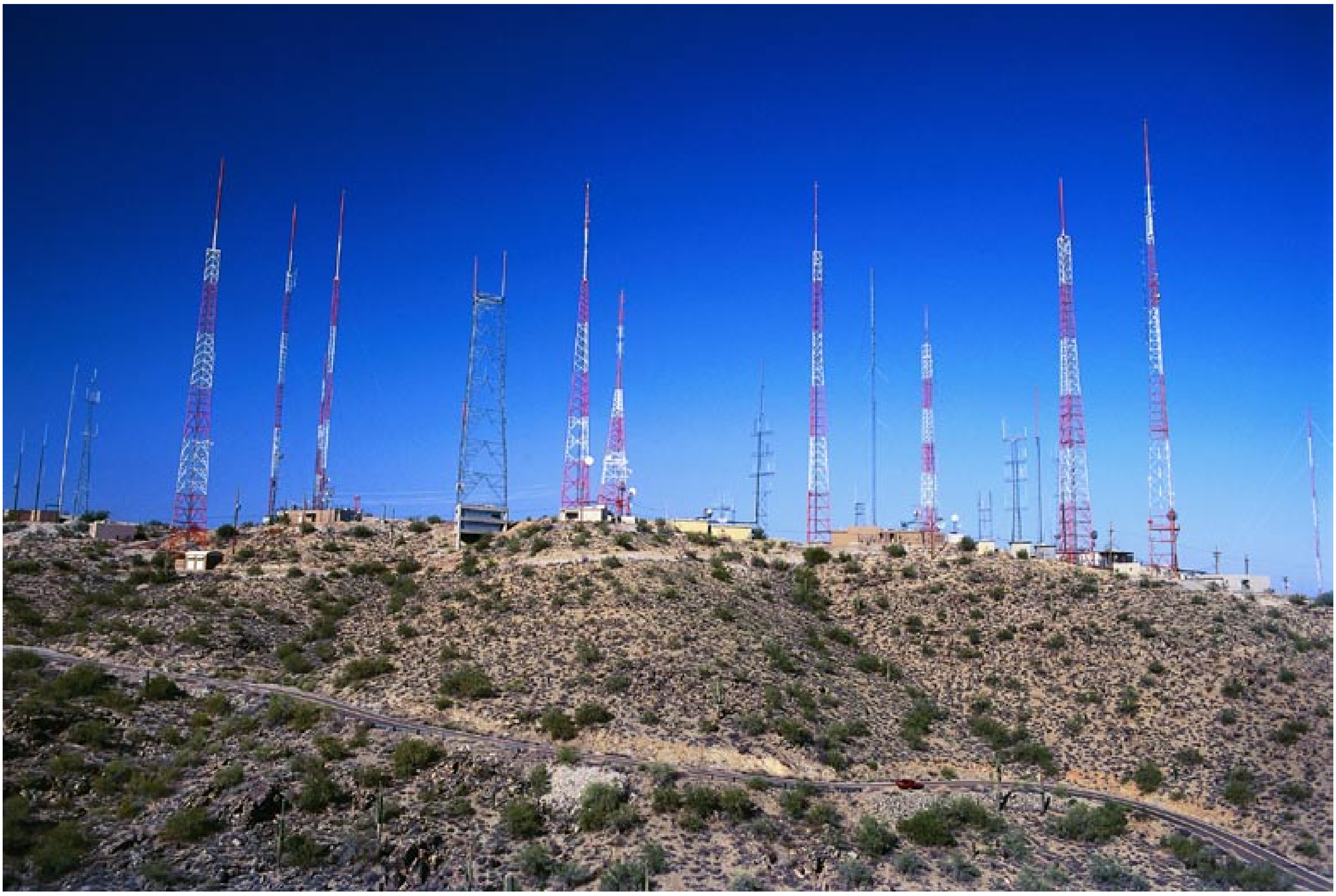
Linear Acoustic

Audio under control.

Product Guide Fall 2009



Welcome

Audio under control

Audio has been my passion for as long as I can remember. 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. Luckily, 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.

Another event that would come full circle many years later was the launch of a certain radio station in New York. I caught the inaugural summer broadcast and was amazed at how that station seemed to launch off the dial. Little was I aware of how this would play out, but it did draw me to the broadcast center of the US: New York City.

