LINEAR ACOUSTIC®
THE TELOS ALLIANCE®

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TV SOLUTIONS GROUP

Leading the Television Industry's Migration to **AoIP**

AMS™ Authoring and Monitoring System – Delivering on ATSC 3.0

Linear Acoustic AMS™ – Delivering on ATSC 3.0

Linear Acoustic AMS Authoring and Monitoring System is a comprehensive solution for real-time authoring, rendering, and monitoring of advanced audio programs for the ATSC 3.0 Digital Television System. The audio system in ATSC 3.0 provides listeners with a personalized and immersive audio experience using Next Generation Audio (NGA) technologies, including MPEG-H. AMS simultaneously delivers advanced audio for ATSC 3.0 broadcasts and 5.1-/2-ch audio for legacy ATSC 1.0 broadcasts.

Fully supporting advanced features of MPEG-H, AMS is designed for live metadata authoring and rendering audio for use in real-time broadcasts, along with monitoring content in a variety of listening configurations. Using smart metadata and Linear Acoustic APTO (the latest in standards-compliant loudness control), AMS facilitates easy audio mixing and authoring operations, even in the most demanding production scenarios. Individual audio elements are combined with user specified metadata to create immersive and personalized audio programs for the viewer. The AMS web interface allows the authoring engineer to easily control the interactive features of MPEG-H offered to the viewer, and build presets for different listening experiences. This ultimately enables viewers to quickly choose and personalize their sound experience for optimal playback in their environments, from mobile devices up through immersive home theatres.

Simple User Interface

The system provides a web interface for the following functions:

- Device configuration and status
- Input/Output routing
- Loudness processing
- Loudness Meter
- Level Meter
- Monitoring control
- Audio mixing and object panning
- Authoring configuration and control
- User's interactivity
- Logging

Loudness data may be logged to a file, with user-selectable parameters for log intervals and contents. A variety of loudness parameters of each preset program is displayed on the web interface to allow easy and compliant mixing operations.

Authoring Made Easy

Authoring an MPEG-H audio stream using the web interface is as simple as assigning the 15 available mono inputs to the 10 available channel groups. From defining music beds, identifying independent objects, or grouping objects to be switched by the user, full control is available.

Controls include:

- Object position
- DRC
- Loudness parameters
- Interactivity controls
- Upmixing from mono/stereo to 5.1 or 7.1
- Monitor control

The engineer can define presets that apply all attributes of a configuration for ease of use and full recall.

Comprehensive Real-Time Monitoring

Linear Acoustic AMS can be used in production to actively author and monitor metadata, or in other parts of the broadcast chain to monitor and validate audio streams containing a control track which were authored upstream. In both scenarios, AMS is capable of outputting 15-channels of discrete audio with a metadata control track, a 5.1-channel rendered output, 2-channel rendered output, and a dedicated monitoring output, all simultaneously.

Full monitoring controls are available to the operator to check individual source elements, or to emulate the consumer listening experience for any of the programs and presets while not affecting the other 3 outputs. The operator can choose to listen to speaker configurations from stereo up through 7.1.4 at the monitoring output, and at the press of a button, change the output configuration in order to emulate what the viewer at home will experience.

Loudness Adaptation by Linear Acoustic APTO™

AMS is equipped with Linear Acoustic APTO, the state-of-the-art loudness adaptation technology designed to carefully control audio levels in a way that preserves the transients, sonic image and artistic intent of the source, while ensuring loudness consistency and compliance for any desired target.

The xNodes xFactor

In any broadcast environment, engineers must be able to deal with a multitude of I/O configurations. The Telos Alliance xNode family of AoIP devices allow Linear Acoustic AMS to be configured with optional analog, AES/EBU, GPIO logic, and SDI I/O, all with full Livewire+ AES67 Audio over IP support.

Telos Alliance SDI xNodes can de-embed 3G/HD/SD-SDI inputs, extracting up to 16 channels of audio to the Livewire+ AES67 port. The audio can be re-embedded into the SDI output stream with full video delay compensation for each SDI input, ensuring that audio video synchronization is maintained.

Using the web interface, up to 36 outputs may be configured, including 16 channels of authored audio plus control channels, up to 12 channels for monitoring, and two additional sets of output channels for rendered broadcast outputs. Outputs may be routed to an SDI xNode for embedding in the SDI signal, and to any AES67 compatible device.

Linear Acoustic AMS™ system configuration:

One main processing unit, which implements functions for authoring, rendering and monitoring, and web hosting of the user interface

- MPEG-H authoring, rendering and monitoring functions
- 16 channel audio input via Livewire+ AES67 over RJ-45 Ethernet
- 36 channel audio output via Livewire+ AES67 including authored audio plus control channel, monitor output, and up to two legacy rendered outputs
- 1 RU high by 1 RU width
- 95-240 VAC, 50/60 Hz dual power supplies

Optional Controllers

Linear Acoustic AMS supports:

- Industry standard Windows and Mac OS (Microsoft Edge, Safari, Firefox, Google Chrome browsers)
- Touch screen devices
- Mobile devices
- Hardware control panels, including Axia and 3rd party controllers
- 3rd party Joystick controllers for object positioning

2 x Telos Alliance 3G SDI xNode interfaces, each supporting 2 SDI signals

Each SDI xNode provides:

- Two relay-bypassed 3G/HD/SD-SDI inputs with access to all audio channels
- Dual, independent compensating video delays
- Up to 16 channels of audio via Livewire+ AES67
- 1 RU high by ½ RU width
- 95-240 VAC, 50/60 Hz, 15W maximum Redundant power sourcing available via external +12V DC input
- Two RJ-45 connections available for management or AoIP interface

4K video is routed in and out of the system using quad 3G SDI signals

Multiple choices of alternative or additional I/O are available

Specifications are subject to change without notice.

Production Library for the MPEG-H TV Audio System licensed by FRAUNHOFER
https://www.iis.fraunhofer.de/de/ff/amm.html

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