulation as an alternative for broadcasters. SSBSC is backward compatible with existing radio receivers. Benefits which are perceptible to a listener include: a reduction in multipath induced distortion, additional protection to spectrum used for RBDS, SCA, and HD Radio® signals. There is an additional, separate benefit in the receiver which improves the signal-to-noise ratio when SSBSC is transmitted and the receiver is designed to capture the SSBSC signal. SSBSC for FM-Stereo has been deployed recently, under experimental authorization from the FCC, with ongoing testing in the lab, and in the field.

COMPETING FOR EVERY POSSIBLE LISTENER

FM radio has a good fight on its hands. As a media transom to the public, it battles a multitude of additional delivery services like never before. Until recently, the competition was from television, phonograph records, compact discs, or tape. Actually FM-Stereo has outlived numerous media forms such as the long playing record (LP), cassette and 8-track tape, mini-disc, soon the compact disc (CD), as well as a few others. Now, with the advent of good quality portable audio playback devices, and wireless streaming, there are many additional franchises available to steal the listener away from radio. What can FM radio do, technically, to improve sonic performance so a listener has less reason to abandon it as an outlet?

HD Radio® was introduced to the marketplace within the last ten years, and it's still trying to make an impact on the casual listener. What's needed, in the meantime, is an improvement to the existing infrastructure, which does not require any change or added expense to the listener. Within present day radio listening, FM is still the preferred choice. Recent technical research and development unveiled a unique way to improve the performance of FM-Stereo. What follows is the result of those efforts, along with a recommendation for FM broadcasting.

SILVER ANNIVERSARY OF FM-STEREO

In April of 2011, it marked 50 years since the Federal Communications Commission (FCC) approved FM stereophonic transmission in the United States. The Commission, after evaluating fourteen proposals, decided upon a system that was of similar design from both Zenith and General Electric.

The rules governing stereophonic performance have not been altered since the mid 1980's (in the USA) when they were modified to allow an additional 0.5% total modulation (maximum of 110% total), for every 1% of SCA modulation, when an SCA was being utilized. The rules governing the requirements of the FM-Stereo baseband signal are quite explicit, and leave little room for improvement of the stereo transmission system.

A quick refresher course on FM-Stereo transmission, courtesy of subpart 73, of the FCC Rules and Regulations:

§ 73.322 FM stereophonic sound transmission standards.

a. An FM broadcast station shall not use 19kHz +/-20Hz, except as the stereophonic pilot frequency in a transmission system meeting the following parameters:

   1. The modulating signal for the main channel consists of the sum of the right and left signals.
   2. The pilot subcarrier at 19kHz +/-2Hz, must frequency modulate the main carrier between the limits of 8 and 10 percent.
   3. One stereophonic subcarrier must be the second harmonic of the pilot subcarrier (i.e., 38kHz) and must cross the time axis with a positive slope simultaneously with each crossing of the time axis by the pilot subcarrier. Additional stereophonic subcarriers are not precluded.
   4. Double sideband, suppressed-carrier, amplitude modulation of the stereophonic subcarrier at 38kHz must be used.
   5. The stereophonic subcarrier at 38kHz must be suppressed to a level less than 1% modulation of the main carrier.
   6. The modulating signal for the required stereophonic subcarrier must be equal to the difference of the left and right signals.
   7. The following modulation levels apply:
      i. When a signal exists in only one channel of a two channel (biphonic) sound transmission, modulation of the carrier by audio components within the baseband range of 50Hz to 15kHz shall not exceed 45% and modulation of the carrier by the sum of the amplitude modulated subcarrier in the baseband range of 23kHz to 53kHz shall not exceed 45%.

ii. When a signal exists in only one channel of a stereophonic sound transmission having more than one stereophonic subcarrier in the baseband, the modulation of the carrier by audio components within the audio baseband range of 23kHz to 99kHz shall not exceed 53% with total modulation not to exceed 90%.

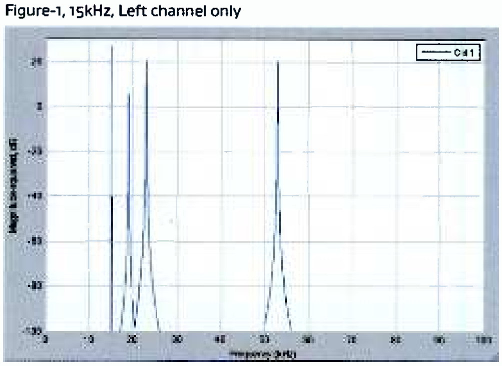
The FM-Stereo system, as described above, has worked quite well for 50 years, but not without challenges. Most notable is multipath distortion, especially in areas of congested buildings, hills, and/or mountainous terrain. Also, radio broadcasters have added incremental signals within the multiplexed spectra. Radio Data Services (RBDS) based at 57kHz, as well as a 92kHz SCA can additionally occupy the signal. The modulation index of the FM carrier is further reduced with each and every added signal, thus increasing the sensitivity of multipath distortion in the receiver.

Since the inception of stereophonic broadcasting, there has been no technical change to the infrastructure of the Zenith/GE system at all. The FCC rules are quite specific regarding the multiplex spectrum, and its interoperability as a system. Considering the above mentioned challenges, and the alternatives a listener now has, it makes practical, as well as good business sense to investigate improvements to the present system. It stands to reason that any means proposed must be backward compatible with existing stereo receivers. After 50 years of marriage to FM, a silver anniversary present seems to be in order!

TECHNICAL CHALLENGES FOR FM-STEREO

Multipath is easily the largest annoyance to a radio listener. Broadcast markets located in cities with many large downtown buildings, and/or mountainous terrain, suffer even more on account of it. Increased multipath is a direct result of low modulation index within the FM carrier. As more spectra is utilized within the multiplex signal, the index of the carrier is reduced. The following condition creates maximum stress of the FM-Stereo system, and generates the lowest modulation index: A single audio channel, either Left or Right only, utilizes the most amount of spectra within the system. By example, a 15kHz tone in the left channel only, will produce multiplex spectra at 15kHz, 19kHz (stereo pilot tone), 23kHz, and 53kHz. Each of these signals will reduce the modulation index to its smallest level, which increases sensitivity to multipath in the receiver. Figure–1 is an illustration of this.

Figure-1, 15kHz, Left channel only



Note the 30kHz difference in the L-R subcarrier of the two sidebands located at 23kHz and 53kHz. This is generated by the DS-BSC process of (38kHz – 15kHz) for the lower sideband at 23kHz, and (38kHz + 15kHz) for the upper sideband at 53kHz. During an instance of multipath, as the multiple reflections of the FM carrier arrive at, and then become demodulated in the receiver, the time delay difference created by the multiple carrier reflections will offset the phase of the upper and lower sidebands. During the demodulation process and decoding, stereo separation at these frequencies is reduced, along with generated distortion, as the recovered L-R level is negatively altered due to phase shift brought on by multipath.

Bandwidth of the conventional analog FM channel is allocated for 99kHz of spectrum use. The FM-Stereo system requires 53kHz (0Hz – 53kHz) of this available real estate. The remaining 46kHz (53kHz – 99kHz) is used for RBDS and SCA services. Common practice requires the use of audio processing to insure proper peak level and bandwidth control of the various signals present in the multiplex spectrum. Current generation processors are capable of creating near-theoretical multiplex signals. In these cases, there are little, if any, transmission difficulties for the signal.

Some broadcasters choose to employ a form of processing known as composite clipping. This technique inserts a hard limiter (clipper) at the output of the stereo baseband generator, and will induce up to as much as 3dB of clipping to the multiplex signal. These devices provide no additional filtering to remove unwanted harmonic content from the clipping process. The additional harmonics will cover the entire 46kHz, and beyond, used for RBDS and SCA services. This creates interference and distortion to those signals. Also, these harmonics may interfere with the digital carriers generated for HD Radio, as these carriers are set 120kHz out from the main channel carrier.

Alternate Approach: Single Sideband Suppressed Carrier (SSBSC)

What if we were to take the FM multiplex signal that looks like the one in Figure-2 . . .and transmit it like the one in Figure-3.

Figure-2

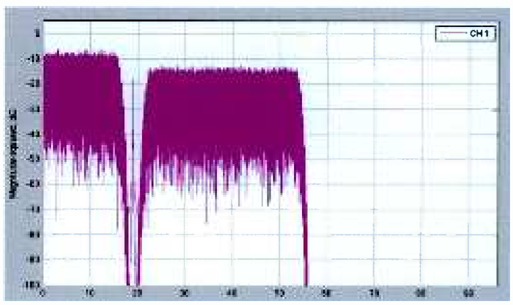
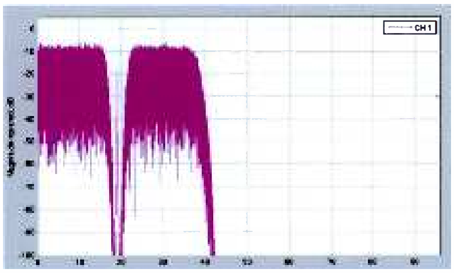


Figure-3



An alternative approach for stereo transmission would be the use of single sideband suppressed carrier (SSBSC) as the mechanism to carry to the L-R payload. The lower sideband is chosen as it reduces the occupied spectrum from 53kHz down to 38kHz. In order to support the correct L+R/L-R matrixing in the receiver, the amplitude of the lower sideband is increased by 6dB. This offers numerous benefits to the receiver:

1. Reduction of occupied bandwidth in the L-R subchannel range increases the FM modulation index by a factor of two. This directly reduces multipath distortion.
2. Narrows the overall FM transmission bandwidth and reduces degradation of stereo performance caused by finite bandwidth of passband filters, cavities, multiplexing systems, and antennas. If adopted internationally, this further benefits broadcasters where 100kHz channel spacing is used in some countries, as compared to the 200kHz spacing used here in the USA.
3. Creates additional and significant protection for RBDS, SCA, and HD Radio signals. Note: With the HD Radio power increase, reduction of the composite spectrum benefits conventional receivers due to less demodulation overlap of the HD Radio signal.
4. Backward compatible with all existing modulation monitoring systems.
5. Backward compatible with conventional receivers.
6. Less harmonic content generated throughout the channel spectrum when composite clipping is employed in the transmission audio processor.
7. Improvement of demodulated signal-to-noise (SNR) by 4dB in receiver, when SSB is transmit, and multiplex decoder is of SSB design.

The concept of utilizing SSB modulation for the L-R payload is not without precedent. A white paper [1] on SSBSC transmission was presented by William Gillman at the 1997 NAB Engineering Conference. Reviewing Mr. Gillman's paper, and subsequent testing by this author, confirms his findings. Since 1997, when Mr. Gillman's paper was published, technological advances in transmission firmware makes this concept much more plausible.

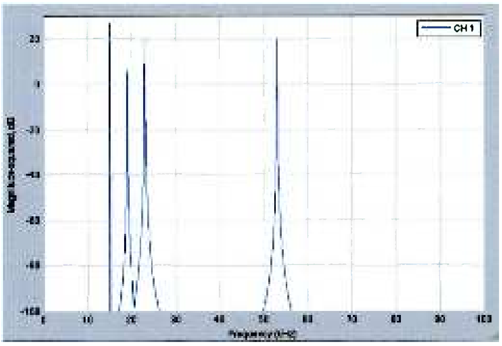
In researching SSBSC for this paper, it was a considered method during development and testing of FM-Stereo during the late 1950's. A possible reason why DSBSC was chosen over SSBSC was in part due to the complexity involved to design and build SSBSC circuitry, reliably, in the world of analog electronics. Even though SSB technology has been utilized considerably in communications overall, it does require additional technical attention, when deployed in the analog realm. There was some work done on this topic during the mid 1980's in New York City [2]. It appears that effort encountered various challenges due to the state of analog technology available at that time.

Today, SSBSC generation, and decoding is easily accomplished reliably, with digital designs that are possible on numerous platforms. Prior to advances in algorithm development, and firmware, SSBSC – while possible – was not an easy implementation. Hence the reason it's been awhile, since 1997, before the concept is capable of coming to life.

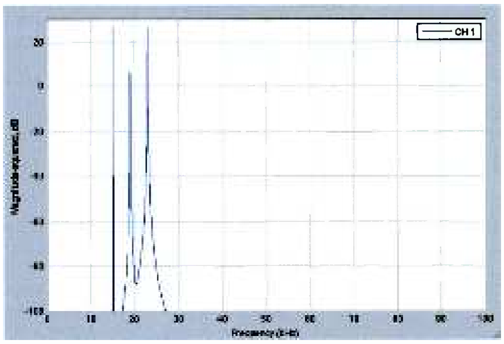
TECH FINDINGS

Implementing SSB modulation of the L-R signal is relatively easy to accomplish using DSP. Figure–4 is a spectral diagram of a 15kHz single channel tone using DSBSC system. Figure–5 is the same condition, except SSBSC modulation is utilized.

Figure–4, 15kHz tone, single channel, DSBSC



Figure–5, 15kHz tone, single channel, SSBSC



Note the 6dB increase in level of the SSB carrier in Figure–3. This illustrates the manner in which the L+R/L-R mathematics are upheld, when decoded in the receiver.

Easy to observe the significant difference in spectrum used. The DSBSC method forces the single channel condition of 15kHz to exist across a broad range. The fundamental is at 15kHz, and the two sidebands are at 23kHz and 53kHz respectively. The DSBSC example illustrates the fragility in faithful reproduction of stereophonic high frequencies during instances of multipath. The group delay at 15kHz, 23kHz, and 53kHz becomes non-linear, during multipath, and this is why stereophonic high frequencies are so fragile, and easily distort, during multipath.

Compare the spectra of Figure–4 with that of Figure–5. The close proximity of the 15kHz fundamental and the 23kHz SSB carrier improves high frequency robustness during multipath. Due to the closeness of these two frequencies, there is less adverse affect when multipath non-linearity is in existence, thereby high frequency stereophonic performance is audibly improved.

RF CHANNEL OCCUPANCY

SSB subchannel modulation enables efficient FM channel occupancy. The following examples illustrate FM deviation at +/75kHz. Using the Bessel null method, 31,189.4Hz creates the first carrier null, when +/-75kHz deviation is achieved. As a reference level, Figure-8 illustrates the deviated RF spectrum, and Figure-9 the input peak level at the modulator.

Figure-8, +/-75kHz Deviation, 31,189.4Hz

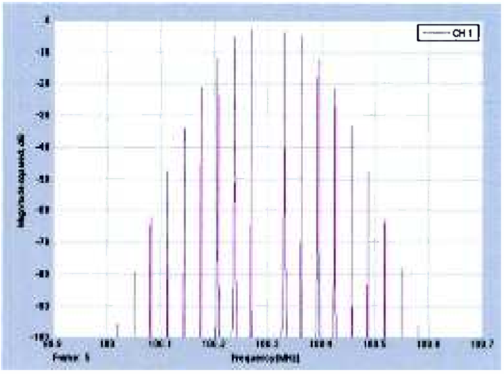
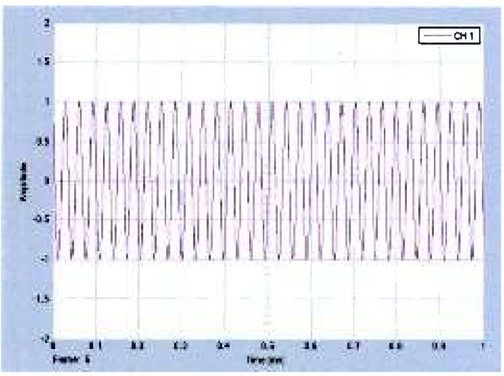


Figure-9, Peak Level, 31,189.4Hz



A DSBSC composite signal comprising 15kHz in a single channel, 19kHz pilot, lower/upper sidebands of 23kHz and 53kHz are depicted in Figure-10 showing RF deviation at +/-75kHz, and Figure-11 showing peak modulation level.