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 best broadcast facilities.

Next stop was Dolby's San Francisco headquarters to take on the role of 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, no one ever checked to see if the majority of consumers really want or need the additional complexity. There seemed to be so many places where things could go wrong and the idea that some sort of overall protection was not being created began to seem ludicrous.

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, we showed it at the SMPTE convention in Pasadena, CA. Thankfully, we caught someone's attention.

Steve Smith was 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 analog 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.

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 high quality and lower cost 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 StreamStacker. We also innovated some metadata tools and a really slick audio and metadata monitoring system.

Sales were increasing and we were feeling the increasing demand for DTV audio solutions. The ever-present question on all of our minds was: "Can we keep up?"

Meeting of the Minds

In 2006 it was time to look at the next generation of television audio processing solutions and to see if there was some synergy between us and an established processing company called Omnia, headed by none other than the guy behind that NY radio station I mentioned before: Frank Foti. Frank had merged his processing company, Cutting Edge, into Telos, a company started by Steve Church. They re-branded Cutting Edge into Omnia and released the world's first DSP-based radio processor. By the way, way back when I was at Dolby, Frank and Steve were also the first to demonstrate to me an idea for a television processor that could be based on metadata. Hmmm, what an interesting idea....

At the NAB show in 2007, we announced that our collaboration would in fact be a merger of Linear Acoustic into the Telos group of companies. Certainly we had been propositioned by other companies over the years, but the fit was never right. However, coming together with Frank, and Steve, and Mike Dosch (president of Axia) was a natural next step. We had found so many areas where we complemented each other that the idea immediately clicked.

Today, the end of analog over the air television in the US is a reality and loudness problems are rampant- as 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 and from my time at Dolby all sprinkled with a healthy dose of the expertise from Frank, Steve, and Mike. 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 of the chain. We also have the leading stereo to 5.1 channel "upmixing" product and licensable technology that stands alone as also being perfectly stereo compatible, which explains why it is the choice of top broadcasters worldwide.

Together, the Telos companies have the largest R&D group in the world focused solely on audio for broadcast. In addition to having some of the best ears in the industry for audio, our ears are also sensitive to your feedback and suggestions. Most all of our products are based on direct suggestions (and sometimes 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.

Tim

Tim Carroll

President, Linear Acoustic

Linear Acoustic

Audio Under Control.

From post production to distribution and encoding for transmission to consumers, Linear Acoustic products solve problems, provide missing links, and simplify and solidify the process of delivering HD audio with HD pictures.

Controlling Digital Television Audio Issues

Have to Admit It's Getting Better

Television audio has reached an interesting place. It is currently wedged somewhere in between the low-impact, narrow range stereo or mono audio of yesterday and the wider range, higher-impact audio found in movie theatres, on DVD, Blu-Ray Disc, and streamed content. The former seemed to fit nicely into the small mediocre performance television sets found in places like the corner grocery store or the home kitchen. Prior to the competition of new digital media formats, consumers had accepted this audio as "just the way it was." Those days are over.

Broadcast standards have also kept pace and the capability for delivering high definition pictures and 5.1 channel audio is standardized around the world. The many paths to the consumer all support delivering an exceptionally high quality version of the original.

Modern television sets have better pictures and better built-in sound systems to handle the higher performance of new media. Content production has also improved as theatre-quality mixes can now be delivered to consumers. Sports broadcasting has been a major catalyst for driving the unprecedented sales of flat panel high definition displays and has accelerated the already healthy DVD-driven sales of 5.1 channel home theatre audio systems.

We Won't Get Fooled Again

While picture quality continues to rapidly improve, the audio portion of most every digital television standard is a solid performer. International listening tests have shown that the quality of the original audio and what reaches the consumer can be so nearly identical that any differences are considered statistically insignificant. This is an amazing step forward from previous formats like BTSC and NICAM stereo, however it only describes the very last step: transmission.

More challenging are the differences in production technique from the myriad sources of 5.1 channel programming. There is a disparate mix of film mixes meant for 400+ seat theatres, talk shows with a dynamic range intended for a 3-inch television speaker, and the age-old loud commercial that is now annoying in surround.

