MATRIX3™ AUDIO CONTROL SYSTEM: STRUCTURE AND CAPABILITIES

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1. INTRODUCTION

As our culture becomes immersed in multimedia on a day-to-day basis, there is a greater demand for sophisticated and reliable control for the delivery of the audio portion. Live theater, sound reinforcement, location-based entertainment, themed retail, education, houses of worship, stadiums, and conference centers all rely on audio to maximize the impact of their messages [1]. Home entertainment systems with surround sound are decreasing in price and at the same time are increasing in performance and sophistication. Catalog and web sales are starting to take off. In order for entertainment venues and retail establishments to attract customers, they have to exceed what is available at home, both in service and in the total multimedia experience — picture and sound. This has driven sound designers to "up the ante" with increasingly sophisticated tools. At the same time, these establishments are relying on lower-cost, in-house technical support to operate and maintain these systems. Integrating functionality that has traditionally been allocated to different pieces of equipment into one product—with one software interface—reduces cabling, centralizes control, and decreases the cost of spares and maintenance. The challenge is clear: Build highly integrated products, with creative capabilities, that are reliable and easy to maintain.

Level Control Systems' Matrix3 is a new modular computer-controlled multichannel audio system. Matrix3's hardware and software structure, programming approach, and creative capabilities are discussed. The audio system design for a multipurpose venue is proposed.

1.1 Background

Theatrical traditions have their origins in prehistoric religious ritual and ceremonial practice, and practitioners through the ages have used every available device (music and sound effects, costuming and props, scenography, lighting, mechanical effects) to transport the audience (or congregation!) to a magical world apart from everyday reality. From rituals in the caverns at Lascaux, to Balinese shadow play, to Wagner's Gesamtkustwerk, to contemporary Virtual Reality immersion environments the impulse and methodology are much the same even though the technologies have evolved [2].

Theater designers are ever ready and eager to adopt "whatever works" from any source, and theater sound designers are now using tools appropriated from the recording studio and electronic music performance. Chief among the new tools are computer automation and digital signal processing, and a new awareness of surround sound has given a boost to the development of multichannel mixing and diffusion consoles for theater. Granted, many of these systems are limited by the prevailing paradigm for surround sound, namely the 5.1 format deriving from cinema sound, but a number of newer matrix mixing systems now provide output capacity ranging from 8 to 512 discrete outputs.

A truly compelling multimedia experience is incomplete without the highest quality of audio reproduction. The ear is more difficult to fool than the eye, but the extra effort is essential. Hearing is a more intimate sense than vision. Sounds envelop us and touch us, while visual images remain at a distance. As the American playwright/director Richard Foreman wrote about his play "Film is Evil, Radio is Good," visual images enter the eye and go direct to the brain where they are intellectualized. Sounds enter the ear and go straight to the heart [3].

1.2 Requirements for Modern Audio Systems

Multichannel sound systems are called upon for a wide range of functions, including electroacoustic enhancement, sound reinforcement, theatrical sound, and large scale distributed audio systems. Each application domain presents distinct challenges. However, they can be fundamentally served by a core system that adheres to a set of design principles.

Full audio automation: All elements of the audio control system must be programmable. The character of the audio environment may change from scene to scene, and processing of mix, equalization, delays, and reverberation all contribute to the representation of a desired audio environment. The computer software and host operating system should be robust, fast, and easy to operate.

Dynamic control: Real-time updates of faders, mutes, and panners are often required to achieve dramatic effect. The system must be able to fade and pan distinct audio elements smoothly and at different rates, with individually programmable offset times. Operators must be able to override the automation quickly to respond to changes (planned or otherwise) in the performance.

Flexible electronic architecture: Often times, the requirements of the audio system will evolve during the production period of a multimedia event. It may be determined, for example, that more speakers are required to achieve a desired effect, or additional signal processing is required for the processing of actors' microphones. The core of the audio system should provide sufficient flexibility to accommodate potential changes in the production.

