2pAAa8. A new third generation time variant electro-acoustic enhancement system

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This paper describes the hardware and software implementation in a new generation of time variant electronic acoustic enhancement systems designed for use in medium and large sized venues. Examples of currently installed systems and applications will be discussed, as well as capabilities for sound file storage and playback, multi-channel film surround sound, 2D effects panning, and 3D live tracking.

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Introduction

In 1988, Dr. David Griesinger and the author worked with Neil Muncy to document the state of the art in electronic acoustic treatments available at that time; determine the fundamental shortcomings of these systems, and develop a new solution that addressed the fundamental problem that all systems of this type must contend with – acoustic feedback. The system we developed became known as LARES (Lexicon Acoustic Reinforcement and Enhancement System), and the first installation in 1989 (engineered installed with Mr. Muncy’s expertise) at the Elgin Theatre in Toronto is still in daily use. Since that time, hundreds of systems have been installed throughout the world in a wide variety of venues and applications. This paper describes the latest developments in both system hardware and software for the third generation systems.

Background

Affordable digital signal processing was relatively new at the time that we undertook the development of the first systems. Digital processing important to the successful integration of moderately large multi-channel systems was only available in component form. In other words, one could incorporate multiple digital delay units, multiple digital equalizers, etc. – but the means to perform large scale signal combining in the digital domain did not exist commercially. The advent of multi-channel digital signal processing platforms that incorporated the functionality of the individual components made electronic architecture systems practical for a wider range of venues. The practicality of “second generation” systems, however, is somewhat short lived. These digital processors were constructed using readily available ASIC’s developed for the audio industry, and their fabrication typically utilized parts germane to the desktop computer industry. As the nature of personal computing shifted from the desktop to mobile devices, these parts often became scarce, and the products reached end of life – only to be replaced by new platforms that were not backwards compatible.

Development

Product development for our latest systems began with an investigation of hardware longevity. Our experience with the first generation Mainframe platform was a good model. The bus architecture and circuit boards were engineered in house – which allowed software engineers considerable latitude in developing system upgrades. Even though the circuit boards were modular, a significant hardware upgrade represented new development. Most of the time, the cost analysis for such development favored a new approach – primarily because the older components that remained in place represented a potential “Achilles heel” in additional future re-engineering. With the pace at which hardware had potential to be superseded, a new ground up hardware design using audio DSP was not favorable. In years following the release of our first generation systems a new application that required significant processing power, reliability, and component longevity emerged. The modern internet created the demand for server class processors. These processors are designed to provide years of continual service. The system architecture provides on-board diagnostics that can be monitored at board level. The emergence of multi-core processing made these systems an ideal candidate for audio system integration. Most important is that the production quantities required to meet the demands of the server industry are several tens if not hundreds of orders of magnitude greater than the requirements for the entire audio industry.

Acoustics Software

The previous generation algorithms were based on years of digital reverberation development for the music industry, and were tied to an ASIC developed specifically for this purpose. Our ongoing research into human perception pointed to the need for a fundamental change in structure of the algorithm. We had also developed a version of the original algorithm with reduced functionality for use in small practice rooms that operated in hardware developed for consumer electronics. In developing new software, one of our goals was to create an algorithm with optimum sonic performance that could operate in the footprint of a processor designed for appliance applications. This would enable us to use the same core development in both hardware for typical systems, as well as hardware specifically designed for smaller, stand-alone systems. This goal has been achieved, and the new Mainframe III acoustics processor provides four independent virtual acoustics “machines” each equivalent to the original Mainframe acoustics processor. Each machine is capable of independently processing any ratio of direct, reflected and reverberant energy; and each provides adjustment of all critical acoustical parameters.

Signal Processing

The past generation of signal processing used hardware that downloaded and ran a compiled instruction set generated by a program operating on an external PC. Thus, the “devices” used in the signal path had to be manually
selected and interconnected in software to form signal flow through the processor. This allows virtually unlimited flexibility for implementing designs for specific sound reinforcement applications, but is time consuming for creating large multi-channel systems.

Conversely, our new Matrix Processor utilizes a large point-to-point matrix that is scaled for specific systems. This saves considerable time, as the basic structure for signal flow is already in place, and remains consistent project per project. Parameter values are communicated using human readable ASCII, and for the most part are executed as SET and GET commands (example SET CHANNEL INPUT 1 MUTE ON or GET CHANNEL 1 MUTE which returns the state). The Matrix Processor hardware comes in two forms. E-Architecture is built for the largest conceivable systems with over 8000 internal channel capacity and as many as 1024 output channels. This system can be scaled to as few as 64 Inputs and outputs. E – Performance uses the same matrix structure, but is limited to 128 inputs and outputs. This is more than sufficient to service small and medium sized auditoria, and houses of worship. Figure 1 depicts basic signal routing in E-Architecture and E-Performance systems.

Two Matrices

The software matrix is actually two independent matrices that terminate to system outputs. The Live matrix processes signals from microphones, the Mainframe III acoustics processors, sound effects, film surround channels direct reinforcement signals, and any other conceivable input; routing each to appropriate outputs. The second matrix – the playback matrix – enables sound files stored on the system to be cued and played to any of the system output channels with the same parameter adjustments available to any input signal.
Audio Transport

There are a number of Ethernet based multi-channel audio transport formats available such as Cobranet, Ethersound and Dante. Each of these systems has unique features – but our experience with IT management on large projects like arenas led us to choose MADI for audio transport. Since MADI does not use Ethernet interconnection, termination, or ancillary hardware, it does not fall under the jurisdiction of the IT department. MADI also conforms to the AES/EBU audio standard, and supports 64 full bandwidth 24 bit audio channels using either coaxial cable or optical terminations. It also supports transmission paths up to 1000’ for coax terminations and up to 3000’ for optical terminations. Thus, MADI is the optimum transport method for systems of this type.