With the advanced quality of the transmission systems comes responsibility. Previously, a transmission processor was installed for compliance reasons, to give a broadcast station a "signature sound" and to tame loudness variations. Unfortunately, the worst content generally dictated how aggressive the processing had to be set.

Today's transmission systems have background controls called metadata that allow the audio to be better tailored to the needs of different viewers. Effective metadata takes time and experience to create and manage, and can vary in quality and accuracy as much as the audio itself.

The answer is clearly not a traditional processor or simple wide-band controller. Traditional audio processing and tools meant for restricted legacy delivery systems will produce restricted results. Viewers have experienced better and expect HD audio to match their HD pictures.

Set The Controls

After budgets, probably the next scarcest commodity in modern broadcast is time, followed rapidly by experienced staff. Broadcasters must do more with less than ever before. Getting the extra burden of the audio and metadata systems in digital broadcast and production under control is the focus of Linear Acoustic. The products on the following pages are designed to control loudness, support metadata and provide streamlined methods for producing and delivering high impact 5.1 channel audio appropriate for all viewers on time and within budget.

AERO.air™

Transmission Loudness Manager

Inconsistent DTV audio loudness, or the so-called "loud commercial problem" is the number one complaint of television viewers and it is driving them away. Is your station delivering the best sounding audio possible? If the answer is no, do you have time to manage the problem? Two of the largest audio issues in digital television are controlling loudness and keeping the audio image stable.

The AERO.air offers all the loudness control, audio processing, decoding and encoding a broadcaster needs for television audio. It combines the latest air-proven loudness control with 'Hollywood approved' upmixing capabilities to allow seamless integration of local two-channel audio and network 5.1 audio.

The unit accepts 5.1-channel and two-channel audio via AES or optional HD/SD-SDI inputs, plus a dedicated analog or digital EAS/Aux input. Audio is then processed by the multiband, multistage ITU-compliant loudness core resulting in smooth audio with appropriate dynamics. Two-channel audio can be automatically upmixed to produce a consistent surround-field that is perfectly downmix compatible for all stereo viewers.

Optional internal Dolby® Digital (AC-3) and Dolby Digital Plus (Enhanced AC-3) encoding is available with HE-AAC (Dolby Pulse) supported via firmware upgrade. Optional Dolby E/Dolby Digital (AC-3) decoding can also be included.

If present, audio metadata can be used to control upmixing and improve loudness control while minimizing impact on source audio. Extensive fallback options enable the unit to compensate for missing or incorrect metadata.

Included AutoMAX-II™ fixes the annoying problem of two-channel audio being sent to consumers but signaled as 5.1-channel audio. AutoMAX-II handles this situation effortlessly and without risking 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 for simple local monitoring. This output is also available on the analog and headphone connectors.

Extensive standard I/O includes ten main AES inputs and outputs plus stereo balanced analog in and out, and a front panel headphone connector. Optionally, HD/SD-SDI I/O allows for de-embedding and re-embedding up to 16 channels of audio plus SMPTE 2020 (VANC) metadata.

Three versions of the AERO.air processor are available, and units can be field software upgraded to add features as needed. The AERO.air (TV) provides dual stereo processing, the AERO.air (DTV) provides 5.1 channel loudness control, upmixing, and downmixing, and the fully-featured AERO.air (5.1) adds an additional stereo processor for local input, to support programs up to 5.1+2+2.

A color TFT display, large rotary encoder, and four control keys provide for straightforward menu navigation and adjustment. Simple remote control is provided by parallel TTL inputs and status outputs. A 10/100 BASE-T Ethernet connection is available. In addition, the AERO.air contains dual redundant power supplies that are hot-swappable, and hard relay bypass of the digital audio, SDI and metadata signals for mission critical applications.

AERO.one™

DTV Audio/Loudness Manager

It is clear that with so many television services transitioning to digital, a simple and cost effective solution is needed to manage loudness, upmixing, metadata, signal routing, and audio coding. The third generation Linear Acoustic AERO.one DTV audio and loudness manager is the answer.

The one rack unit AERO.one protects viewers from loudness shifts and loss of surround sound in a simple, cost effective, compact, and feature rich manner. Offering Built-in loudness control, metadata control, and optional transmission encoding make the AERO.one the ideal choice for affiliate stations that need to match local and network content and provide a high-quality and seamless surround sound experience for their viewers.