Figure-10, DSBSC, 15kHz L-Ch, +/-75kHz Dev

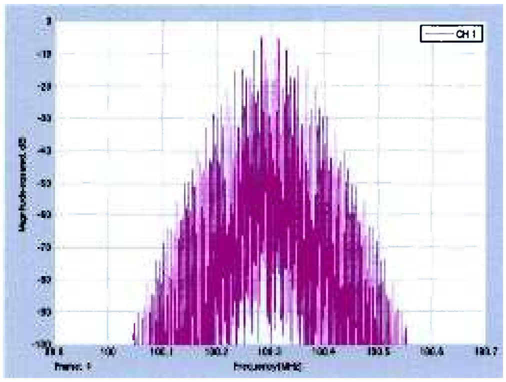
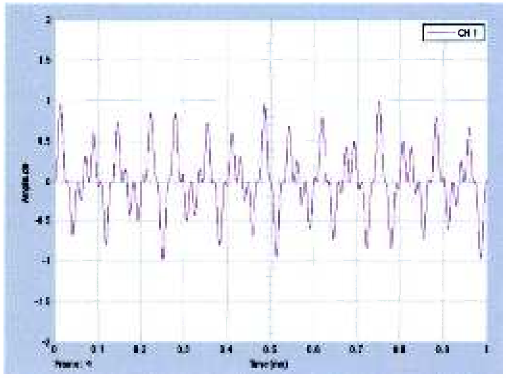


Figure-11, Peak Level, 15kHz, L-Ch



Next, a SSBSC composite signal comprising 15kHz in a single channel, 19kHz pilot, and lower sideband of 23kHz are depicted in Figure-12 showing RF deviation at +/-75kHz, and Figure-13 showing peak modulation level.

Figure-12, SSBSC, 15kHz L-Ch, +/-75kHz Dev

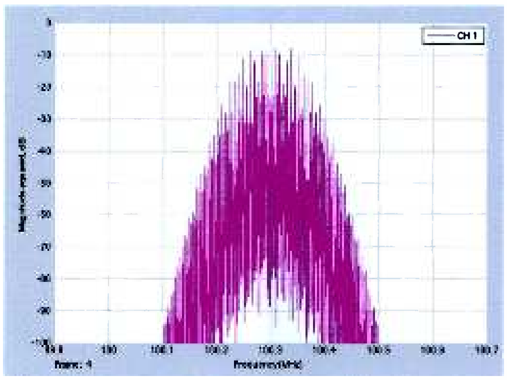
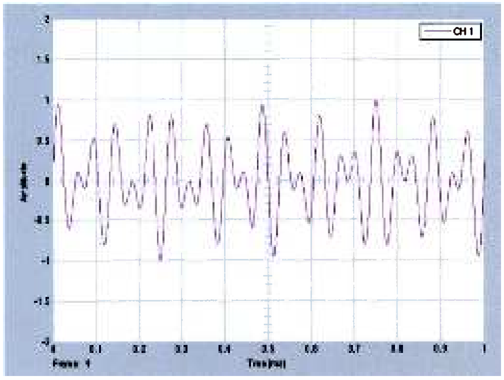


Figure-13, Peak Level, 15kHz, L-Ch



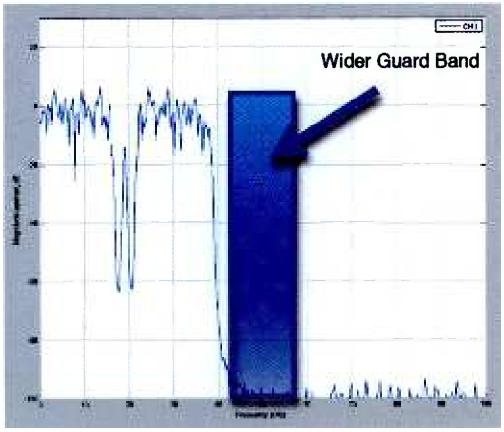
Notice the reduction in utilized RF spectrum. The signal shown in Figure–12 will pass through narrow cavities, combiners, and mal-adjusted antennas with better stereo performance than the broader signal shown in Figure–10.

Additionally, there are less sideband pairs of the carrier signal. Less sideband pairs equates to less signals, which can be interfered with during instances of multipath.

In the United States, FM channel spacing is maintained at 200kHz. This is a bit of a luxury compared to the rest of the world where channel spacing is usually 100kHz. Consider the probability of less channel-to-channel interference when SSBSC transmission could be used. This would appear to be an improved alternative to the ITU BS-412 MPX power regulation, which is now in force in some European countries.

Another mentioned benefit SSB brings to the transmission method is added spectral protection to RBDS, SCA, and HD Radio services as observed in Figure–14. Single channel only pink noise is used to generate the baseband signal with SSBSC modulation. Notice the extremely wide guard band that exists between 38kHz and where the first SCA carrier would appear at 57kHz. The reduction in cross-talk to ancillary services is exceptional!

Figure-14, Reduction in Cross-Talk to Ancillary Services.

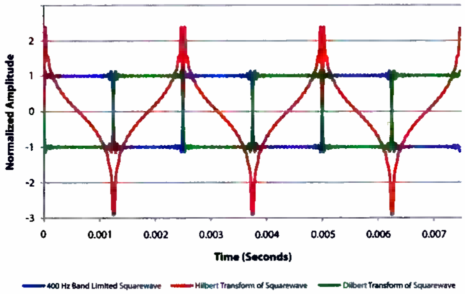


SSBSC AND MODULATION PEAK CONTROL

Implementing SSB can be accomplished using numerous techniques. The most common method is through use of the Hilbert function, where a 90 degree broadband phase shift is used to cancel the undesired sideband. It can also be achieved using a Weaver modulator, or a low pass filter set to critically limit the desired passband, and the undesired sideband is removed through filtering. All of these methods provide satisfactory SSB operation, but there is a critical element that must be considered. . .peak control of the overall MPX signal. In each of the afore-mentioned SSB methods, there will be alteration to the phase relationship of the sideband signal. This alone will generate overshoot to MPX encoded signal [3]. It is paramount that SSB modulation must not add any overshoot to the signal, and it must not add any unwanted non-linear components, in the form of audible overshoot peak limited harmonic content, i.e. clipping by-products. The sonic performance of the SSBSC modulator must perform sonically, exactly the same as the DSBSC counterpart. Switching from DSBSC mode to SSBSC should not change the resulting sound in stereo separation, audio quality, and peak control.

Theory indicates that a 90 degree phase network, in the form of a Hilbert filter will cause overshoot to a square wave. What's interesting is by adding a second Hilbert function, the overshoot is removed, and the square wave is recovered. Use of the double Hilbert function has been referred to as the "Dilbert" function [4]. An example of this is provided in Figure-15.

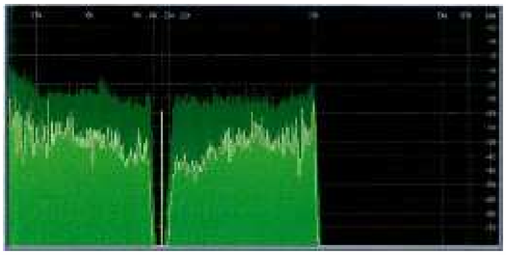
Figure-15, Hilbert Affect on Square Waves



Just as an audio processor is known to employ non-audible methods to eliminate peak overshoots in the required 15kHz low pass filters, there are non-audible algorithms employed in the SSBSC generator which insures that any overshoots are eliminated, and done so without any sonic change or affect to audio quality. Figure-16 is a screen capture from a digital oscilloscope that was measuring real world MPX signal at the output of an audio processor. Figure-17 is the spectral reproduction of the same signal. Note exceptional peak control along with the well maintained spectrum around the 19kHz pilot, and the sharp drop off after 38kHz.

Figure-16, MPX Peak Control, SSBSC

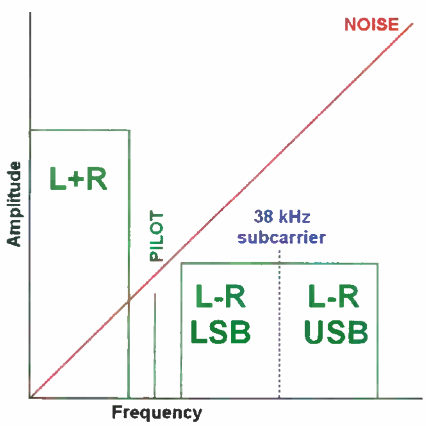
Figure-17, MPX Spectra, SSBSC



SSBSC AND DECODED SIGNAL-TO-NOISE

Another known challenge for the system is the compromised signal-to-noise (SNR) level when broadcasting stereo. FM transmission noise will rise in a triangular fashion at 6dB per octave over the channel's passband range of 0Hz-99kHz. This is the product of the modulation/demodulation process. The use of preemphasis in transmission, along with complementary deemphasis in demodulation, improves the high frequency noise response. It has been generally accepted that FM-Stereo suffers a 23dB overall noise penalty compared to monophonic broadcasting. This is due to the rising noise floor over the subcarrier range of 23kHz – 53kHz, as compared to the SNR over the range of 0Hz – 15kHz, which is used for mono. Figure–18 is an illustration of the composite baseband signal, and it shows the 6dB/octave noise floor slope of an FM channel, as it would appear at the output of an IF section in a receiver.

Figure-18, FM System Noise Plot



It has been theoretically calculated [5], and technically demonstrated, there is roughly a 4dB broadband improvement in recovered signal-to-noise performance of the SSBSC transmit/receive function, as compared to the conventional DSB transmit/receive iteration. When transmitting SSBSC, and decoding only the lower sideband spectra (23kHz – 38kHz), an interesting event occurs. Stereophonic noise is about 10dB better for decoded 15kHz. This is due to the frequency inversion of the lower sideband. The triangular noise is lower at 23kHz, where 15kHz resides in the lower sideband region of the L-R signal, as compared to lower frequencies, which are located near 38kHz and triangular noise is greater. Figure-19 is the recovered noise floor of a DSBSC transmission/reception. Compare the amplitude of the noise floor at 15kHz in this figure with that of Figure-20, which is the recovered noise floor of a SSBSC transmit/receive system.

Figure-19, Recovered Noise, DSB

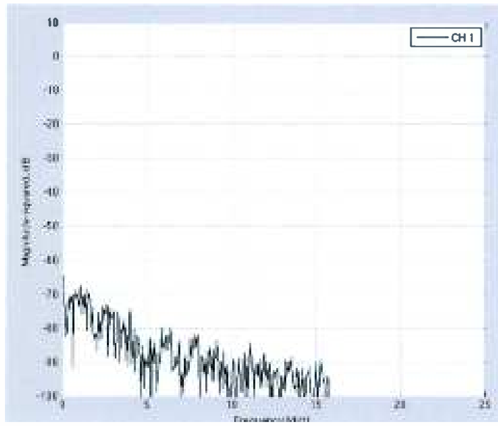
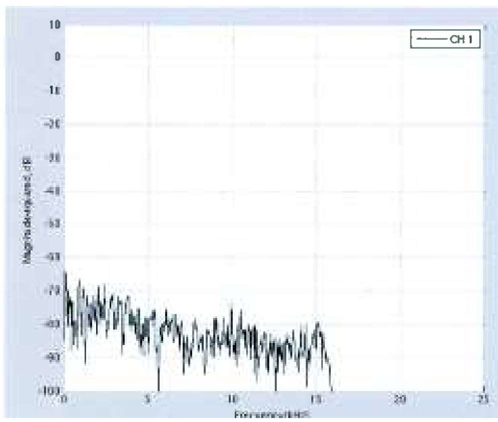


Figure-20, Recovered Noise, SSB



Consider the annoying hiss a listener hears at the output of the FM receiver. The predominant range of audible hiss is the high frequencies. As observed in Figure-18, there's an improvement of 10dB in signal-to-noise in the audible hiss range. Hopefully, this might encourage receiver manufacturers to consider adding SSB decoding into conventional receivers. The above test results were realized using a SSBSC stereo decoder designed and implemented, real time, in MatLab by the author.

REAL WORLD ACTIVITY: IN THE FIELD, AND IN THE LAB

Transmitting SSBSC modulation of the FM-Stereo signal can be done right now! Software exists to implement this method today. One minor item must be addressed: FCC rule 73.322, section (A), subpart (4) which states "Double sideband, suppressed-carrier, amplitude modulation of the stereophonic subcarrier at 38 kHz must be used." Seems there was a time, when rule 73.322(A)(4) was required. Times have changed. Both transmission and reception firmware have improved significantly to enable a change in the rules and regulations governing FM-Stereo, at least to allow the use of SSBSC as an option for the broadcaster.

At present, based on the theory, testing, and findings presented here, the FCC allows Experimental Authority (EA) operation, which enables broadcasters to implement the SSBSC transmission method. Benefit occurs immediately to those whom employ SSBSC, especially those in areas of rough terrain with significant hills, and mountains. As of this writing, SSBSC is on-the-air in multiple major markets, and all users report a reduction in perceived multipath. The general consensus is how a mobile receiver operates less in the blend function. As the radio comes out of blend, when SSBSC is used, the appearance of added high frequency content is perceived. Many radios reduce the high frequency range, along with blending stereo separation during instances of multipath. In some cases the change is quite noticeable, and in others it has been observed to be a small improvement. It should be noted that severe multipath will cause annoyance to either form of transmission: DSBSC and SSBSC.

While most feedback is of the subjective anecdotal variety, there has been some initial lab testing done to determine, at the very least, if SSBSC offers any degradation to FM service. Using a multipath generator, that offered repeatable multipath profiles in a controlled environment, it was possible to gather data from a receiver operating under an impaired signal. Testing was done with DSBSC and repeated for SSBSC. A simple test of transmitting a 1kHz tone in a single channel, and then monitoring the recovered Left/Right channels in a mobile receiver would indicate any degradation between DSBSC and SSBSC. The test was done over a twenty-four second period. Figure-21 illustrates the plot of the transmit 1kHz tone in the Left channel, along with any crosstalk that spilled over into the Right channel due to hits of multipath. The multipath instances can be observed as the sections of the Left channel where the signal degrades. Figure-22 is the result of the same test done in SSBSC mode.

Figure-21, DSBSC

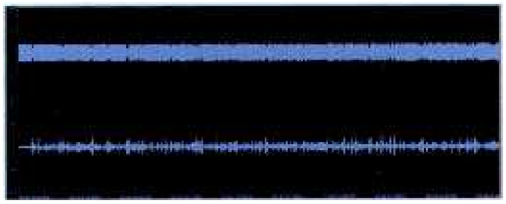
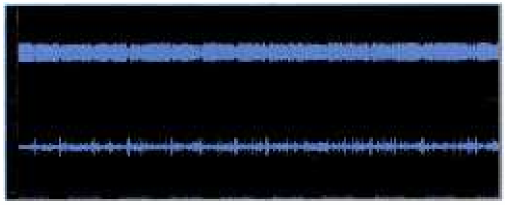


Figure-22, SSBSC



Notice there is virtually no difference between the two plots. Had multipath distortion been more severe for either mode, the amount of crosstalk into the Right channel would have increased. This test therefore indicates that SSBSC offers no perceivable degradation to the FM service signal.

TAKING IT TO THE NEXT LEVEL...

In addition to those broadcasters whom are using SSBSC under an EA from the FCC, there is continued testing being done in the lab. The topic is also an active action item within the AM FM Audio Broadcast (AFAB) sub-group of the National Radio Systems Committee (NRSC). As with any consideration to possibly change the rules, testing, data gathering, and system evaluation must be done. Additionally, viability must be shown to indicate public benefit.

To this extent, criteria has been brought forward to propose tests which would help answer questions regarding the feasibility of SSBSC as an optional transmission method to the present means. What follows is the body from a paper offered by John Kean, of NPR Labs.

"Conversion to a single-sideband suppressed carrier stereo subchannel for FM broadcasting represents a technical change in terms of FCC rules that is sufficient to require thorough documentation in the public record. Indeed, comments filed recently with the NRSC suggest that while a SSBSC system may offer benefits, such as reduced noise and interference to IBOC digital sidebands, the system also may increase FM audio distortion under multipath reception conditions [2][3]. These potential issues should be evaluated objectively and made available to the radio industry through the NRSC. This paper discusses a suggested approach for tests that can determine the compatibility of SSBSC, as well as potential improvements offered by SSBSC.

Evaluation of a new transmission standard may be considered in three main areas:

- » Receiving compatibility with the host station's signal
- » Potential for reception enhancements
- » Effect on stations on adjacent frequencies (allocation compatibility)

The first area, compatibility with the host, may be considered for the following:

- Stereo FM
- SCA subcarriers (including digital SCAs)
- RBDS digital subcarriers
- Extended Hybrid IBOC sidebands

SSBSC generates modulation peak overshoots and increased sideband amplitudes, at least theoretically, which may increase audio distortion of the demodulated FM signal under multipath reception conditions. It is essential, then, to test the above transmission modes with multipath propagation. NPR Labs has worked extensively with both over-the-air and laboratory-simulated multipath; in our experience, laboratory simulated multipath can be made indistinguishable from over-the-air multipath conditions, and they avoid the signal instability, environmental noise and signal interference that hinder the accurate comparisons. These other degradations can be added in controlled amounts to the receiver under test, if desired, although they do not appear to be necessary for this testing.