The Third Matrix

Another advantage of our MADI implementation is the on-board matrix assignments available on each MADI card. This allows input and output channels to be re-assigned or duplicated as required. The MADI cards also have on-board signal processing that provides real time monitoring of all channels without loading the CPU in any way. Each card also has a dedicated headphone output that can be assigned to monitor any or all of its inputs and outputs. Figure 1 diagrams the signal flow through the matrix processor. The list to the right of the signal flow is a partial list of parameters that are available at each input, cross point, and output in the matrix.

![Figure 1: Diagram of signal flow through the matrix processor.](image)

FIGURE 2. This is a basic diagram of signal flow from MADI inputs and playback signals through the Matrix Processor.

2D Sound

The Matrix processor can communicate with industry standard show control systems using MIDI, MIDI Time Code, and native cue commands. Every parameter in the system has a FADE command. This enables multiple live channels, as well as playback files, to be panned across the matrix in real time. Productions that use sound effects playback systems can assign these as static channels with pre-mixed level delay and EQ that are fed to specific output channels and zones.

![Figure 2: Diagram of 2D Sound signal flow.](image)
3D sound

The matrix processor can also interface with location servers to dynamically pan audio based on the positioning of sources in the environment. Hence, in addition to acoustic enhancement and sound effects panning the system can also accommodate multi-channel direct sound processing such as delta-stereophony, high order ambisonics, and wave field synthesis.

Control

An industry standard control system provides user control through either a dedicated touch screen interface, or by using a mobile application that communicates with the control system. All of the processors are interconnected using a private network and use industry standard Ethernet termination and TCP/IP protocols. This also provides the means by which to log into the system remotely to perform diagnostics.

Sound Server Capability

The Matrix processor has the capacity to service multiple independent venues that are part of a performing arts complex, or college campus. Acoustics machines can be assigned to specific output channels that service various independent venues including rehearsal facilities as well as the primary performance spaces. Processing can even be allocated by time date and function to various venues within a facility.

Example 1 – Stonebriar Community Church

Stonebriar Community Church features fan shaped sanctuary with seating capacity close to 3000 (see Figure 3,4). Unlike many sanctuaries with geometry, the programming at Stonebriar includes a large pipe organ, antiphonal organs, a 140 member choir, and a symphony orchestra. The reverberation time and level provided by the architectural acoustics are both sufficiently low as so to maintain good intelligibility of the spoken word. This enables the electronic architecture to increase both the reverberation time and level in keeping with a volume of this size. The system incorporates six independent acoustics machines in two Mainframe III acoustics processors. The first is assigned to the stage area and provides support and cross-stage communication for the choir. The second machine is fed with six hyper-cardioid microphones in close proximity to the choir, and its output feed the large format loudspeakers indicate in blue that form the electronic reflector. This is used to increase impact and maintain intelligibility, while also maintaining localization to the ensemble. The third machine uses signals from cardioid microphones over the orchestra, and distributes early energy to the loudspeakers throughout the sanctuary. The fourth machine uses the same microphones and provides reflected and reverberant energy to the same loudspeakers. The fifth and sixth machines use cardioid microphones located over the right and left sides of the congregation to uniformly join them in the reverberant field. These machines also pick up the antiphonal organs, and provide optimum timing of these signals toward the front of the sanctuary. All of the microphone signals and outputs from the Mainframe III processors terminate to an E-Architecture Matrix Processor that provides 192 independent inputs and outputs. Thirty six independent subwoofers are also incorporated to provide low frequency support for the pipe organ.

FIGURE 3. Photo of Stonebriar Community Church Sanctuary
FIGURE 4. Plan view of loudspeaker locations for Stonebriar Community Church

Example 2 – Pius High School

Pius High School Auditorium is a renovation of an existing gymnasium space that has been re-designed to provide a multi-purpose lecture and performance venue (Figure 5.) The auditorium geometry is a narrow fan shape with seating for approximately 500 (Figure 6). A substantial amount of absorptive material was applied, with spacing that provided even distribution throughout the venue. The rear wall has additional treatment that eliminates reflections from the side walls, and breaks up what would otherwise be echoic return to the stage. The system uses one Mainframe III processor that provides four independent acoustic engines. The first machine is assigned to the stage and provides independent acoustic control for different types of performances. The second machine provides early energy throughout the auditorium. The third machine provides reflected and reverberant energy to the ceiling; and the fourth machine provides this energy to the lateral and rear arrays. An E-Performance Matrix Processor is used with 48 independent output channels.
FIGURE 5. Photo of Pius High School Auditorium

FIGURE 6. Loudspeaker locations for Pius High School

Conclusions

These new systems represent a leap forward in functionality, longevity, and overall cost. In addition to the core role of acoustic enhancement, the new hardware and software provides the means to integrate a wide range of additional functionality as part of a multi-channel electro-acoustic installation. A range of hardware platforms have
been developed that enables cost effective solutions for small, medium and large sized systems and applications, with no compromise in system performance. Most important is the development of an entirely new acoustics algorithm based on our most recent neurological research that provides a substantial improvement perceived sonic quality.