The AERO.one is also well suited as a processor for the backup transmission path, allowing the audio quality of the main path to be matched in a cost effective manner. The AERO.one is also available in a dual stereo (2+2) version for stereo and SAP processing. An additional dual stereo processor can be added to support 2+2+2+2 processing making it ideal for processing stereo feeds intended for delivery to cable and satellite providers for their SD or analog services.

Similar to other Linear Acoustic processors, the AERO.one accepts three pairs of PCM audio to handle two channel to 5.1 channel network or local audio programming. The unit can apply multiband, multistage loudness control and upmixing to the local audio allowing it to be matched to 5.1 channel network audio. Loudness matching is maintained by managing the loudness of local content allowing it to be better matched with network or other content.

Upmixing is provided by the air-proven and industry standard UPMAX algorithm which provides compelling 5.1 channel audio while remaining completely downmix compatible. The AERO.one also includes the new AutoMAX-II auto-detection algorithm to smoothly and automatically bypass upmixing when applied content is 5.1 channels. The AutoMAX-II algorithm prevents any loss of dialogue or cause any switching artifacts. Upmixing and EAS (processing bypass) modes can also be controlled by a combination of GPI contact closures and applied metadata.

A full-time downmixed version of the main program is provided as the fourth AES output pair. This signal can be either a stereo LoRo downmix or an industry standard LtRt surround encoded mix compatible with all legacy consumer decoders.

Metadata is accepted as a standard RS-485 serial input for control of upmixing and processing functions. Extensive metadata processing is included to protect viewers from the audible effects of incorrect or missing settings. Metadata can also be extracted from the vertical ancillary (VANC) space of an applied HD-SDI signal.

Available options include built-in 5.1 channel Dolby Digital (AC-3) encoding via a Dolby manufactured encoder, HD/SD-SDI I/O, and dual, auto-ranging medical grade power supplies.

A bright LED display, rotary encoder, and four control keys provide for easy menu navigation and function adjustment on the unit. Hard relay bypass of audio, metadata, and SDI signals for trouble-free operation in transmission critical environments.

AERO.qc™
Audio Quality Controller

The new Linear Acoustic AERO.qc contains everything necessary to get audio under control at the beginning of the program chain. During ingest or quality control, AERO.qc provides ITU loudness measurement, manual or automatic correction of program loudness to match a target value, selective upmixing to create 5.1-channel audio, and outputs audio and metadata that match plant or externally supplied delivery specifications.

The AERO.qc can also be used during program production. Channel level meters are displayed next to a numeric indication of program loudness, allowing levels to be managed in real time. Upmixing can be used for integrating two-channel audio into a 5.1-channel program, and dynamics control can be applied to the audio prior to output.

Experience has shown that managing loudness at each section of the program chain results in better loudness and sound-field control with fewer side-effects. Using AERO.qc as early in the production process as possible delivers audio that requires little additional processing while still satisfying viewers and regulators.

Simply relying on one final stage to blindly correct loudness issues means that all programming will be affected to the degree necessary to correct the worst pieces of content. While this approach does control loudness, it does so at the expense of modifying properly produced content to compensate for content that is incorrect.

Using the AERO.qc at the point of ingest or production allows loudness and other important aspects of the audio program to be measured and set correctly at the earliest stage possible - even by inexperienced operators. Final loudness control processing can then be backed off to provide protection without constant modification.

Calibration of the monitoring environment is a basic but critical part of managing loudness and the AERO.qc includes extensive tools to help get this right. Calibration noise and channel ID tones including BLITS are coupled with at least eight parametric equalizers per speaker output and a real-time audio spectrum analyzer to allow for precise alignment. Settings can be saved to presets allowing different setups to be instantly recalled for different rooms or remote OB trucks.