The difficulty with fixed multipath scenarios, whether over-the-air (stationary) or simulated, is that they represent only one condition, requiring measurements or audio sample recordings with many separate amplitudes and phases of the "paths" to represent the scenario. NPR Labs has been successful in putting the scenario "into motion," causing the scenario to pass through many combinations of amplitudes and phases within one time interval of the multipath simulator. Multipath profiles should include an urban condition (short path delays with higher amplitudes), rural (longer path delays with lower amplitudes relative to the direct path) and a no-multipath condition.

The time interval of the multipath simulator, including multipath fading, can provide an audio sample for assessment by listeners in a controlled subjective test. Listeners provide the basis for fair and understandable ratings of reception quality. NPR Labs has also used Fast Fourier Transform analysis to produce frequency distribution histograms from digital (wave file) recordings of the multipath interval, thereby providing an objective measure of the distortion products. Either method is appropriate to this study: the listener-based tests are more expensive but simpler to interpret, while the FFT analysis is faster and permits more conditions to be tested.

It is important to test reception compatibility with a variety of receivers, as the impacts may vary with the internal architecture and performance of the receiver. A test matrix involving the quality rating with different multipath profiles for each receiver would be an appropriate output to demonstrate the levels of compatibility. The matrix could include other processing conditions for the SSBSC transmission as well.

Testing compatibility with RBDS and IBOC DAB are simpler since the failure of digital reception can be used to determine the potential impact of SSBSC transmission. It is possible that severe degradation of analog FM stereo occurs before failure of digital reception. This would simplify the extent of these tests.

Coverage enhancement is a simpler, and optional, consideration. The improvements could be determined by changes in audio signal-to-noise ratio with stereo FM receivers equipped with suitable SSBSC decoders. NPR Labs' standard approach uses a frequency-weighted quasi-peak psophometer, compliant with ITU-R Recommendation 468-1, which correlates well with listener's assessments of noise-limited reception. A more comprehensive test would include multipath reception conditions, to ensure that the potential improvements are not degraded by multipath propagation effects.

The test of effects to reception on first and second-adjacent channels is conducted similar to the above: WQPSNR is measured by psophometer as the ratio of undesired (SSBSC) carrier to desired carrier is varied. The RF protection ratio at which the same WQPSNR is achieved with SSBSC, relative to standard DSB-SC, is noted. The undesired carrier should be modulated by an audio program signal, or simulated program signal, that represents the RF spectral occupancy of typical FM stations. More than one program modulation could be considered, such as high-density music and low-density music. The test matrix would tabulate the change in RF protection ratios against a variety of receiver types. Again, a more comprehensive test should introduce multipath profiles to the matrix to ensure that multipath propagation does not increase the RF protection ratio" [6].

As of this writing, the author is in the process of assembling a proposal for the AFAB subgroup of the NRSC that will propose formalized industry testing of the SSBSC transmission method.

IN CLOSING...

An opportunity presents itself to our industry. The chance to improve the sonic performance of conventional FM-Stereo radio. Even if a subtle improvement, through reduction of perceived multipath, offers the possibility of people listening longer to FM radio, everyone gains. Together, equipment designers, receiver manufacturers and broadcasters can work together to further investigate the viability of SSBSC as an optional transmission method. Thus far, the initial results look very positive, based on feedback from broadcasters. It is possible that some hurdle exists, and hopefully through joint, mutual effort of our industry, we'll be able to determine what to do, should that be the case. It must be noted there is an extremely large and positive interest in this topic. Should the reader desire to become involved, please contact this author, or a member of the AFAB subgroup of the NRSC.

After 50 years of stellar operation, a modification to the rules and regulations governing FM-Stereo, would be a wonderful way to celebrate this technology! More importantly, the benefactors are the general public-radio listeners, as audible annoyances will be suppressed, and in some cases, eliminated. At a time when broadcasters are looking to find every possible way to enhance their customers (the listener's) experience, this change in the rules would benefit everyone. This concept offers total upside, with – as of yet -no downside at all.

For references and acknowledgements see:
omniaaudio.com/downloads/white-papers/MPX-SSB-White-Paper.pdf

AXIA

Ever since Axia introduced broadcasters to the world of studio IP-Audio, we've been having a blast. Early on, there were future-minded broadcasters who got it right away. Others took some time to appreciate the power and flexibility of networked audio. Now of course, the technology has become so successful, it's hard to find anyone not planning or installing AoIP studios.

It's mainstream tech now, and other companies have hopped on the bandwagon, but Axia remains the most popular approach to IP-Audio by a wide margin: there are more than 3000 Axia consoles and 25,000 Livewire equipped broadcast devices in service around the world every day. Maybe it's because we invented the technology. Or maybe it's because we keep inventing cool things.

Take, for instance, our new xNodes: compact second-generation IP-Audio interfaces that fit in half the space of first generation nodes. With dual redundant Ethernet ports and power sources. They can operate on Power over Ethernet links, sync to IEEE 1588 clocks, and have audio performance specs that would make recording engineers jealous.

We also keep expanding our selection of mixing consoles, like the modular Element (the most popular IP console in the world), the full featured mid-size iQ and Radius, and the new, compact, fits anywhere RAQ and DESQ 6 fader consoles.

We've taken the hassle out of IP-Audio by putting a zero-configuration, built-for-broadcast network switch in our PowerStation and QOR integrated console engines. These devices speed IP console installation by aggregating the console CPU, power supply, DSP mixing engine and loads of audio and logic I/O into a single, fanless rack unit that can be quickly deployed anywhere.

There's IP Intercom, the only intercom system for broadcast that uses IP-Audio to easily connect intercom stations around your plant and connect those stations directly to your on air console, allowing talent to take broadcast quality intercom audio to air at a moment's notice.

There are new products from Telos that connect to Livewire networks, like the VX broadcast VoIP talkshow system and the iQ6 six-line system that integrate into Axia consoles using just one CAT-5 cable for audio, control and mix-minus. And ProSTREAM, the networked streaming appliance that can process and encode audio directly from your network, then send it to your streaming provider for delivery to thousands of Internet listeners.

And we've opened our technology with the ground breaking Livewire Limited License, sharing Livewire's inner workings with a host of broadcast technology companies eager to include Livewire connections on their own broadcast hardware and software products.

As you're planning studio upgrades in the future, remember: Axia offers the most complete line of products, the most advanced technology, the most compatible partners, and is backed by the largest and most driven group of AoIP fanatics you'll find anywhere.

Michael "Catfish" Dosch
President, Axia Audio

ELEMENT

CAN A RADIO CONSOLE BE OVER-ENGINEERED? ONLY IF YOU THINK "GOOD ENOUGH" REALLY IS GOOD ENOUGH.

SILVER

BRONZE

Building great consoles is more than punching holes in sheet metal and stuffing a few switches in them. Building a great console takes time, brain-power and determination. That's why we've hired brilliant engineers who are certified "OCD": Obsessive Console Designers, driven to create the most useful, powerful, hardest-working consoles in the world. And that description certainly fits Axia's modular Element 2.0 mixing console. Scalable from 2 to 40 faders, Element is the ultra-reliable dynamo at the center of over 3000 Axia-powered broadcast studios around the world.

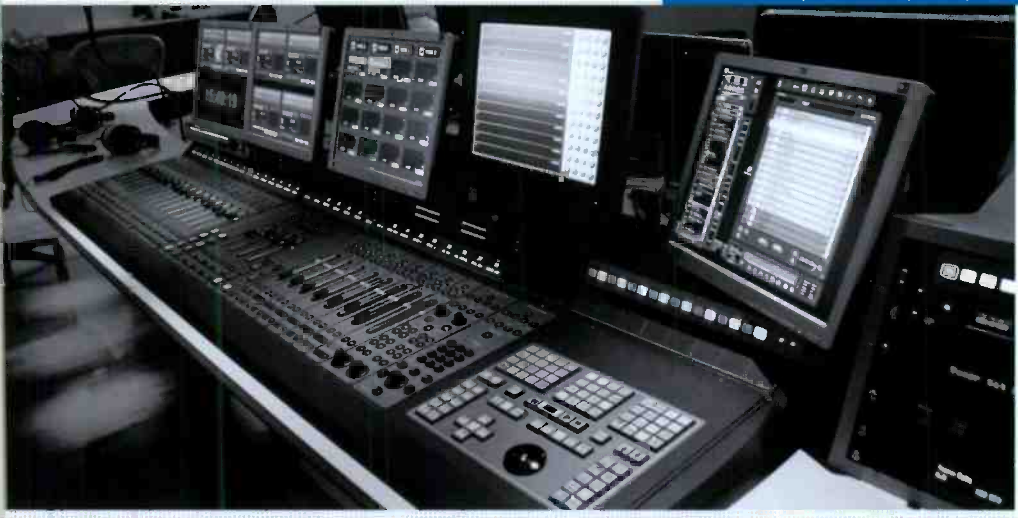
We launched Axia in 2003 to make digital mixing consoles — but with a twist: Axia consoles would be integrated with the routing switcher, and networked to share resources and capabilities throughout the studio complex. This intelligent network of studio devices gives your talent consoles that are more powerful and easier to use than ever. Our team of engineers 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.

And we invented a way to network studios, consoles and audio equipment using Ethernet. It's called Livewire, and it's now an industry standard. Livewire carries hundreds of channels of real-time, uncompressed audio plus synchronized control logic and program-associated data on just one skinny CAT-6 cable. And, because Axia networks are intelligent Ethernet-based routing systems, machine logic always follows source audio. Simply load a source on any fader – in any studio – and 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. Board-ops told us they wanted a powerful console that's still easy to use: user-friendly, but with all the power of a full-on production board. Element is! Show Profiles can recall each operator's favorite settings with the push of a button — audio sources, fader assignments, monitor settings and more. Each jock's Show Profile contains personalized mic processing and voice EQ settings that load every time they're on the air (so the midday guy will stop badgering you for "just a little more low end"). There's even a "panic button": one key-press returns a Show Profile to its default state instantly.

Powerful? You bet. Element has 4 Program buses plus 4 Aux sends and 2 Aux returns, and 16 5-channel "virtual mixer" that lets you mix multiple audio streams using virtual faders. Every voice channel has studio-grade compression, de-essing and gating courtesy of the processing experts at Omnia, plus three-band parametric EQ. There's even built-in headphone processing to save the cost of a separate side-chain.

More convenience: fully-automatic mix-minuses; one for each fader if you need it. Mix-minus settings are saved for each audio source, so that sources, backfeed and machine logic all

NRJ NETWORKS, PARIS, FRANCE



load at once. And every fader has a "Talkback" key to communicate with phone callers, remote talent or other studios using the console mic.

Axia's Livewire Ethernet backbone makes it easy to integrate and control all kinds of different devices on the same network and those controls are right on the console, where they're most useful. For instance, phone hybrid modules with dedicated faders to control Telos talkshow systems; there's even a dial pad so jocks can dial, pick up, screen and drop calls without diverting their attention from the console.

Our IP Intercom system connects to the Livewire network too; drop-in Element modules place multi-station intercom controls right at jocks' fingertips too. Talent can now easily take broadcast-quality intercom audio directly to air with just a couple of button-presses.

You 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 for rigidity and RF immunity. Power supplies are hardened for reliable, continuous uptime, and fan-less for silent in-studio operation. Modules are hot-swappable, of course. Silky-smooth conductive-plastic faders actuate from the side, so grunge can't get in. High-end optical rotary encoders mean no wipers to get dirty or wear out. And our avionics-grade switches are rated for more than 5 million operations.

So, are Axia consoles over-engineered? You bet. 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 is for you.

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.

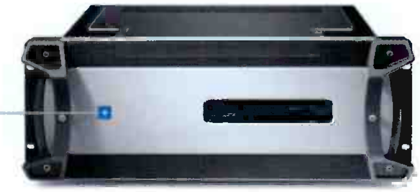
Film-Legendable Switch Module with fixed-function buttons is available in 5-button or 10-button sizes. Backlight colors are changeable, too. Use PathfinderPC software to assign buttons custom routing functions, audio device controls and more.

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.

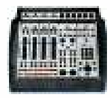
High-impact Lexan overlays with color and printing on the back, where it can't rub off. We don't just stick the Lexan to the top of the module like some folks do: our overlays are inlaid on the milled aluminum module faces to keep the edges from cracking and peeling — expensive to make, but worth it. 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.

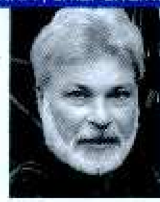
BRONZE
SILVER

PRODUCTION

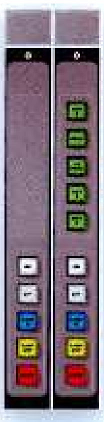
The Production module gives your talent direct access to frequently-used production options, such as Send bus levels, EQ settings, and panning tools. Includes overbridge selector panel with alphanumeric source display.

BRONZE
SILVER

CHUCK KNAPP, CHIEF ENGINEER, KMST



"I would highly recommend the Axia system for any broadcast facility. Installation is fast and easy. The system is very affordable and lends itself to expandability. We installed our Axia system in Aug of 2008 and have had zero failures so far! Also important, the factory support has been prompt, knowledgeable and extremely professional! We love our Axia system!"



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

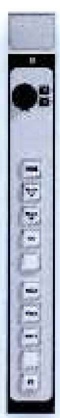
5 and 10-button SmartSwitch modules feature backlit LCD displays; functions, colors and text can change dynamically in response to user input. Use Axia PathfinderPC software to program SmartSwitches with custom salvos, machine-logic commands or other complex routing operations.

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

THE INTEGRATED, INDESTRUCTIBLE CONSOLE ENGINE.

This is PowerStation, the muscle behind our best-selling Element 2.0 mixing console. PowerStation combines four separate devices — a DSP mixing engine, a console CPU and power supply, audio I/O, GPIO and a custom, built-for-broadcast Ethernet switch – into one unit, a self-contained console engine that's engineered to ensure years of reliable, trouble-free service.

PowerStation makes it easy to get your studios up and running. Just connect PowerStation to your Element 2.0 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!

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.

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. No other console company makes AoIP this easy.

And there's more. Want more audio I/O? Simply connect a PowerStation Aux to instantly double your mic, analog, AES and GPIO ports. PowerStation Aux also adds a redundant, backup power supply with built-in switchover. Most redundant supplies protect only the console, but with PowerStation, the mixing engine, audio I/O and network switch are protected as well — another Axia exclusive.

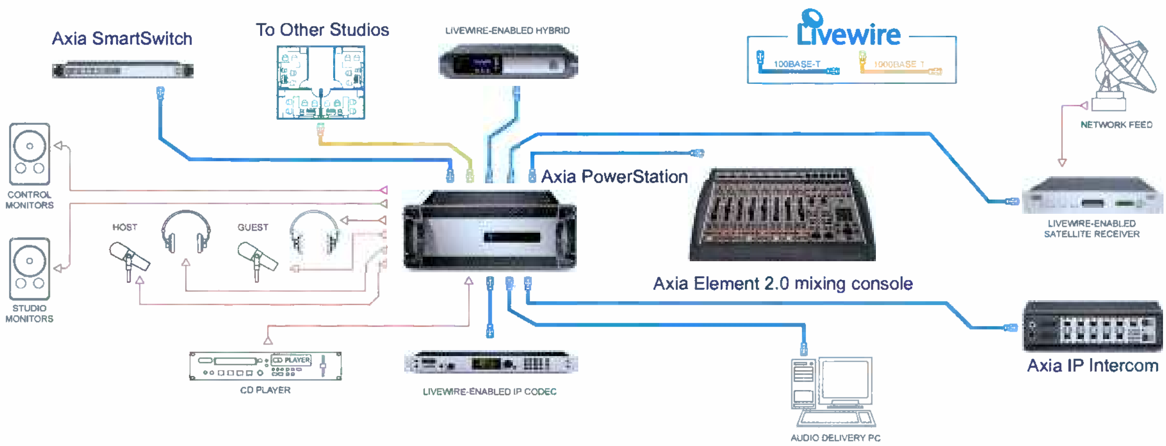
Naturally, PowerStation is over-engineered to Axia standards. Every part of PowerStation is chosen for its ability to give constant, uninterrupted service, 24 hours a day, 7 days a week, 365 days a year. There are no compromises: PowerStation uses only best-of-the-best components, like studio-grade mic 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.

SELF-CONTAINED, BUT PLAYS WELL WITH OTHERS.

PowerStation and Element combine to make a networked broadcast console that doesn't need a network. It's completely self-contained: it works flawlessly as part of a large network, but if you unplug its network cable, it's completely unaffected. Think of it as an "island of reliability."