Large speaker arrays: Large-scale productions often use over 64 individually programmable speakers. This has become increasingly unwieldy. In addition, specialized speakers such as overhead speakers or shakers under the floor or under a seat require precise programming to maximize their effectiveness. This control must be efficient, effective, and easy to use. Non-standard and non-optimal speaker placement must be accommodated.

Scalability: It is frequently desirable to produce multiple versions of the same event, sometimes for venues with varying requirements for numbers, types, and placement of speakers. The amount of reprogramming necessary to deploy a production in a new venue must be minimized. A system that is scalable will allow for the addition or subtraction of audio inputs and outputs while maintaining the essence of the original programming material.

Distributed Audio Transmission: It is often necessary to transmit multiple channels of audio over a long distance, particularly in theme parks or other outdoor attractions. Or perhaps some audio I/O modules are to be positioned at the stage, and others at the front of house mix position. The audio system should accommodate multichannel audio transmission.

Audio Playback: Playback of multitrack audio should be integrated to allow the same mixdown capabilities as external audio sources. It is often required for audio sections to play back asynchronously as well as synchronously. Themed environments often require the looping of audio elements to build up a complex audio bed.

Electroacoustic Enhancement: Facilities must often accommodate a variety of uses that have conflicting acoustical requirements. The ability to modulate the acoustics electronically to suit the performance is desirable [4].

Control of External Devices: The audio system is one piece of a puzzle that interconnects the automation of video, lighting, and scenic elements. The audio system should be able to send and receive arbitrary messages to allow the facile integration into the overall show control system. This should include the ability to play back automation events at specific times designated by time code and time of day. Audio events should be triggered via external closure or serial messages. This provides the system designer with flexible control and execution of the preprogrammed automation sequences.

Tactile Control: Although computers are excellent tools for programming automation and displaying a large amount of information in a small space, their use is problematic during live control of a sound system. Generally only a couple elements may be changed at once using a mouse or keyboard. Control surfaces with faders, knobs, buttons and displays are necessary elements of a sound system used with on-stage performers. Motor memory of control element position helps the mix engineer pay attention to the action on stage while making changes to the sound.

Distributed (Remote) Control and Monitoring: The system should be controllable from multiple sources simultaneously. Communication using MIDI, RS-232 and RS-422 serial, and Ethernet (TCP/IP) should use a common underlying published specification. External control systems should be able to interrogate the system for status information.

Fault Tolerance: The system is only as strong as the weakest link. Subsystems should be designed to accommodate redundant or alternative components.

Alarm System Interface: The system should be controllable via simple contact closure to allow integration with and support to other facility alarm systems.

2. MATRIX3 STRUCTURE

The Matrix3 Audio Control System is a new system that is designed to be a flexible platform upon which to implement solutions for a wide range of applications.

The central component of Matrix3 is a three rack-space module with expansion slots. Multiple frames may be linked as a larger system by sharing a common digital audio and control data bus. This bus utilizes industry standard Fibre Channel technology [5], and provides 400 channels of sample-synchronous audio. 32-bit floating-point samples at 48kHz are supported. Maintaining audio in floating point format throughout the system increases internal dynamic range and provides for consistency when dealing with different I/O formats.

To provide sufficient signal processing capabilities and minimize power requirements, each frame includes a main DSP card that utilizes the Texas Instruments 6701 DSP [6], supplying 600 million floating point operations per second.

Audio I/O cards may be added to the frames in three provided slots. Current and planned cards provide 8 channels analog input, 8 channels analog output, 8-in 8-out AES/EBU, 8-in 8-out ADAT Optical Digital Interface, and CobraNet™ [7]. CobraNet is a technology, licensed from Peak, Inc., which supports the transmission of multichannel digital audio using standard Ethernet technology. The CobraNet card for Matrix3 can be configured for 32 in x 0 out, 24 in x 8 out, or 16 in x 16 out.