Available features and options include:

- ITU-R BS.1770 plus true peak metering
- Running line graph of loudness history
- UPMAX and UPMAX-II upmixing
- 8-channel analog output
- Extensive monitor calibration features
- Real time 1/3rd and 1/6th octave analyzer
- Optional 5.1-channel loudness processing
- Optional Dolby E/AC-3 decoding
- Optional HD/SD-SDI I/O
- Optional dual power supply

Options include 5.1-channel multiband, multistage AERO-style loudness control, Dolby E/Dolby Digital (AC-3) decoding, HD/SD-SDI de-embedding and re-embedding for audio and metadata, and 8-channel balanced analog monitor outputs with inputs for remote fader, mute, and return to reference controls and indicators.

A detailed color OLED display provides channel audio channel levels, ITU loudness with true peak meters, plus key metadata information simultaneously on one screen. Front panel control is via two navigation clusters that allow quick menu navigation and control of monitor and headphone levels.

Loudness Control Product Comparison

FEATURE	AERO.air (5.1)	AERO.air (DTV)	AERO.air (TV)	AERO.one (DTV)	AERO.one (TV)	AERO.qc
Channels	5.1+2+2	5.1	2+2	5.1+2	2+2 (4x2 opt.)	5.1+Downmix
Chassis	3RU	3RU	3RU	1RU	1RU	2RU
Input 3-Band Paramteric Eq	Y	Y	Y	Y	Y	Y
ITU Compliant Loudness Control	Y	Y	Y	Y	Y	Y
Input AGC	Y	Y	Y	Y	Y	Y
Multiband AGC (mono)	Y	Y	Y	Y	Y	Y
Multiband AGC (Stereo)	Y (x4)	Y (via 5.1)	Y	Y	Y	Y
Multiband AGC/Limit (5.1)	Y	Y	N	Y	N	Y
Number of Bands	4, 5, 6	4, 5, 6	4, 5, 6	5	5	4,5,6
Look-Ahead Peak Limiting	Y	Y	Y	Y	Y	Y
Upmixing	Y (dual)	Y	N	Y	N	Y
Local Input	Y	N	N	N	n/a	n/a
Downmix of 5.1 Input	Y	Y	n/a	Y	Y	Y
Metadata Input	Y	Y	Y	Y	Y	Y
CrowdControl	Y	Y	Y	N	Y (optional)	Y
Remote Control	TCP/IP	TCP/IP	TCP/IP	TCP/IP	TCP/IP	TCP/IP
GPI	8	8	8	4	4	8
GPO	2	2	2	4	4	2
User Presets	>64	>64	>64	16	16	>64
Software Update	Front USB	Front USB	Front USB	Ethernet	Ethernet	Ethernet/Rear USB
Display	QVGA	QVGA	QVGA	Graphical LED	Graphical LED	Color OLED
User Interface	4-button, Rotary Enc	4-button, Rotary Enc	4-button, Rotary Enc	4-button, Rotary Enc	4-button, Rotary Enc	Dual Rotary Encoder
Latency (PCM)	25msec	25msec	25msec	15msec	15msec	25msec
Utility Delay	0-100msec	0-100msec	0-100msec			0-100msec
Dolby Digital Encoding Option	Y	Y	Y	Y	Y	N
Dolby Digital (AC-3)/E Decode	Y	Y	Y			Y
HD/SD-SDI Input and Output	Y	Y	Y	Y	Y	Y
Reference Input	AES/SDI	AES/SDI	AES/SDI	AES/SDI	AES/SDI	AES/SDI
Balanced Analog Outputs	Stereo (LtRt/LoRo)	Stereo (LtRt/LoRo)	Stereo (LtRt/LoRo)	N	N	8-channels w/remote
Dual PSU	Y	Y	Y	Y	Y	Y
Headphone Output	Y	Y	Y	N	N	Y
Warranty	2 years	2 years	2 years	2 years	2 years	2 years

UPMAX:neo™

5.1 Channel Production Upmixer

Creating consistent, high quality continuous 5.1-channel audio is challenging. The Linear Acoustic UPMAX:neo 5.1-Channel Surround-field Synthesizer has been designed to ease the production of 5.1-channel audio programs by creating an infinitely adjustable multichannel signal that is completely downmix compatible.

The Linear Acoustic UPMAX:neo also provides the most effective and compatible solution for integrating legacy two-channel material into today's 5.1-channel programs. Listeners 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 metadata is not available to switch the Dolby® Digital (AC-3) encoder.