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 to expand your Axia network — up to 10,000 stereo streams).

It's easy to build IP studios with PowerStation. With 32 built-in I/O ports, DSP mixing engine, console CPU, bulletproof power supply and network switch all in one fan-free enclosure, you'll be making great radio in no time.



MULTILINGUAL, MULTI-STUDIO

IP-AUDIO AT RADIO FREE EUROPE / RADIO LIBERTY

In the decade since IP-Audio networking technology first debuted to the broadcasting community, thousands of studios, from large to small, have been built around it. Still, in conversation, people not familiar with the technology still ask, "Is AoIP really robust enough for round-the-clock radio?" On those occasions, the exciting operation that is Radio Free Europe / Radio Liberty, located in the Czech Republic capital of Prague, comes quickly to mind as a shining example of just how robust Audio-over-IP is.

The impressive new home of Radio Free Europe / Radio Liberty

For more than 60 years, Radio Free Europe and Radio Liberty have been broadcasting news, information and discussion to millions of listeners throughout Europe and Central Asia. RFE/RL, as it's commonly known, originates more than 1,000 hours of programming per week, in 28 languages and broadcast to more than 20 countries. That's a lot of programming, and it requires a stout infrastructure to keep it all running smoothly, hour after hour, 365 days a year.

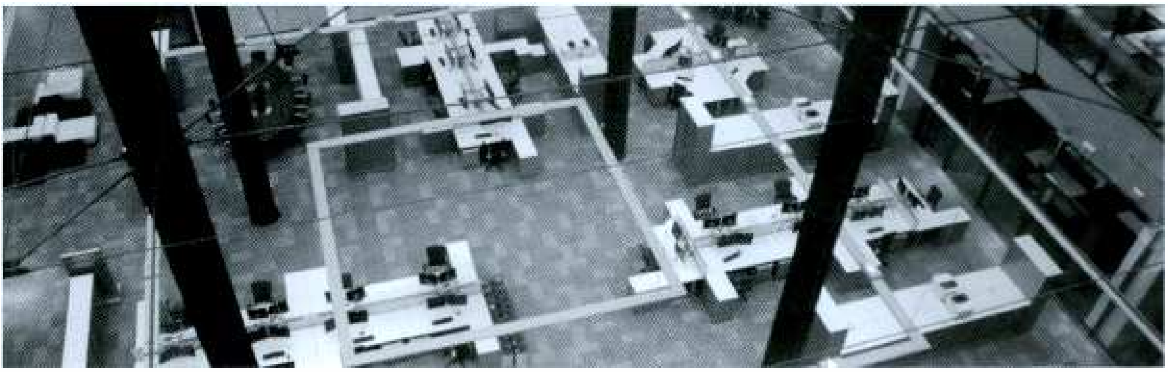
Originally headquartered in Munich, Germany, RFE/RL moved to Prague in 1995. For the next 13 years, they occupied what had been the Soviet-era Czech Parliament building in historic Wenceslas Square. It served its purpose, but the structure had been built to fulfill a different kind of mission – one almost diametrically opposed to the one it now served, as a free disseminator of news and information. As the service approached its sixth decade and looked to its future, it was clear that to both grow and update, it would have to move to a new HQ.

The plan, as it evolved, was this: relocate RFE/RL to a new, modern, purpose-built facility in Hagibor, outside of Prague's bustling center. RFE's new house would be big, with more than 200,000 square feet to contain the activities of over 500 employees daily.

Bill Cline inspects RFE's Axia "test studio," September, 2007

RFE/RL's Director of Broadcast Engineering, Bill Cline, was intimately involved in the new studios' design. He liked the flexibility of networked audio, and oversaw the building of studios using Axia consoles and IP-Audio networking equipment in 2007; the success of those tests confirmed that Axia Livewire AoIP was up to the operational requirements of a large operation like Radio Free Europe's.

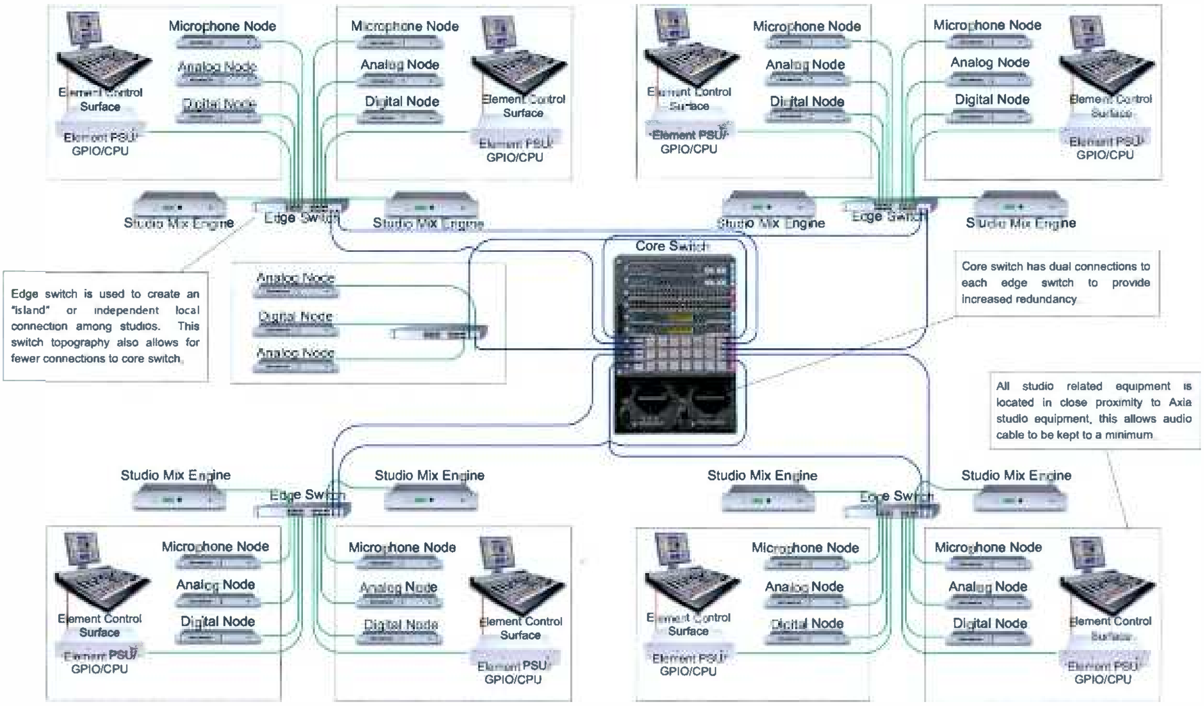
RFE/RL's Central News Room, providing news and information for 27 different language services.



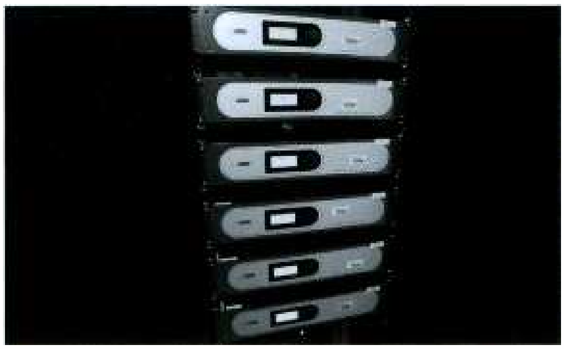
In 2009, the new RFE/RL headquarters was placed in service. It is one of the largest radio origination facilities in the world; in addition to radio programming, the new facility houses an enormous news gathering organization and TV distribution facilities as well. The radio broadcasting "suite," if something so large can be called that, consists of 49 on-air and production studios. Axia consoles and IP-Audio equipment are the basis for them all. While large, this is well within Axia's system capacity, which is able to route and transport up to 30,000 stereo audio channels.

Among RFE/RL's unique operational requirements is the ability to monitor any of its programs from anywhere in the facility. With so many studios producing programs in so many languages, this may seem like no small undertaking! However, Axia and Telos technology and equipment make this easy and elegant. Banks of Telos iPort MPEG Gateways are employed to handle the task. These devices connect directly to Axia networks; using iPort, RFE converts all of its audio programming directly from Livewire IP-Audio to Multicast MP3 streams, which are available at listening stations or on PCs anywhere in the facility. Each iPort is capable of encoding up to 16 MP3 streams at once.

RFE/RL studio setup features super-redundant distributed core/edge architecture with multiple network connection paths.
Diagram illustrates only a small portion of the RFE network.



This rack of Telos iPort codecs provides 96 MP3 audio streams that RFE/RL employees can audition from anywhere in the 6-story plant.



Satellite feeds likewise travel around the facility via IP, with uplinks and downlinks arriving at their destinations as low-latency, routable Livewire real-time audio.

Studios are equipped with Axia Element 2.0 mixing consoles of various sizes. Some are large production studios with control of associated on-air rooms, with large, 28-position consoles. Others are outfitted as combination production/broadcast booths, with smaller mixing boards. All make use of editing stations from D.A.V.I.D. Systems, an Axia partner. These systems connect directly to the Livewire network using the Axia IP-Audio Driver for Windows; this eliminates a significant amount of discrete cabling, and bypasses the need for hardware audio sound cards as well.

One of the 49 studio spaces equipped with Axia Element consoles, Telos broadcast phone gear.
A touch of the Show Profiles button can instantly reconfigure the entire board.

The studios are further customized for their users through the use of Show Profiles, a console "snapshot" capability that's built into Axia consoles. These console configuration presets – up to 99 of them - can be saved on each Element console allowing source selections, channel assignments, audio EQ settings, and even monitor source preferences to be stored in named Profiles that talent can select in seconds. With so much going on, this is essential for RFE/RL, whose operators regularly use the feature to completely reconfigure an entire console to a personalized layout at a moment's notice. And of course, on-console control of call-in systems, editors and codecs helps to streamline operations even more.

In a facility of this size, reduction of discrete cabling results in significant cost savings. More networked equipment integration is achieved through use of Telos Nx-series multi-line phone systems and Zephyr Xstream ISDN codecs in-studio, which connect to the Livewire network using a single CAT-5 cable.

Bill Cline shows off the meter/status wall in RFE/RL Master Control.

One of the most impressive places in the new RFE facility is Master Control. The nerve center of the radio operation, Master Control allows RFE engineers to monitor every audio feed in the facility, as well as satellite up- and down-links (including Voice of America backup links), signal status with channel audio metering, network bandwidth utilization – right down to information as granular as the status and output levels of each studio's program audio outputs. All of this is arranged on a massive array of flat-panel screens that occupy the space of nearly an entire wall. It's a truly remarkable sight. Engineers can also make system-wide routing changes to the Axia network, using hardware button panels mapped to routing commands customized using Axia PathfinderPRO routing control software.

RFE/RL began officially broadcasting from its new HQ in April, 2009 – nearly 3 years ago as of this writing. During that time, its Axia IP-Audio consoles and network have provided unfailing 24/7 service, to the great satisfaction of all involved. So when someone next asks the question, "Is AoIP really that good?" I'll smile and ask a question in return: "Have you heard about Radio Free Europe?"

Clark Novak
Marketing Manager
Axia Audio

iQ

THE SMART CONSOLE FOR THOSE WHO THINK FOR THEMSELVES.

We know you. You're the guy who stays up late, prowling the Net for new ideas, reading the newsgroups and listserves. The guy who worked all weekend getting that new studio ready for Monday morning. You're creative, energetic, and excited about what you're doing. You're the independent thinker.

Axia works just as hard as you do. Our dedicated team of obsessive console designers is never satisfied with success. They're always trying to top their last breakthrough, working constantly to devise new and more inventive ways to save a dollar. Always asking "what's next?"

The answer to that question is iQ. More than just a pretty face, iQ is a control surface 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.

Thanks to all those built-in goodies, iQ is perfectly happy working as a self-contained, standalone console in an individual studio. But should you wish to expand and network with other studios, iQ is ready to 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 faders, eight at a time, to create consoles as large as 24 faders. With your choice of optional expansion frames, you can even control telephone systems and GPIO routing functions right from the console.

More smart stuff: iQ remembers. Four Show Profile memory positions let you set, save and recall snapshots of console settings

RADIO MIRCHI SHOWS THEIR iQ SOME LOVE!

RADIO HTTY, CZECH REPUBLIC

for later use. High-resolution Organic LED meters offer switchable VU or PPM metering styles, and 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.

And like all Axia consoles, iQ is built solid, like a Mack truck, to withstand even the beatings a weekend overnight jock can give. LED button lighting, long-life conductive-plastic faders, and anodized 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 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.

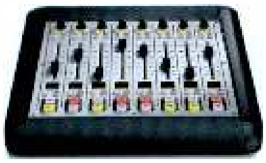
iQ

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

WITH iQ, YOUR PHONES ARE SMARTER, TOO.

Up to now, adding phone support to your console could be time-consuming. iQ6 (the new Telco gateway from Telos designed for 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.

You want off-console control? No problem. Use a Telos VSet phone, with its big color screen and animated icons, to make short work of pre-screening tasks or use the provided XScreen software from Broadcast Bionics for a full-featured software screening environment. iQ6 accepts POTS or ISDN phone lines, and comes equipped with two high-performance Telos hybrids with Digital Dynamic EQ — the same advanced hybrids found in our Nx phone systems.

Naturally, since it's part of the iQ system, iQ6 saves you money and time. It plugs right into one of the Livewire ports on your QOR.32 console engine, eliminating tedious soldering of discrete I/O cabling and connectors — drastically reduces installation time, too. One skinny CAT-5 cable carries six lines' worth of audio, hybrid control and mix-minus.

BALLYFERMOT COLLEGE OF FURTHER EDUCATION IN DUBLIN, IRELAND

JIM REESE, MANAGER OF DIGITAL TECHNOLOGY, KUT

"I will not build a studio without Axia ever again."

RADIUS

MORE BANG FOR YOUR BUCK.

Everybody knows you get what you pay for. And sometimes, quite a bit less. Ever notice how "affordable" radio consoles are usually missing stuff? Important stuff. 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 believe that having a reasonable equipment budget shouldn't mean having to settle for a stripped-down, poorly built, featureless plastic excuse of a console. We've decided you should get more than you pay for — much more. Meet Radius, the IP console that proves you can have your cake and eat it, too.

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! Radius has 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. Radius generates automatic mix-minus for phone callers and remote talent, helping ensure seamless, error-free shows. Bright multi-segment LED meters are switchable between VU and PPM styles, and high-resolution OLED displays for each fader show source assignments, audio options and more. Show Profiles can be saved and recalled to instantly load frequently-used console configurations.

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.

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 OLED high-rez displays on every fader.

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.

MILOS NEMCIK, AXIA PRODUCT SPECIALIST

"Remember those old, boxy boards that used to be in every radio studio? They were about as efficient to use as a Diesel locomotive (and nearly as big). RAQ and DESQ, our newest Axia consoles, do more than those old clunkers ever would, and use a lot less space to do it. Smart, efficient, elegant...Just the way it should be."

MEET RAQ AND DESQ

SPECIALIZED CONSOLES FOR SPECIALIZED PROJECTS

Sometimes, you don't need a big console, with all of its bells and whistles. What you want is something you can tuck into a rack, or place on a countertop without taking too much space. A mixer whose small footprint belies its big capabilities.

Sure, you could rummage around in the closet for an old mixer, and you'd probably find one, complete with peeling blue paint and held together with packing tape and rubber bands. "C'mon, it's 2012," you think. "Do I really want my talent creating programming with something that needs a battery to make its meter bounce?"

Introducing RAQ and DESQ, two new 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. Compact, yes, but don't let that fool you — they're packed with IP-Audio goodness. Which means they easily out-class and out-perform mixers that take up a lot more room.

RAQ

THE RACK-MOUNT MIXER FROM AXIA

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! Your 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.

(By the way: If the design of RAQ looks pleasingly familiar, take a peek at the photo on the bottom of Page 25. That PR&E Stereomixer was also designed by our own Michael "Catfish" Dosch, about 25 years ago! Just goes to show that you can't keep a great idea down.)

DESQ

THE COMPACT DESKTOP MIXER FROM AXIA

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 got a machined-aluminum work surface to take all the tough stuff jocks can dish out. Our familiar 100 mm. side-loading faders feel like silk under the fingertips, and you'll also find our familiar avionics-grade switches with LED lighting, 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.

RAQ and DESQ also have "big board" capabilities you won't find in other consoles of this size. Like automatic per-fader mix-minus. Built-in EQ available for any voice or codec source. And the ability to instantly load new local or networked sources to any fader with just the turn of a knob.