A communications card may be added to the frame to provide MIDI, RS-232 and RS-422 serial control interfaces, and SMPTE Time Code input and output.

Four additional expansion slots will accommodate existing and planned expansion cards for additional processing. The link card is used to bus audio and control signals between frames.

The EtherTracks™ card is an embedded computer that uses the Motorola MPC8240 PowerPC at 300 MHz [8]. This card uses a version of the Linux [9] operating system. Ethernet and SCSI connections are provided. This card provides the platform for the control software for CueConsole™ modular mixing surfaces, Wild Tracks™ audio playback software, and external TCP/IP communication over Ethernet. A system may use more than one of these cards. The EtherTracks card can transfer 16 channels of audio to the host DSP, as well as control data. Wild Tracks software allows these 16 tracks to be arranged synchronously or asynchronously. For instance 12 channels might be used for music, and four channels for random access sample-based effects.

CueConsole is a modular mixing surface that in conjunction with the aforementioned audio frames can replace a traditional front of house console. Four modules are provided. "Transporter" is used for cue list control and system masters. "Faders" provides 16 motorized faders and 16 LCD buttons. These are typically mapped to inputs or virtual groups. "Meters+" provides audio metering, highresolution Parameter Adjust encoder, low resolution Parameter Select encoder, Mute, Isolate and Edit Select for 16 channels. These can be mapped to inputs or outputs. "Editor" is used for adjusting parameters such as EQ and delay for up to two channels simultaneously. Two Copy/Paste buttons may be used to compare settings, and to move them to other channels. These control modules connect to the system via RS-422 to a Network Interface. The Network Interface connects to CueConsole Server software that runs either on a dedicated computer or on the EtherTracks card. NetMeters metering software connects to the same server as CueConsole modules. This client/server architecture allows the CueConsole modules and NetMeters software to be distributed throughout the facility, not necessarily in close proximity of the audio mixing hardware.

CueStation™ is the software interface for programming the automation. Two sets of windows are provided to support the signal processing (mixer windows) and creation of cues (automation windows). The hierarchical structure of cues promotes small file sizes and self-documentation.

The SpaceMap™ panning system is integrated with the system. This algorithm was originally developed in 1986 [10] by the author and subsequently refined before its first commercial use in 1993, during two Las Vegas theatrical productions. Cirque du Soleil's Mystère is one of these, and continues to this day as a strong example of the integration of many media to enhance the performance experience.