The Linear Acoustic UPMAX:neo uses proprietary processing to create downmix compatible Left, Right, Center, Surround, and LFE channels from the two-channel main input. This "surround-field" can then be infinitely adjusted via the Center Channel Width control and the Surround Channel Depth control. This allows programming ranging from simple stereo to LtRt to be appropriately reproduced from a 5.1-channel playback system.

The unit accepts three AES pairs of audio and will upmix the first pair, and allow all three pairs to pass via a crossfade when upmixing is disabled allowing the unit to remain in a 5.1 channel path. Upmixing and bypass can be controlled by GPI contact closure and applied metadata.

In addition to the industry standard UPMAX algorithm, the UPMAX:neo also includes a Linear Acoustic-tuned version of the DTS® neo:6 algorithm. In post-production, this provides the user with additional tools that might be better suited to certain content. The neo:6 algorithm produces a very stable 5.1-channel version of two-channel inputs. The 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.

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

The unit can be controlled via contact closures applied to the GPIO port or via applied audio metadata. The metadata generation option adds Dolby-standard metadata creation to the unit, allowing 5.1 audio and accompanying metadata to be easily created.

Standard audio I/O is via four AES inputs and outputs. Optional HD/SD-SDI I/O is available allowing access to all 16 embedded audio channels. Audio can be processed or passed without modification from any embedded input pair to any embedded output pair.

The Analog Output option provides eight channels of balanced audio capable of output up to +24dBu. This option includes volume and mute functions that can be adjusted from the front panel or via the remote control port which adds a return-to-reference function and outputs to drive LEDs.

A bright LED display, rotary encoder, and four control keys provide for straightforward menu navigation and function adjustment on the unit. The unit features an auto-ranging medical-grade power supply for worldwide operation and hard relay bypass of audio and metadata signals for trouble-free operation in transmission critical environments.

L.A.M.B.D.A.-II™

Loudness, Audio, and Metadata Monitor

With Optional Dolby E/AC-3 decoding

Audio Monitors Are Not Created Equally.

Broadcast audio has been taken a quantum leap forward in quality and complexity, and it is no longer sufficient to simply display metering information and play audio out of speakers.

Modern digital broadcast audio has grown beyond stereo and also supports 5.1 or more channels of surround sound. Audio is now accompanied by metadata, or data that describes the audio data which is used to configure downstream coding equipment and is passed along to consumers to automatically optimize their playback systems. This can work well assuming the audio and metadata are correct.

This means that in order to properly monitor audio inside a plant, an audio monitor must carefully take into account metadata and appropriately adjust metering and playback audio levels. If this is not done, operators will be unable to determine if the audio they are viewing and hearing is truly correct and could result in bad sounding decisions.

Enter the Experts

Introducing the Linear Acoustic Monitor for Broadcast Digital Audio, LAMBDA. This unit combines a unique understanding of audio and metadata through the entire broadcast chain from production to consumer.

LAMBDA can display and reproduce 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 can be 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 analog 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.

A long-life and durable vacuum fluorescent display, plus two intuitive navigation clusters for menu and monitoring functions provide straightforward system navigation and audio adjustment. Quick navigation is accomplished by four dedicated function keys that allow direct channel selection, instant preset recall, and downmix configuration.

Optional Dolby Digital (AC-3) and Dolby E decoding or Linear Acoustic e-squared decoding can be applied to any of the discrete AES or embedded audio pairs. Decoded audio on AES outputs, decoded metadata is displayed and can be applied to the audio and metering signals. An optional second power supply is available for redundancy.

StreamStacker-HD™

High Density Audio Distribution

Carrying multiple programs with surround sound, audio metadata, and associated auxiliary data finally has a simple answer- the Linear Acoustic *StreamStacker*-HD™ system. Using advanced multiplexing and coding technologies, up to sixteen channels of PCM and emission rate audio can be combined with audio metadata and carried together in a single, easy to manage AES pair.

Unlike other systems, the StreamStacker-HD bitstream called e^2 (e-squared) can be switched on AES frame boundaries allowing video frame or field edits, is exceptionally resilient to errors, and keeps audio and video tightly synchronized via built-in compensating video delays included as part of the HD-SDI option.