QOR.16

INTEGRATED CONSOLE ENGINE

RAQ and DESQ are powered by our QOR.16 integrated console engine. It's fanless: silent and convection-cooled, 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 broadcasting, or as personal mixers. What else could you use them for? The possibilities are endless...

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

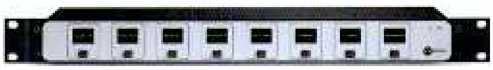
FINGERTIP CONTROL JUST WHERE YOU NEED IT.

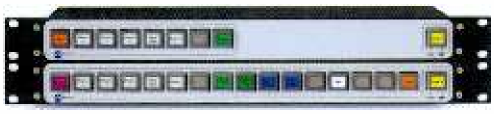
Axia control panels, used with Axia Pathfinder routing control tools, let you put routing power anywhere — in a studio turret, a TOC control panel or an equipment rack. These accessory control panels work with Axia's Pathfinder software and Pathfinder Core routing control hardware, allowing you to map routing commands – from simple contact closures to complex logic-driven events – to any button for fast execution. .

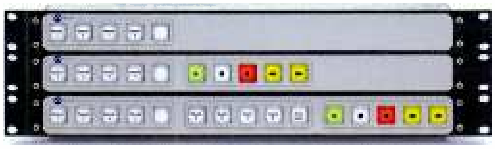
Film-cap controllers (in rack-mount, cabinet-mount and Element console drop-in versions) with backlit keys can be illuminated with a choice of colors; keycaps are film-legendable for quick function identification. SmartSwitch panels have dynamic, backlit LCD buttons that can change color and text with 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.

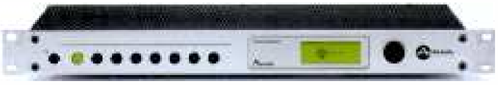
A family of three Routing Control Panels (X1, X2 and XY) allow convenient on-the-fly routing of networked sources from anywhere in your facility. X1 and X2 controllers let you route any networked audio source to a pre-selected fixed output; XY controller lets you dynamically route any source to any output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

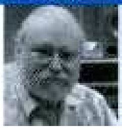

5, 10 AND 15-BUTTON FILMCAP PANELS


ROUTER SELECTOR NODE


X1, X2 AND XY ROUTER CONTROL PANELS


CABINET MOUNT BUTTON CONTROLLERS

BRIAN URBAN, CHIEF OPERATOR, KUT RADIO



"Building a new production room, we started loading equipment on a cart at 12:15. We went to lunch at 1:45 with the room usable. That's how easy the installation is."

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.

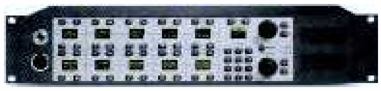
So you've gotta be a genius to use it, right? Actually, anyone with an index finger can operate this system with ease. The web interface makes setup simple. Sharp, high-contrast OLED displays are easy to read from anywhere in the room. And our clever callback feature makes sure you'll never miss a call, 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

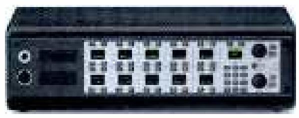
IC.10 has 10 station presets with OLED displays. Like its big brother, the IC.20, it has a built-in speaker, front and rear-panel mic connections, 4-pin locking headset jack, analog I/O presented on both XLR and RJ-45 connectors, and a GPIO connection for speaker mute/dim and external line-status tallies.

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

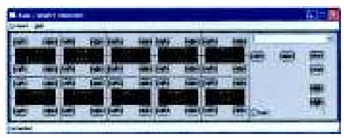
IC.1D has 20 preset stations presented on LED-backlit button caps; an economical way to add intercom function to any space.

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.

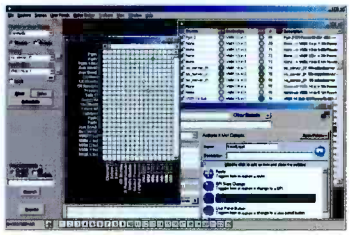
With Axia, you can make use of a wide variety of hardware and software-based tools designed to help you easily harness this power. After all, what good is power without control?

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.

MIKE SPRYSENSKI, MARKET ENGINEERING MANAGER, CLEAR CHANNEL , ORLANDO, FLORIDA

"The on-site training with Pathfinder was fantastic and definitely has put us down the right path on getting the new AIR, Master Control, and Talk studios online for 5 of my 7 stations in the coming months. My plan is to continue installing Axia gear for the rest of the studios including Production Rooms, Newsroom, and other studios until everything is converted over to the Axia platform. As I have said before, Axia has the best support in the broadcast business and it has made me a customer for life!"

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

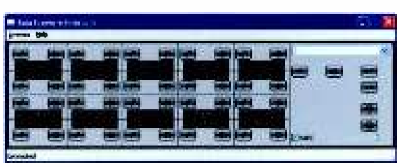
Axia's networked version of the popular Telos ProFiler logging software lets you simultaneously capture up to 24 stereo audio channels to time-stamped MP3 audio logs directly from your Axia IP-Audio network — no audio cards required.

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

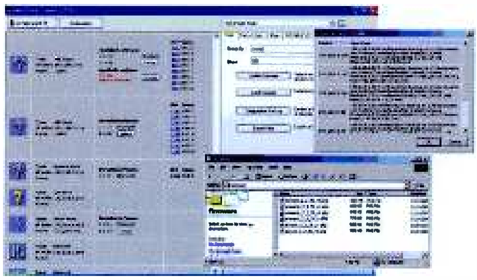
Make any networked PC with a sound card part of your Axia IP Intercom system. Axia SoftCom software works just like our rack-mounted Intercom hardware panels, with preset locations for 20 frequently-called stations. Auto-answer and hands-free functions are provided; an easy drop-down station finder gives 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.

AXIA | NETWORKING

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."

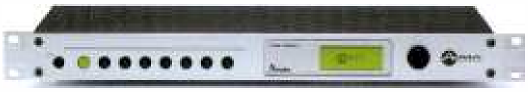
GPIO xNODE

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.

AUDIO NODES

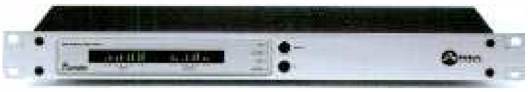
IP-AUDIO INTERFACES

Axia Audio Nodes are perfect for rack rooms, TOCs, studio turrets or anywhere else you have audio inputs. Like xNodes, they're fan-free and have studio-grade audio response and front-panel confidence meters. A built-in webserver makes setup easy using any PC with a Web browser. Like all Axia gear, they're over-engineered for 24/7 service.



ROUTER SELECTOR

The Router Selector Node's LCD screen lets users browse and select from a list of all available sources; eight "radio buttons" provide instant access to favorite sources. There are analog, AES3 and headphone outputs, and even a convenient analog and AES3 input — ideal for production or news studios where operators typically both create and play audio streams.

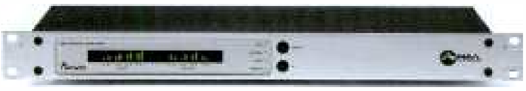


ANALOG LINE

The Analog Line Node has eight balanced stereo inputs and eight balanced stereo outputs, all on RJ-45 connectors, and each input is switchable to accommodate consumer-level -10dBv or professional level +4dBu equipment.

MICROPHONE

The Microphone Node has eight studio-grade mic pre-amps with selectable Phantom power and software-adjustable gain. There are also eight balanced analog line outputs for headphone and studio monitor feeds. Inputs use XLR connectors; outputs are on RJ-45's.



AES/EBU

The AES/EBU Node provides eight stereo AES3 inputs and eight AES3 outputs. Sample-rate conversion is available on all inputs; the unit can be used to sync the Livewire network to a house clock.

GPIO

The GPIO Node provides 8 logic ports for machine control, each with 5 opto-isolated inputs and 5 isolated outputs that can be associated with any audio stream.

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.

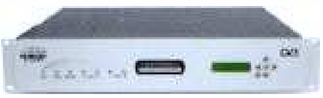
LIVEWIRE® INTERFACE CARD

PARTNER PRODUCTS

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

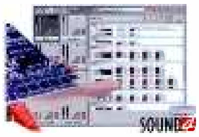
25-SEVEN AUDIO TIME MANAGER

AUDIOSCIENCE 6585
LIVEWIRE SOUND CARD

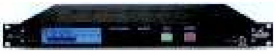


INTERNATIONAL DATACASTING
SFX4104 EXP SATELLITE RECEIVER

FRAUNHOFER CONTENTSERVER



SOUND4 IP PCI
AUDIO PROCESSING CARD



PARAVEL SYSTEMS IROUTE

NAUTEL VS1 FM TRANSMITTER

WHY DOES AXIA OUTSELL EVERY OTHER IP CONSOLE?

WE'VE GOT CONNECTIONS

Did you know that there are over 3000 Axia consoles on the air? That's more than all other AoIP consoles — combined. Is it because our ads are so irresistible? Our marketing guys think so... but, no. It's because broadcasters know that a network's value increases with the number of devices that talk to it. And nobody connects to more IP-Audio devices than Axia.

With this huge installed base of broadcast studios around the world, we've attracted dozens of partner companies, all offering Livewire™ compatible products. A device with a Livewire port is instantly available to any other device on the network. So, if you're shopping for IP consoles, be sure you ask: "How many partners do you have?" Because a network that only plays with itself isn't very well-connected... is it?

EVOLUTION

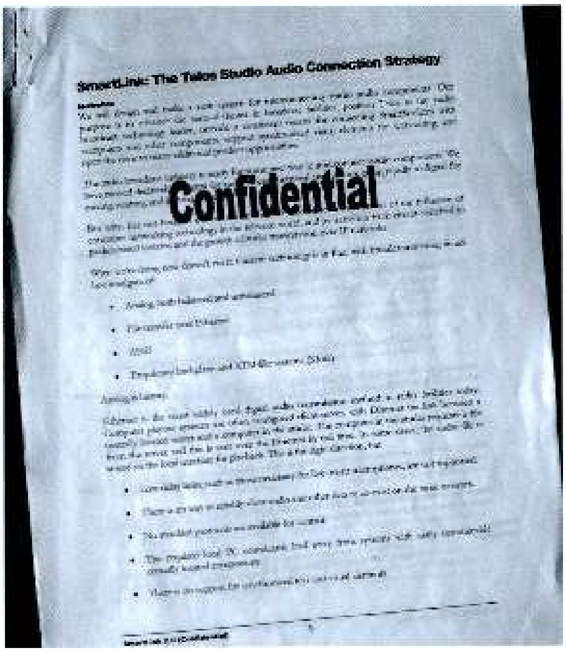
AoIP IN BROADCAST ENGINEERING

This year we celebrate a decade since the first public demonstration of AoIP. Does that seem too long a time? It's not — although officially revealed a year later in 2003, Livewire actually debuted at NAB in 2002. Sitting hidden inside Telos display furniture, it was secretly powering the demo of our SmartSurface mixing console. So it seems appropriate that we look back on how AoIP began and evolved, see where it has taken us today, and maybe take a peek into the future.

PRO AUDIO OVER IP? THAT'S CRAZY! OR IS IT?

By the late 1990's, data networking was already present in everyday activities. There was a good selection of reasonably priced networking devices and cabling materials on the market, and connecting a basic office data network was a relatively easy, almost plug-and-play task, thanks to the huge effort that the computer and data networking industry had been making for many years.

Radio broadcast facilities were no exception — on the office side, at least. But in the studio? Racks of tremendously expensive special-purpose equipment, and tons of cabling could still be found there. The equipment at the heart of broadcast radio was, back then, far behind that of the data networking industry.


SmartLink: The Telos Studio Audio Connection Strategy

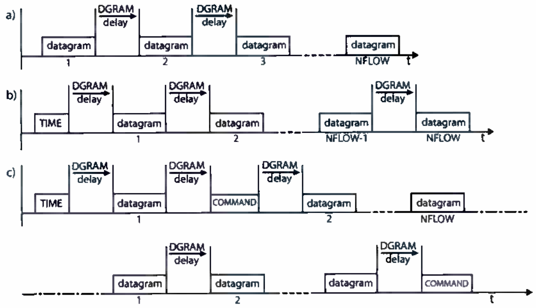
Confidential

In October, 2000, Steve Church visited the Real-Time Systems laboratory at the University of Latvia, bringing with him an extremely interesting document: a technical outline of a completely new approach to building audio infrastructure for a modern radio broadcast facility, based on using standard Ethernet/IP-based protocols and off-the-shelf data network equipment.

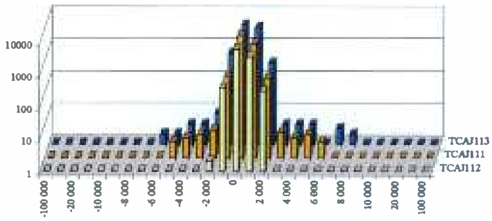
Two major misconceptions stood in the way of introducing modern data network technologies into broadcast applications; namely that *Ethernet = Internet*, and that *PC Platform = MS Windows*. In reality, Ethernet is capable of delivering huge bandwidth at excellent performance and very reasonable cost — switched-duplex Ethernet links do not suffer from the problems that are so common on the Internet. And x86 hardware was a universal high-performance processing device that could support sub-microsecond timing resolution. The OS was what slowed things down, not the hardware.

Understanding these two fundamental facts allowed Telos to start one of the most ambitious R&D projects the radio broadcast industry has ever seen. Its successful implementation in just a few years opened the doors to a major technical breakthrough in many broadcast facilities all over the world.

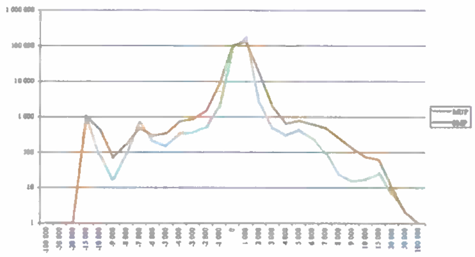
The work started in October of 2000, at the University of Latvia, and the beginning was rather academic. A team of highly skilled software and data networking experts started examining various performance aspects of the Linux operating system and switched Ethernet. Many hours of tricky experiments produced CPU/OS and Ethernet throughput estimates, multicast and QoS behavior test reports, network packet latency distribution graphs at different conditions, and more.



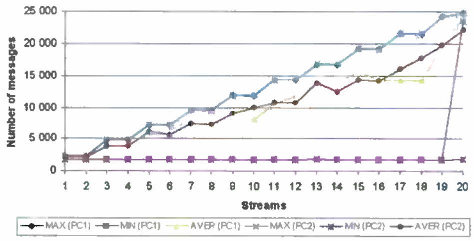
Compare performance of latency and jitter test under different load in RTAI



Compare jitter using MUP and SMO schedulers (logarithmic view)



Measurements after Leave, t_between = 225 ms



The results were very promising, and soon what had been research began quickly transforming into software design outlines and pieces of working code. A lab demo of a fully working Livewire link was set up in June of 2001, at the University of Latvia in Riga, and we began two years of intensive work devoted to bringing Livewire to broadcasters. We built prototypes of a networked mixing console, a mixing engine, and audio IO devices of several types, as well as designed and implemented intelligent and user-friendly software. In April, 2003, the result of this highly involved effort of an entire engineering team resulted in the public introduction of Axia Livewire IP-Audio at the Las Vegas NAB show.

NAB 2003, Las Vegas.

Livewire technology receives its first award.

GROWTH

Somewhat surprisingly, considering the radical nature of our new idea, those who saw the Axia demonstration at NAB 2003 seemed convinced that it would work. What we didn't realize is that, while they approved in theory, they certainly didn't want to be the ones whose facilities proved the theory sound!

The first sale of Axia Livewire equipment happened in late 2003, about half-a-year after its introduction. It was a simple pair of Axia audio nodes, used as a digital snake to take a few audio sources between two locations via optical cable.

New York's WOR

The first large station cluster with Axia. That's SmartSurface, our first console design. It caught a lot of people's attention.

In early 2004, after months of showing our new baby around, the first commercial AoIP studios went on-air at Radio Skonto in Riga, Latvia, followed closely by Auburn University's WEGL-FM in Auburn, Alabama.

After that, acceptance of IP-Audio started growing rapidly. Broadcasters of all sizes took the plunge, from low-budget college radio, to privately-owned media operating both single studios and networks of all sizes, to big public-funded networks. By 2007 the total number of studios had reached 500, and accelerated to reach over 3000 active on-air studios equipped with Livewire by the time of this writing (early 2012).

A few notable events, illustrating the scope of this evolution:

» In May, 2005, New York's WOR became the first large facility to select Livewire AoIP, choosing it as the basis for its new facility on Broadway. WOR installed 7 networked studios along with several hundred audio sources.

» In 2006, Minnesota Public Radio in Minneapolis/St. Paul installed 20 networked Axia studios. Also, Radio Free Asia chose to install Axia in their Bangkok, Thailand news center.

» During the next years a number of other highly reputable broadcasters chose Axia Livewire too – Clear Channel, RTL, Univision, Southern California Public Radio, and more. At the same time, many smaller private and public stations in the USA, Canada, South America, all parts of Europe, India, China, Australia, and other countries all around the world decided to go with AoIP as well.

» In 2009, a fascinating government-funded project in New Zealand called PungaNet utilized Axia AoIP to cover the whole island with 21 networked studios, interlinked by means of MPEG-coded channels and a custom-designed RTP router.

» Also in 2009, Radio Free Europe/Radio Liberty – one of the world's biggest and most reputable broadcasters – went on air with 50 networked Axia Livewire studios from its new facility in Prague, in the Czech Republic.

Radio Free Europe

One of 49 Axia studios at Radio Free Europe / Radio Liberty

The steady growth of AoIP didn't go unnoticed by other equipment manufacturers, who quickly realized the value of making their products connect directly with the growing AoIP community. ENCO Systems was the first, followed by many other software and hardware manufacturers, covering a wide product range that grew to include consoles, sound cards, codecs, various processing devices and playout systems. In 2010, Nautel Ltd. announced the Axia Livewire AoIP interface would be included into one of their transmitter product lines — the "missing link" that finally made possible an all-AoIP broadcast chain.

A few other companies (after first denying that the technology would work) launched their own, proprietary, AoIP solutions. Unfortunately, none of these are compatible with each other, or with other broadcast hardware.

But, good news: the value of interoperability between products from different manufacturers is clearly recognized these days, and significant efforts are currently being applied in that direction by several organizations.

THE AoIP LANDSCAPE IN 2012

Today, AoIP has earned wide recognition. This point is driven home by the multitude of products present on the pro audio and broadcast market, and in field, ranging from single devices up to complete multi-vendor system solutions. There has also been much attention given by the trade press, including hundreds of different articles online and in industry magazines. There is even a book, Audio Over IP: Building Pro AoIP Systems With Livewire, published by Focal Press/Elsevier.

Along with this attention have come specialized seminars and conferences on AoIP technologies organized by the Audio Engineering Society, the NAB, the Society of Broadcast Engineers, the European Broadcasting Union, and others. There are also interoperability initiatives and ongoing specification work at several private companies and International organizations.

Livewire, originally developed by a single company, today is an open solution providing interface documentation, a complete reference design, and an off-the-shelf Dolby-E plug-in interface board. Although it is not, and never attempted to become a formally recognized interoperability standard, Livewire is by far the dominant AoIP technology in broadcast. It counts over 3000 programs on-air, 25 000 devices streaming a few hundred thousand audio sources, and more than 30 different broadcast manufacturers offering Livewire-enabled equipment.

We at Axia are extremely happy and proud to see how the initiative started by us more than 10 years ago has transformed one dream into reality – the dream of affordable, efficient technology for building pro audio networks.

TECHNOLOGY

Now would be a good time to look at the reasons behind IP-Audio's remarkable growth, and some of the technologies now available.

WHY IS AoIP SO GOOD?

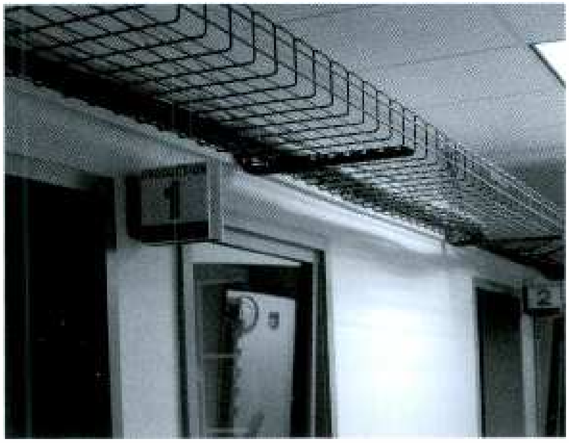
It's high tech at an affordable price. There could be many different answers to the question above, but this is probably the most fundamental one. AoIP leverages decades' worth of huge investments by the computer and data networking industry, offering extremely high technology at mass-market prices. Custom-designed systems often offer either high technology or an affordable price, but very rarely both at once. The ability to do so shines a very favorable light onto AoIP.

The following points are, to a great extent, the benefits of this. High technology allows building efficient and intelligent applications, which is what AoIP solutions have always been about.

Universal network infrastructure. The inherent capability of IP networks to multiplex a variety of protocols and applications on common cabling and interfaces accommodates nearly anything, and in any combination. So the network infrastructure can be shared by many very different tasks – audio streaming, machine control, program metadata, even regular office work.

Huge bandwidth over a compact physical media. The most common lower-layer transport media for IP is Ethernet. Everyone knows how thin an Ethernet cable is, but in terms of raw digital bandwidth, a 1Gb/s Ethernet link is equivalent to about 300 AES-3 links, or about 500 E1's, or about 650 T1's. Which means significant savings on both the space requirements and cost of the cabling.

See the empty cable tray?



This is what happens when one plans for TDM, but gets AoIP.

Flexibility and scalability. Huge physical bandwidth and packet switching make IP networks nearly limitlessly flexible. You can easily allocate the entire bandwidth of a 10 Gb/s IP/Ethernet link to a single media stream — or use it to deliver one million low-fidelity audio feeds. IP-based solutions scale easily; any newly installed or reserved capacity can be instantly allocated for any purpose – no system configuration, and no network redesign needed. This greatly simplifies planning and building audio applications, and opens the door to convergence of audio and video handling systems.

Ability to operate over different lower layer technologies. While the copper Ethernet LAN is doubtlessly the most widely used MAC/physical layer for IP, the IP protocol itself is not bound to any specific underlying protocol or physical transmission media. To mention just a few most widely known technologies, IP can be carried also over ISDN, ATM, MPLS, IEEE 1394, or DSL links. This fact makes the IP-based transport nearly future-proof, thanks to its ability to propagate through various network types, including mixed-technology ones, and the ability to survive evolutionary changes in the carrier networks.

Virtual soundcards. A small detail? In fact, it is huge. No other technology but AoIP allows converting a regular PC into a high-quality multi-channel sound card array without spending a cent for special hardware. The cost of expensive hardware sound cards is crossed off the list and replaced with a software component at a small fraction of that cost.

The opportunity to use standard software. Being IP-based, AoIP can immediately benefit from many useful tools developed by the Internet community for rather different purposes.

- IP audio can easily be monitored on a regular PC, using standard audio player software.
- Device management is made easy – all that you need is a standard web browser. One might argue that a non-IP system can do that too, and in fact many do, but it costs to add yet another hardware interface and a network link just to serve this one purpose. An AoIP device can get it for no additional hardware cost, excepting the tiny fraction of the link bandwidth that the HTTP management takes.
- Inexpensive, up to completely free, diagnostic tools are available for development and most field troubleshooting needs. Usually a network protocol analyzer is an expensive piece of equipment, and it may get even worse with a closed proprietary system, where the manufacturer may prefer to keep the technical secrets and restrict availability of monitoring and analysis tools to the authorized personnel only. IP networks are probably the only exception, where complex monitoring and analysis is feasible without any investment in tools – on a standard PC, using free software like WireShark, for example.

LIVEWIRE

Why, exactly, is Livewire so popular? Here are a few reasons:

- It's an open solution, built on the basis of standard technologies.
- It ensures a less-than-1ms network hop delay for low-latency audio streams, and can accommodate over 10,000 sources in every isolated network.
- It employs standard IP over switched Ethernet as the carrier network, and RTP/UDP for audio transport – no proprietary schemes.

» Livewire's patented synchronization filtering algorithm in slave devices ensures exceptional stability in presence of network jitter, and works equally well both with the original Livewire sync protocol and the IEEE 1588 (PTP) standard. Under certain conditions, Livewire networks can achieve better sync performance than IEEE 1588 alone would deliver.

» IP multicast-based routing ensures efficient one-to-many connectivity, as well as instant setting up, rerouting, and clearing of individual connections.

» Advanced device and source discovery protocol ensures instant availability of dynamically changing data throughout the network.

RAVENNA

RAVENNA (an acronym for *Real-time Audio Video Enhanced Next-generation Networking Architecture*) is a new offering from Lawo, the German manufacturer of ultra-high end broadcast consoles. Just recently they have started to demonstrate real device prototypes.

From the fundamental technology viewpoint, in the area of pro audio, RAVENNA is actually making a second round on the basis of IETF documents. Although it has its own independent roots, this effort essentially results in a generalization of the Livewire solution. RAVENNA follows exactly the same way of thinking as does Livewire, which is to say that it is aimed at building open systems on the basis of widely adopted public standards – so it's not a surprise that they, too, are building their technology on IP, UDP, RTP, and the related protocols.

There is a difference in the business model though:

» The Livewire project has been focused on bringing a practical solution to market since its very inception. It was primarily designed for a specific product line, although fully recognizing the value of being standards-based and open. The focus on a product resulted in selecting a technically sufficient, and at the same time economically justified, functionality set. This precisely defined functionality is consistently supported across the entire Livewire product line, ensuring universal interoperability.

» Unlike Livewire, RAVENNA started from offering a highly generic specification, which taken alone can not effectively ensure multi-vendor interoperability due to too many variables that are left unresolved in the framework. This will be addressed by means of providing precisely defined interoperability profiles – a work currently in progress.

As to the actual technical differences in the application overlap area, there is not much to speak about. Besides allowing multiple interoperability profile definitions, the biggest difference is the selection of a synchronization method. While Livewire uses its own synchronization protocol, RAVENNA has selected the IEEE 1588 (PTP) standard, which simply did not exist yet at the time Livewire was developed. However, today even this difference starts to disappear, as Livewire has introduced support for IEEE 1588 beginning with the debut of Axia's xNodes. And of course, Livewire retains the option for users to operate with its original sync protocol as well.

To put all this into context, in terms of the RAVENNA framework, the audio streaming part of Livewire would correspond to a specific profile. Paradoxically, if such a profile were defined, at the time this is written, Livewire would be the only existing RAVENNA profile that is implemented in real products, widely deployed and field-proven.

There is also open specification work in progress yet, especially at the higher application layers of RAVENNA.

OTHER AoIP TECHNOLOGIES?

Thanks to the popularity of Livewire, some companies have announced their own competing AoIP protocols, but these are closed, proprietary solutions, and information about them is limited. None have achieved wide acceptance in broadcasting.

» Dante: 100Mb or Gigabit Ethernet, IEEE 1588-2002, UDP unicast or multicast, 48/96/192kHz sampling rates, 24-bit proprietary coding, Bonjour discovery

» Q-LAN: Gigabit Ethernet or higher, IEEE 1588-2002, UDP unicast, 48kHz sampling rate, 32-bit floating proprietary coding, proprietary discovery protocol

» WheatNet: Gigabit Ethernet, UDP multicast, 48/44.1kHz sampling rate, 24-bit RTP coding

THIS IS NOT AoIP!

Ethernet is not AoIP. Probably the biggest point of confusion for those seeking an AoIP solution is confusing Ethernet with IP, since they are nearly inseparable. But there is a significant difference between technologies built on the IP layer (layer 3) and those implemented directly on the Ethernet data link layer (layer 2), or even the Ethernet physical layer. The biggest difference is that non-IP technologies are not routable, significantly limiting their usefulness.

For example, none of these otherwise great technologies are AoIP, even though they use an Ethernet transmission link:

» Cobranet – legendary for its time and niche, but it is an Ethernet layer-2 technology

» AES50 – Ethernet layer-1

» Ethersound – Ethernet layer-2

» AVB – Ethernet layer-2

AVB is not AoIP. Surprised by this? Even with the understanding that AVB is not AoIP, it is often touted as a possible replacement, or even a superior technology. While AVB is a great advancement in the area of audio streaming, designed to deliver high-resolution audio at a guaranteed low latency, it's a pretty tricky proposition for the system engineer willing to build an AVB application.

AVB claims to be an inherent capability of standard networking equipment, implying that, unlike AoIP, AVB would not require an engineered network. But when you look closely, this seemingly huge advantage starts fading.

Support of IEEE 802.1AS (subset of IEEE 1588) is mandatory in all switches and terminal devices. Support of IEEE 802.1Qat, the Stream Reservation Protocol (SRP), is mandatory too. Since AVB traffic cannot be routed through network equipment that is not AVB-enabled, it effectively still requires an engineered network – at least in terms of careful selection of the switch models (until such time as AVB is included in all industrially made switches).

And even with all AVB-enabled network equipment in place, significant restrictions remain. AVB is layer-2, not routable, so the audio traffic cannot pass the boundaries of LAN subnets. Don't even think about WAN links. And only 75% of the bandwidth can be reserved for actual AVB traffic. While generally this kind of a restriction makes sense for mixed-application networks, it robs 25% of the bandwidth from a dedicated audio setup.

WHAT'S IN AoIP'S FUTURE?

No one can say for sure. But there are some things peeking over the horizon that are worth mentioning.

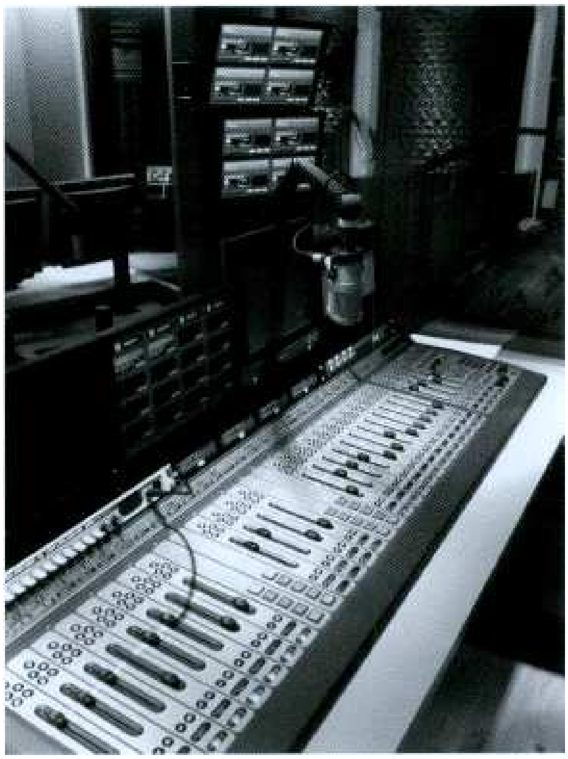
INTEROPERABILITY

The value of a networking technology is only as great as the number of devices it allows to interconnect. Users and manufacturers realize this – several interoperability initiatives have appeared, and some of them have already produced useful results. These initiatives indicate a new level of AoIP evolution.

EBU – ACIP project: The *Audio Contribution over IP (ACIP)* project lead by the network division at EBU, is perhaps the earliest major initiative that has already produced practically useful results. Recognizing the fact that more and more equipment is using IP links for audio contribution, EBU launched a project group in 2006 to develop minimum interoperability requirements for devices interconnected via IP. Agreement on a common standard was reached in September of 2007. The standard is based on a number of IETF documents, in particular RTP over UDP for audio streaming, and SIP for session management. It also defines a number of mandatory audio codecs to be supported by all compliant devices, as well as a number of recommended codecs. The project proved to be successful – most of the manufacturers taking part in the plug tests in 2009 appeared compatible.

AES – X.192: The effort started by the EBU ACIP project is logically continued by the AES X.192 high-performance AoIP interoperability project, attempting to reconcile the existing in-facility AoIP networking solutions. This project has chosen an approach to identify an area of overlap, where the different technologies might be able to interoperate, hopefully with no, or minor enhancements. A task group, of which Axia is a member, has been created and is working on a draft document. More information can be read at www.x192.org.

NRJ Networks, Paris



World's largest AoIP mixing console? Likely. This 36-position, 28-fader Axia Element is in the master control room of NRJ Networks, Paris.

WIDE-AREA AoIP NETWORKS

Network bandwidth and latency, and (to some extent) the processing power of network devices are the main factors limiting AoIP applications. But these factors are also the ones we can observe evolving quickly and steadily. The cost of processing power is reasonably low, and keeps dropping, allowing network devices to become faster as well as functionally more capable. Huge amounts of bandwidth are added to backbone links worldwide every month; more and more connections get upgraded to high-speed copper and optical links, and new wireless technologies cover low-density areas. This stimulates services that were originally conceived for use only on a LAN to break the walls of the closed networks and go out to the WAN.