Up to 16 channels of PCM audio (48-96kHz, 16-24 bits) can be input to the encoder as eight AES pairs, optionally through HD-SDI, to create a 20-bit, 48kHz e^2 output bitstream. Soft bypass inputs are provided on the decoder to allow formats such as Dolby E to be externally decoded and passed through to ensure that legacy and new e^2 content can coexist.

Metadata functionality includes internal generation and regeneration of audio metadata. Audio metadata can be provided via the legacy 9-pin RS-485 format or from within the VANC space of the HD-SDI input.

The flexible e^2 format can optionally support the inclusion of asynchronous data such as emission rate audio or other codecs; contact factory for details.

The Model LA-5174 Decoder follows the e^2 input stream and de-multiplexes signals to their appropriate outputs. PCM channels are clocked to an applied local AES or HD-SDI reference, and output as AES-3 signals and optionally re-inserted into an applied HD-SDI signal.

Metadata output is available on a standard 9-pin connector for interfacing to products supporting audio metadata via serial input, and can also be embedded into the VANC space of an HD-SDI signal applied to the unit.

Internal metadata generation is provided for cases where incoming metadata might be absent or incorrect. Metadata frame synchronization is included in both the LA-5173 Encoder and LA-5174 Decoder to guarantee reliable metadata regardless of channel error conditions.

A software decoder is available for licensing into third party equipment and applications running in real time or non-real time. Contact the factory for details and an SDK for development.

A bright LED display, rotary encoder, and four control keys provide for straightforward menu navigation and function adjustment on both the LA-5173 e^2 Encoder and the LA-5174 e^2 Decoder. Dual, redundant, auto-ranging power supplies allow for trouble-free operation worldwide.

MetaMAX™

Metadata Processor, Generator & Analyzer

The Linear Acoustic LA-5180 Audio Metadata Frame Synchronizer and Generator is an integral part of any facility relying on the successful creation and continuity of metadata for proper audio encoding. Useful in digital television transmission, remote production, and DVD creation environments, the flexible unit accepts professional audio metadata via a standard RS-485 serial input, or as delivered inside the VANC space of an HD-SDI stream. The unit then corrects or replaces bad or missing metadata packets, outputting stable metadata via RS-485 and also selectably back into the VANC space of the applied HD-SDI signal.

The LA-5180 can also be used as a stand-alone audio metadata generator, with eight presets available to store all required parameters for both the professional metadata (a.k.a. Dolby E metadata) and the consumer metadata (a.k.a. AC-3 metadata).

The LA-5180 can be located in transmission or production areas, remote OB trucks, or anywhere metadata needs to be processed or generated. Standard NTSC or PAL black burst video reference is required.

AMAS Analysis Software

Advanced AMAS™ metadata analysis package is optionally provided via TCP/IP connectivity to a remote PC. AMAS provides detailed display and analysis of critical metadata parameters and user-definable limits for values drive error indication and logging. Interactions between related parameters can also be trapped and alarmed, helping to prevent unforeseen interactions that could cause loss or damage to decoded audio. Metadata and logging information can also be recorded for compact storage and future analysis.

LA-5448

AC-3 Framesync & Rateshaper

Maintaining consistent, error-free AC-3 bitstreams is critical for high quality audio transmission. Bitstreams delivered from sources outside of the local plant must be synchronized to local reference, must be free of errors and must be of a consistent data rate to prevent multiplexer issues.

The Linear Acoustic Model LA-5448 Frame Synchronizer and Rate Shaper satisfies these requirements and more, all without re-coding and in a high-density 4-stream 1RU package.

Silent Frame mode keeps the output of the synchronizer error-free. Based on CRC values present in every AC-3 stream, error detection and concealment is accomplished by repeating the last good frame. If the error condition persists longer than one frame, silent AC-3 frames are inserted, maintaining the data rate and channel mode of the incoming AC-3 stream.

Rate Shaping mode keeps the output of the synchronizer fixed at a single, consistent data rate, regardless of the input data rate. This is accomplished by transforming the input AC-3 frames of any rate equal to or lower than 448 kbps to a fixed 448 kbps. While Rate Shaping is enabled, Silent Frame mode is also automatically enabled, thereby producing a consistent output data rate that is devoid of errors.

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