MPLS (Multi-Protocol Label Switching) is a technology that helps greatly here. Although MPLS doesn't create new bandwidth, it allows utilizing WAN bandwidth for LAN protocols and applications without any redesign of the latter. It is capable of tunneling Ethernet frames to transparently link distant clusters of a virtual LAN.

The Virtual Private LAN Service (VPLS) is well known and widely deployed, but has yet to find its way into broadcast applications. When backbone networks build up sufficient bandwidth, AoIP networking over VPLS may become an extremely attractive alternative to today's point-to-point compressed audio contribution links. As the capacity of the networks grows, a natural trend will be to allocate more bandwidth to audio links, allowing the use of codecs with less aggressive compression. As this bandwidth expansion cycle repeats, we will eventually achieve the ultimate goal of sending linear audio over WAN links.

Internet2 is a major initiative led by the research and education community. The Internet2 consortium comprises over 200 US universities, as well as corporations, government agencies, and national research and educational organizations from over 50 countries. The consortium operates an evolving high-performance network that currently spans the US. It provides network capacity for educational, research and community services, as well as actively engages the community in the development of new technologies and Internet applications.

The capability of data transport at multi-gigabit rates over long distances is opening doors for advanced networking applications in arts and education, including live music performance over networks, interactive sound production, teaching, and many other areas.

A SIDEBAR: usually the discussion goes about how technical limitations narrow the artistic quality. However, it appears that the networking phenomenon that has become an important part of our everyday life is provoking new creative ideas itself. It allows, for example, a group of performers to collaboratively play music on a networked instrument, or exploit network delay or packet jitter as "sound art". Also, special music is being composed for networks, assuming a certain delay between the participants of the ensemble, in which case the network delay becomes an inseparable part of the musical composition. To find out more, simply search the Web for "networked music performance", or related keywords, and read on!

WHAT'S NEXT?

No one knows the future, but there are, really, three fundamental things we would wish to receive from networking technologies in the coming years:

» Complete transparency to the native resolution of the source material

» Complete transparency to application functionality

» Zero latency (Ok, let's be realistic – close to zero)

The good news is that, although it sometimes may be costly, none of these wishes is fundamentally impossible to realize even within the scope of technologies known today. It may be that, one day, processors will be inexpensive and fast enough to satisfy any fancy application – devices easily handling message rates at the audio sampling frequency and higher. Imagine the globe, wrapped in a lightning-fast ultra-broadband protocol-transparent carrier network – no link congestion ever, network nodes forward the traffic instantly, and the packet overhead is no more an issue. Right now it's a dream, but certainly not an impossible one.

And after that? Well, the electromagnetic field is slowish. Travelling half the globe in an optical fibre takes about 100ms. Can this be improved?

An old legend from the early years of the 21st century was telling: "AoIP will never work."...

Gints Linis
Research & Development Project Manager
Axia Audio

YOU ROCK!

TO ALL OUR 3000+ USERS WORLDWIDE.

METROPOLITAN OPERA, NEW YORK
AGORA RADIO, PONZAN POLAND
96FM, PERTH, WASHINGTON
WAY-FM, NASHVILLE
GOOM RADIO, PARIS
SAGA COMMUNICATIONS
DAVE RAMSEY STUDIOS
VOICE OF NASCAR, DAYTONA BEACH
SIRIUS XM RADIO, TORONTO
...

MINNESOTA PUBLIC RADIO, ST. PAUL
RADIO AMERICA, HONDURAS
HOPE 103.2, SYDNEY
UNIVISION, HOUSTON
I105-107, ATHLONE, IRELAND
AD LABS RADIO, MUMBAI
POP-FM, CORFU, GREECE
EMMIS RADIO, ST. LOUIS
LEO LAPARTE, THE TECH GUY
...

R†DIO EXPRES, BRATISLAVA, SLOVAKIA
UNITED STATIONS NETWORK, NEW YORK
IDC RADIO, ISRAEL
FAMILY RADIO, TAIWAN
JOURNAL BROADCAST, MILWAUKEE
STUDIO ECOLE, FRANCE
NRJ NETWORKS, PARIS
RADIO SUDITROL, BLOZANO, ITALY
WZLX-FM, BOSTON
...

RADIO FREE EUROPE, PRAGUE
KISS-FM, TAIWAN
KRTU-FM, SAN ANTONIO
CMT RADIO LIVE, NASHVILLE
RETRO-FM, MOSCOW
KWVE-FM, LOS ANGELES
WSTR-FM, ATLANTA
WRLT-FM, NASHVILLE
NEWCAP RADIO, NOVA SCOTIA
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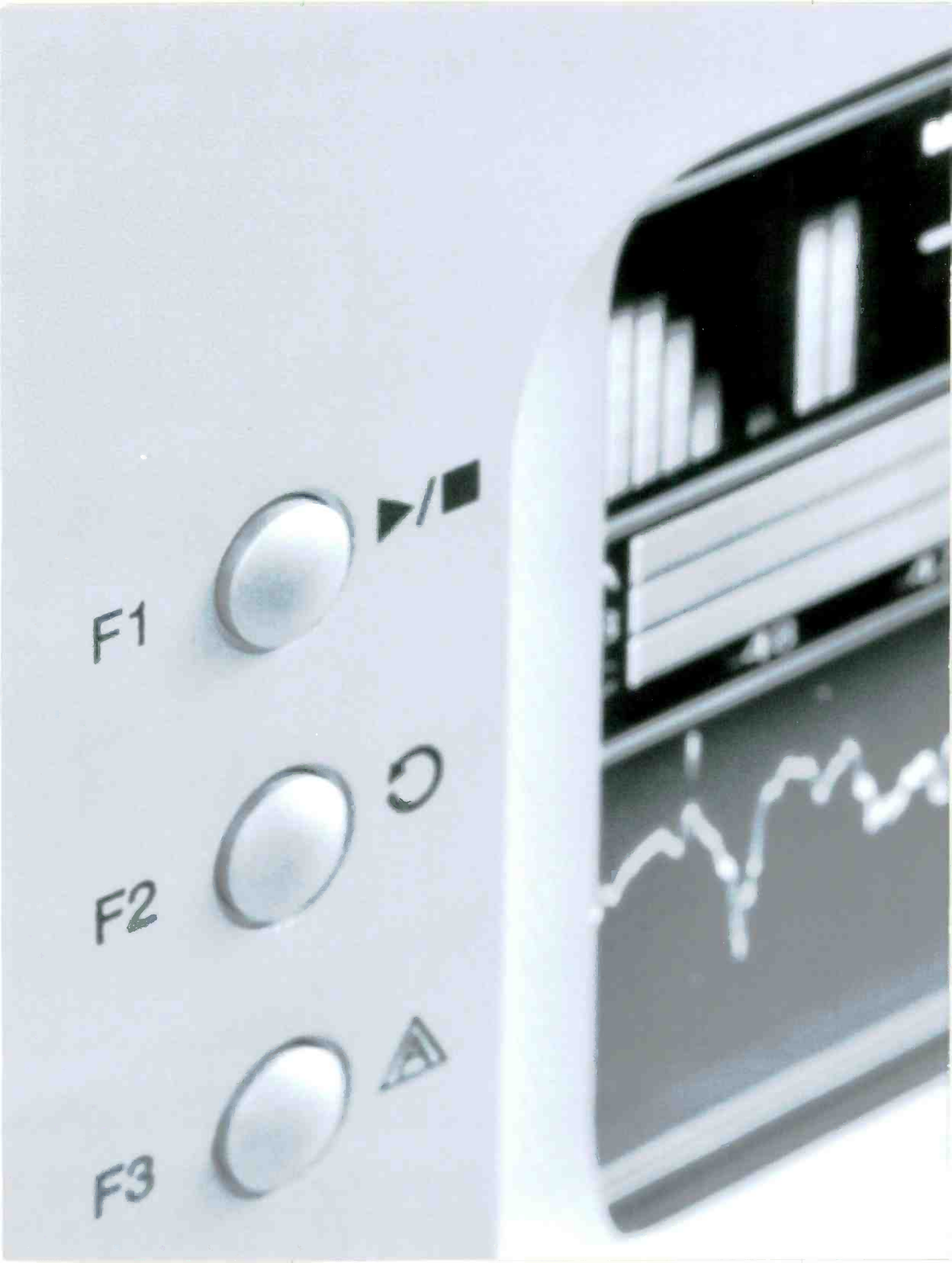

F1
F2
F3

LINEAR ACOUSTIC

Viewers are listening.

That's what it really comes down to.

The advent of digital may have brought whiplash transformation to television broadcast engineering but something else changed along the way - audience expectations.

Viewers are more discerning.

Linear Acoustic products and technologies are designed to manage audio and loudness, help you maintain compliance, upmix, downmix, encode, decode, meter, monitor every audio function along the path from production to transmission intuitively.

But our primary goal is audio quality. Our focus remains on helping broadcasters world wide deliver compelling, engaging audio that is naturally compliant because it satisfies viewers.

Viewers are listening.

Christina Carroll
SVP, Global Sales, Telos Alliance

LINEAR ACOUSTIC

COMPANY HISTORY

AUDIO UNDER CONTROL

Audio has been my passion for as long as I can remember, and growing up in the NY area fed nicely into this desire. From repairing headphones in grade school (long before truly understanding what I was doing, and only occasionally bringing them back to life), to being the chief engineer of my high school and college radio stations. It was in radio that I developed an insatiable curiosity about broadcast and audio processing in particular. Thankfully, I had several very patient teachers that grimaced and looked the other way when I was "understanding" stuff using the disassembly method of learning. Some of it actually made it back together, and in hindsight, luckily some of it did not.

TO DOLBY AND BEYOND

Joining Dolby Laboratories in 1995 was a seminal event. There was not a broadcast group per se, but the market for digital sound on film was growing and engineers were needed. Spending almost four years on the film stages of New York delivered in-depth experience with the production of matrix and discrete surround audio and with the brand new DVD format which specified the Dolby Digital (AC-3) codec for multichannel audio.

Almost simultaneously, the AC-3 codec was mandated for use in the ATSC digital television standard, and soon thereafter was included in the DVB specification. The rush was on to develop the tools for this new format.

Being walking distance from the major television networks in New York allowed me to experiment with early DTV audio products in some of the worlds most advanced broadcast facilities.

Next stop was Dolby's San Francisco headquarters to take on the role of the professional audio product manager. Here a rogue team of engineers and coding experts developed a set of products that laid the foundation for broadcasters to transition to digital video and audio and from mono or stereo to 5.1 channel surround. Never in the history of the company had so many products been developed in so short a time, but it was necessary as we were not just adding more channels but changing the entire path from production to consumer. The dream was that Hollywood audio quality could be delivered to the home via broadcast, and for the first time transmission methods would not get in the way.

There were two problems with this dream. First, it required everything to be in place all at once to make it work seamlessly. This might be practical in the lab but reality dictates it will work out otherwise. Second, there seemed to be so many places where things could go wrong and it was becoming very obvious that some sort of overall protection was necessary.

The time had come to take my bag of collected tricks and hit the road as a consultant to try to help broadcasters and manufacturers take the next steps. However, it quickly became apparent that the technology to avert a potential train wreck would have to be homegrown.

YEP, WE STARTED IN A NJ GARAGE

Romantic, isn't it? Actually, Linear Acoustic was started in the basement of the house I grew up in and expanded into the garage (and the dining room, living room, and at least one bedroom). It also consumed a good deal of space at Leif Claesson's house in California where he turned our good ideas into algorithms. Interestingly, we were never on the same coast during the entire development but overnight delivery service and the Internet made us feel like we were in the garage together.

Once we finished the initial development of the first Linear Acoustic product called the OCTiMAX 5.1, Leif and I showed it at the SMPTE convention in Pasadena, CA. Thankfully, we caught someone's attention.

That someone was Steve Smith, the venerable engineering leader of Liberty Corporation and he was tasked with transitioning his television stations to digital. When we informed him that we were working on a DTV loudness controller, he proposed that if we could make digital television audio as hands-off as it was in analogue and still preserve the quality that he would outfit all sixteen of his television stations. Steve and Liberty became our first customer.

We were working on a shoestring budget creating products that were being installed by some of the top US television broadcasters as they began their transition to digital. Every unit was hand-assembled and carefully tested using tools that are common today, but were new to the industry then. As with any brand new product, there were bugs, but most of ours had four or more legs and were removed with compressed air.

AND THEN WE MOVED TO PA

Soon we outgrew the garage and moved to Lancaster, Pennsylvania to enable us to bring on some additional engineering talent and to take advantage of easier access to better quality high-tech manufacturing vendors.

In Lancaster, we began R&D that resulted in the first ever AC-3 splicer (and they said it couldn't be done), along with a higher density audio transport system called e-squared was used on such high profile events as the Academy Awards and the Country Music Awards broadcasts. We also innovated some metadata tools and a really slick audio and metadata monitoring system.

In 2008, we proudly joined Steve, Frank and Mike to become part of the Telos Alliance.

Today, the end of analogue over the air television is a reality in the US and is in process internationally, and loudness problems are rampant. Sadly, this was predicted.

Thankfully, we are amidst the continuing release of new and useful products that are the culmination of our work since the beginning of Linear Acoustic. Our approach supports industry efforts to solve loudness problems by working on each stage of the chain rather than just slapping a "cruncher" at the end. We also have the leading stereo to 5.1 channel upmixing technology called UPMAX and have recently released a new flavor called UPMAX II to critical acclaim. New software tools allow for the most advanced file-based audio processing available.

AND THEN WE WON AN EMMY

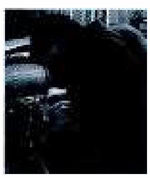
We are incredibly humbled and honored to have been presented with a 2010 Technical Emmy® award for "The Pioneering Development of an Audio/Metadata Processor for Conforming Audio to ATSC Standard" (whew).

We take this honor very seriously and recognize the importance of remaining active participants in standards creation within the ATSC for DTV and Mobile DTV, SMPTE, and the EBU to continue to make audio even better.

In addition to having some of the best ears in the industry for broadcast audio, our ears are also sensitive to your feedback and suggestions. Our products are based on direct suggestions (or commands) from customers.

Broadcast is in our blood: it is what we do, it is what we love to do. It is what links us to you, our customers and our colleagues. Viewers ARE listening, and so are we.

TIM CARROLL
FOUNDER AND PRESIDENT, LINEAR ACOUSTIC



DELIVERING QUALITY SOUND

A FAREWELL TO LOUDNESS

Thank you, loudness. Thank you for your spikes, sudden bursts and consistent inconsistencies. Thank you for transforming serene, satisfied television viewers into an angry, ear-cupping citizenry, banding together to complain to their governments. But most of all, thank you for finally causing the broadcast industry to bolt upright and recognize a seemingly obvious truth: TV is more than simply a picture. And for that, loudness deserves some gratitude.

By reminding TV broadcasters of the importance of audio, loudness, in a way, has made TV quality better than it might have been.

When broadcast engineers focus on the real audio issue – delivering consistent quality sound – loudness becomes moot, consumers are contented, regulators draw back, producers can be satisfied, and broadcasters are happy.

It is also important to remember that while it is easy to blame loudness issues on commercial advertisements or other interstitials, it can also be the fault of the programs themselves. A train crash and explosion might be fine during a matinee but it is not going to go over very well at 3 in the morning with kids asleep.

PLEASE MORE PEOPLE, MORE OF THE TIME

Everyone wants your audio signal to be perfect, at least to their expectations. Who requires it the most? Is it the regulator? The station manager? The program producer? The consumer? In reality, it is all of the above but for different and sometimes opposing reasons.

Each of these targets has their own benchmark for satisfaction. The station wants happy viewers and a happy regulator. The regulator wants no complaints from viewers.

Ultimately then, the final judge is the viewer. It is the viewers who create station ratings and thus a place for paid advertising to be shown which generates revenue to buy programming and pay staff. It is the same viewers who will complain to the regulator when the audio is not right - especially when there are unexpected loudness shifts.

The viewer wants consistent audio, then they will not complain to the regulator and the station is likely safe. How this is accomplished, however, may not satisfy the program producer.

COMPLIANCE OR QUALITY?

Is it possible to regulate an audio signal to the point of being unlistenable? For some governments, nothing is impossible. Although regulators may specify both a loudness target and the method for measuring loudness, they will likely not react unless they receive viewer complaints. However, a regulated target and a metering specification, if approached blindly, may result only in the meter being happy.

Just like in the departed or soon to be departed days of analogue, devices can be installed at the end of the chain prior to transmission that raises or lowers gain depending on how much the loudness of the incoming audio varies from the target. This is commonly referred to as Automatic Gain Control (AGC). AGC systems will more or less treat every shift similarly and will affect the good and the bad. Everything gets a little something whether it needs it or not.

While there are many sophisticated (and some unsophisticated ways) to accomplish AGC, in reality there is no way for any machine, regardless of manufacturer, topology, or promise of magic outcome that can, in real time, know the difference between a good, intentional loudness shift and a bad, annoying loudness shift. Certainly human generated commands and even metadata can be used to change or bypass processing for content that is believed to be good, but this involves a great deal of effort that has proven thus far not to have been exerted. Television mixers have long been used to the idea that what was transmitted via analogue means would be different than what they created and that was just the way it was. In today's digital world, there is no technical reason why before and after cannot match. It happens when films mixed for the big screen are then transferred to DVD and the same audio coding system is used for 5.1 channel broadcasts.

TAKE IT IN DOSES

Since machines cannot automatically know the difference, a better approach is to separate the problem into smaller tasks: matching average loudness, managing transitions, and delivering appropriate dynamic range.

Taken separately, a much better result can be obtained. For example, to match the average loudness of programs, use of BS.1770 along with either manual or automatic control of the meter based on an anchor element such as dialogue, allows the overall average to be measured and then a simple overall one-time level scale to be applied so the target is achieved. This changes nothing about the content and preserves the intent of the producer.

Matching average loudness of different pieces of content does not, however, solve jarring transitions. These occur at program boundaries and are the result of a mismatch of short term loudness at the junction between the end of one piece of content and the beginning of the next.

Think of a program whose average loudness measures at one level and a commercial advert that measures at another level. If the averages are matched by scaling their loudness, then on average they will sound equally loud. However, if a dramatic program is ending with a quiet death scene (as they often do), and compared to average is quiet for the 60 seconds leading up to the advert, guess what is about to happen? Yep, the advert will seem too loud. Guess what else? Meters will be totally happy since they are looking at longer term averages.

To oversimplify things, this is in fact similar to a dynamic range issue. It is not the same dynamic range variation as a loud train crash or gun fight in an action adventure movie, which is expected, but is instead an artificial variation. In fact the difference in loudness in this case may be much less than the gun fight. However, it is perceived as much worse. Likely this is because disparate elements are being glued together not for artistic reasons, but for financial reasons. The commercial must play at a certain time, per contract, whether or not it matches the program that surrounds it. How can this be captured by any meter? So far, it cannot be, at least not in real time.

One way to fix this is to use the AGC techniques described earlier which to minimize variations. Again, this will keep the viewer and the meter happy, but the program will have been irreparably changed and the producers will probably not be thrilled with the outcome.

The other way is to capture it in the program delivery specification. Offering typical ranges for programming and examples of what might happen at program/commercial boundaries will enable mixers to take control of the situation and make better artistic choices than any machine could make.

It is also worthwhile to refer mixers and program producers to recommended practices that offer guidance on speaker calibration in mix environments. Interestingly, monitoring at levels closer to what a typical consumer might listen at results in mixes with more appropriate dynamic range. Since the difference between average loudness and background noise in the mix room is now smaller, the dynamic range of the program must also be reduced.

SUMMARY

The intention of regulators is to satisfy their human constituents. Meters and loudness targets are well intentioned but when relied upon as the sole arbiter of compliance often lead to content that while consistent may be overly so. Like gravy without the occasional lump, the excitement of variance is gone. The trick is to preserve the good variance and manage the not so good variance.

To make this work requires more effort from broadcasters and program producers. Absent metadata systems to manage all of this, broadcasters must supply accurate and achievable program delivery specifications and producers must take into account the typical viewer and what may or may not be appropriate to deliver to them. Since broadcasters may be legally required to satisfy viewers, if content does not fit, they may be forced to make it fit. If both sides know the rules and have goals that are realistic and mostly aligned, it is possible to achieve acceptable balance.

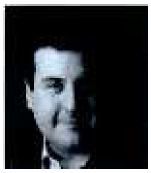
The guidelines for making it all work boil down to one rule we should all post on our walls: don't upset the viewer stupid! This is where it all begins and ends. No complaints=happy regulator=happy broadcaster. The challenge then is to manage the cost of this satisfaction versus preservation of content. It can be done.

If not, there are always AGCs that can smooth out everything. Everything.

Consistent quality sound, delivered with perfectly tuned images. This is what makes great television for program producers, regulators, consumers and ultimately, broadcasters.

And a final hat tip to loudness, without whom we may never have made it to this point.

TIM CARROLL
FOUNDER AND PRESIDENT, LINEAR ACOUSTIC



AERO.air

AUDIO/LOUDNESS MANAGER

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

AERO.air® is audio purity for digital television.

It provides proven loudness control, decoding and encoding, and unmatched upmixing capabilities. Factory presets ensure audio quality and easy set-up, while experienced users will appreciate extensive access to individual controls. Adjust the AERO.air for wideband multi-stage processing, multiband multi-stage processing, or anywhere in between.

AERO.air accepts 5.1-channel and two-channel audio via included AES or HD/SD-SDI inputs, plus dedicated EAS/Aux bypass inputs. Audio is then processed by the multiband, multistage ITU-R compliant AEROMAX® loudness cores resulting in smooth audio with appropriate dynamics. Two-channel audio is automatically upmixed producing a consistent surround-field, perfectly downmix compatible for all stereo viewers.

If present, audio metadata will manage upmixing and improve loudness control while minimizing impact on source audio. Extensive fallback options enable the AERO.air to compensate for missing or incorrect metadata.

Industry standard two-channel to 5.1-channel upmixing is provided by the Hollywood approved UPMAX® and UPMAX II algorithms. AutoMAX-II provides automatic and GPI or metadata guided control of upmixing without risking loss of center channel dialogue.

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 ten main AES inputs and outputs and front panel headphone connector. HD/SD-SDI I/O, with or without compensating video delay, enables de-embedding and re-embedding up to 16 channels of audio plus SMPTE 2020 (VANC) metadata. All AES outputs remain active when SDI option is enabled. Embedded channels can be routed through or around processing. Encoded signals can be de-embedded and re-embedded.

A bright color display, large rotary encoder, and four control keys provide simple menu navigation and adjustment. The AERO.air can be controlled remotely via GPI/O, while Gigabit Ethernet allows TCP control by automation systems.

The AERO.air contains dual redundant power supplies and hard relay bypass for the digital audio, SDI, and metadata signals, necessary in mission-critical applications.

Software options can generally be added in the field. Hardware options such as Dolby encoding or decoding and video delay must be factory installed.

OPTIONS

NIELSEN OPTION:

Generates revenue critical NAES II and Nielsen Watermark audience measurement codes. AERO.air precisely inserts these signals for maximum code recovery – after audio decoding and processing and before transmission encoding.

DOLBY DECODING OPTION:

Allows reference quality decoding of Dolby Digital (AC-3), Dolby Digital Plus (E-AC-3), and Dolby E content from any AES or SDI input signal.

DOLBY ENCODING OPTION:

Two Dolby Digital (AC-3) and/or Dolby Digital Plus (E-AC-3) encoders for 5.1 plus stereo audio.

AERO.AIR IS AVAILABLE IN TWO VERSIONS:

AERO.AIR (DTV):

AERO.air (DTV): Provides 5.1 channel loudness control and upmixing and outputs full-time 5.1 plus a stereo downmix.

AERO.AIR (5.1):

AERO.air (5.1): Supports full 10-channel 5.1 + 2 + 2 and 5 x stereo modes, and includes dual upmixers and CrowdControl™ dialogue protection.

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 audio perfection 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.

AERO.one is well suited for main or backup transmission paths, providing high quality audio in a feature rich and cost effective manner.

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 from two channel up to dual 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 IS AVAILABLE IN FOUR VERSIONS:

AERO.ONE (16) – 2+2+2+2 through 5.1+2+5.1+2 channel loudness control plus quad UPMAX upmixing engines and dual downmixed outputs.

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

AERO.ONE (OTV) – 5.1 channel loudness control plus upmixing and downmixed output.

AERO.ONE (TV) – 2+2 dual stereo programs plus SAP and CrowdControl™. CrowdControl dialogue protection is vital for sports broadcasts, ending the annoying dialogue loss that occurs in mixes rich in sound effects.

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.

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 CARBON™ Hybrid 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 CARBON™ Hybrid Processing is a patent-pending hybrid between 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 flexibility. 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.

LOUDNESS

THE LISTEN TEST

Audio loudness processing is becoming an increasingly important part of the broadcaster's job as regulatory imperatives and sophisticated listeners demand more consistent audio levels. Not doing so can risk fines from regulatory agencies and drive viewers to competing providers that have their audio under control.

There are many options when purchasing audio processors from manufacturers worldwide. Any broadcaster that is about to put out their hard-earned money for a processor needs to understand the differences between competing techniques and claims, and most importantly how to evaluate equipment from different manufacturers.

A logical place to start is with an understanding of loudness measurement techniques and (where applicable) the regulations that reference them. The ITU-R BS.1770 measurement standard is the basis for loudness regulation in the US and Europe, as well as other technical regulations. One point of confusion about this standard is that some processing manufacturers claim they are "BS.1770" compliant, even though BS.1770 is a measurement, and not a processing, standard.

One of the key points of BS.1770 is that loudness is a quantity that is integrated over a certain time period. While shorter integration times can be used, it is valid to measure entire programs (with an infinite integration time that is reset between programs). Unlike PPM and VU level meters, instantaneous loudness measurements have little utility.

This is done for a straightforward reason: audio has dynamics. Digital audio broadcasts provide a much wider usable dynamic range than did analogue transmission standards. The average loudness level of audio content can be measured over time using a BS.1770 while still preserving this dynamic range.

Some broadcasters who have evaluated loudness processors are surprised to see the output loudness level moving. They are especially sensitive when there are imperatives that give target loudness levels. Yet these target loudness levels are specified over an entire program or even an entire broadcast day for a TV station. Within any normal program there are going to be variations around the target loudness level. This is to be expected, since the long term or average loudness is what is important.

The type of program content currently being processed will greatly affect dynamic range as well. Certain types of content - for example, some sports with constant crowd noise and near-constant play-by-play announcers - has relatively constant levels. Other types of content, such as classical music has wildly varying dynamics. Program producers edit their program audio with louder and softer portions for dramatic effect. These same producers will often object if the peak to average ratio of their programs is drastically altered during broadcast. Listeners with higher-end audio reproduction systems will also notice more limited dynamics.

So what is best when processing such varying types of content? As a general rule, preserving dynamics is beneficial. Using loudness processors with a range of preset conditions greatly simplifies set up for the broadcaster. Linear Acoustic Audio Loudness Managers have a range of presets that allow adjustment of all processing parameters. Included in this list are presets that minimally alter the peak-to-average ratio of content while still achieving target average loudness values. Of course there are also "denser" presets that have considerably lower peak-to-average ratio.

Both of these (and all of the other processing presets) will produce audio with long term loudness at the target level. Many other devices from other manufacturers will do so as well. So what are the differences between loudness managers from Linear Acoustic and others?

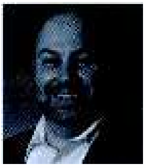
While the very definition of an audio loudness processor requires that the output measure correctly on a meter, it is not nearly enough for that to be the only function. The other critical aspect of a processor is that... it actually sounds good while maintaining loudness. No meter yet invented can convey "sounds good".

The ultimate test of any audio processing device is a listening session using the best audio evaluation tool in existence, our human ears. A near infinite list of subtleties exists when processing signal content with the dynamics of broadcast audio. Many manufacturers claim to have mastered these subtleties, but a few listening tests almost always provide a different story. Imagine a ridiculously extreme example: a loudness processor could consist of a gain circuit and a clipper. While it would stick the loudness meter right on the target value, it would be completely unlistenable.

Sadly this technique is not too far off from some other manufacturers' current product offerings. Other products are simplistic wide band gain control circuits that completely ignore the complexities of the human ear. Fletcher and Munson taught us in the 1930s that our human ears are not at all equally sensitive to stimuli at different frequencies, and that this response varies with sound pressure level. Amazingly some designers of audio loudness processors seem completely unaware of this work done almost 80 years ago.

So evaluating a loudness processor actually turns out to be a fairly straightforward process. First, ensure that the candidate processor will produce the target loudness level over a long time frame, realizing that there will be short term variations about the target level due to normal audio dynamics. Almost every existing processor will achieve this. The next step is quite simple but will be the real test: listen to the processed outputs with a wide range of content. Chances are our human ears will be more revealing than the most in depth data sheet.

Mike Richardson
Director of Products



AERO.file

AUDIO/LOUDNESS MANAGER

AERO.file® brings proven audio technologies developed by Linear Acoustic to the file-based domain. Developed in partnership with Radiant Grid Technologies, AERO.file eliminates the need for custom hardware and integrates audio processes into existing systems and workflows.

Tools can be used for ingest, managing existing libraries, conforming content for different playout services, or any combination.

In the TV audio process, upmixing and loudness range control tools have proven most effective in the file domain. Advanced RadiantGrid transmuxing and transwrapping enables the audio essence to be extracted from a host of popular file wrappers, measured, scaled, and processed, then re-wrapped 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.

Operator controls are simplified to choices of loudness target, whether to use 2-channel to 5.1 channel upmixing and whether to use loudness range control.

WHEN SIMPLE SCALING IS NOT ENOUGH

Whether anchor-based such as with Dolby Dialogue Intelligence™ or overall average with the relative gating methods of EBU R128, scaling aligns the anchor or overall average of content. This can easily be imagined when considering how to match a commercial advertisement with a program filled with dialogue and explosions - what do you match with what?

Sometimes programs have a loudness range, that while appropriate for a movie theatre or a premium channel, is challenging to re-purpose for delivery on other channels and especially mobile services.

This is where sophisticated loudness range management techniques can be employed. Once scaling is applied to the program, the job of loudness range control (LRC) is dramatically simplified and the effects are minimized.

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.

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. Any LQ-1000 can be updated via software to incorporate this feature.

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, HE-AAC, MPEG-1, Layer II, 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 » HE-AAC, MPEG-1, Layer II, 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 (option) » 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.

LAMBDA

PROFESSIONAL DIGITAL AUDIO AND METADATA MONITOR

LAMBDA™ is the ultimate digital TV broadcast audio monitor. Designed specifically for the specialized needs of the modern TV station, LAMBDA combines a unique understanding of audio and metadata through the entire broadcast chain from production to consumer.

LAMBDA displays and reproduces up to sixteen audio channels via AES or HD/SD-SDI input, and accepts industry standard professional audio metadata via 9-pin serial input or by extracting it from the vertical ancillary (VANC) space of an applied HD-SDI input. Audio and metadata are displayed and properly combined to allow for accurate monitoring. A utility audio delay is included to allow up to three frames of compensation for external video monitors.

Any channel, channel pair, or downmix can be monitored through internal speakers, via the exceptionally dynamic front panel headphone output, or from the rear panel balanced analogue stereo and AES digital output. A new 16-channel mode allows all applied audio channels to be displayed simultaneously and reproduced individually or as a 5.1 downmix.

High-excursion full range drivers with aluminum cones are coupled with metal dome HF drivers in an acoustically tuned enclosure to optimize frequency response and power handling. Digital Linkwitz-Reilly style crossovers are combined with low distortion, high efficiency class-D power amplifiers for exceptional audio quality and loudness.

Loudness metering per the ITU-R BS.1770 standard is also included. In addition to a numerical readout, a thin line indicating measured loudness "floats" over audio metering to allow quick verification of program loudness.

WHEN BROADCASTERS SPEAK, WE LISTEN

Having the UPMAX with us in Beijing this past summer was like adding a new friend to the crew. The sound was a nice improvement, and the support from Linear Acoustic was superb. We are still learning about new ways to use UPMAX, and I look forward to using it and working with Linear Acoustic again in Vancouver.

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

The LAMBDA is a top-shelf piece of gear. It is definitely the future of broadcast facility audio monitoring.

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

The Linear Acoustic AERO.one is definitely one of the easiest-to-set-up pieces of audio processing gear I have ever experienced. Plus, it sounds great with little or no effort. Having been in the business for 40-plus years, I have seen my share of audio processing and this unit, by far, is my favorite. It "fixes" the levels the network sends us in a very pleasant way and makes the viewers very happy.

Tom Bondurant
Director of Engineering
WAPT-TV

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

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