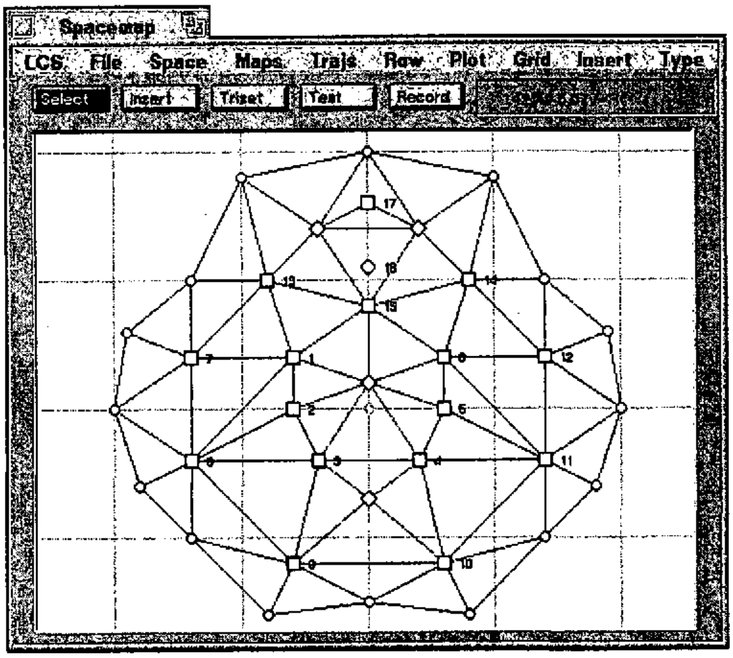

Figure 1: SpaceMap for Cirque du Soleil's Mystère

A SpaceMap is created by piecing together a set of contiguous three-pole panners. Each triangular panner, or "triset", is guaranteed to preserve power amongst the three poles, or "nodes". Nodes may map to any combination of matrix outputs, including none. In addition, outputs may be specified to derive their signal from one or more other nodes. The sound designer may thus create a SpaceMap that uses a highly focused audio image by specifying a single speaker to each node of the SpaceMap, or a more diffuse image by associating multiple speakers with each node of the SpaceMap. The algorithm is efficient to compute, even for a large number of outputs. Flexibility in defining nodes and trisets allows smooth panning to be achieved even with a set of disparate speaker types.

Cirque du Soleil's theater employs two concentric rings of speakers over and around the audience. Vocal performances are literally swung overhead during performance of the trapeze. In this way the sound system is used to enhance the magic unfolding onstage.

Matrix3 provides for both pre-recorded movements and real-time control of audio position. In another themed installation this positional information is fed from an external computer generating 3D graphics and location information in real time in response to user joystick navigation.

In summary, the following table describes how Matrix3 addresses the set of requirements outlined in section 1.

Requirement	Solution
Full Audio Automation	Embedded automation system with hierarchical cue management
Dynamic Control	Programmable fade times per control, trajectories, timecode, timed or externally triggered cues, Automation Isolate, Automation solo
Flexible Electronic Architecture	Modern DSP design, 32 MB RAM, 8 MB FLASH
Large Speaker Arrays	Up to 512 outputs per system SpaceMaps or Matrix presets
Scalability	Modular Frames, Modular audio I/O cards. System size ranges from 8 x 8 to 400 x 512
Distributed Audio Transmission	Native Fibre Channel based 400 channel bus CobraNet
Audio Playback	Embedded Linux computer for Wild Tracks SCSI playback software
Electroacoustic enhancement	Variable Room Acoustics System (VRAS™)
Control of external devices	MIDI, RS-232, RS-422, TCP/IP Message Template definitions
Tactile control	CueMixer, CueConsole
Distributed control and monitoring	Published control protocol with ACK, checksum, interrogation, subscription service, NetMeters monitoring over a network
Fault Tolerance	Relay-enabled outputs Safety Net for audio drives Safety Net for system frames
Alarm System Interface	Contact closures, watchdog relays, analog control inputs, programmable relays

Figure 2: Audio Control System Requirements and Matrix3 Solutions

3. EXAMPLE CONFIGURATION: MULTIPURPOSE THEATER

3.1 System Requirements

To show how these tools can be brought together, following is a design for a theoretical 400 seat multipurpose theater where space is at a premium. For this discussion, the theater is on a new luxury cruise ship. The theater serves three purposes: theatrical performance, classical music recitals, and lectures. The theater was designed for a relatively dry acoustic, with a nominal reverb time of 1.1 seconds. The play has a small cast requiring eight RF microphones. Musical tracks and sound effects are pre-recorded. The schedule in the theatre alternates between the play, the unreinforced performance of a string quartet, and marine biology lectures. During the lecture, a single podium microphone is used, and a DVD video is presented. The string quartet plays mostly Mozart, requiring a reverb time in the room of about 1.8 seconds. The captain would like a feed—in

5.1 surround— to his cabin three decks up. One operator is responsible for all audio mixing and triggering of the lighting cues. The equivalent of six seats is allocated for the mix position.

The system requirements include:

Function	Inputs	Outputs	Notes
RF Mics	8		
8 tracks music playback	8		Matrix3: EtherTracks
8 tracks sample playback	8		Matrix3: EtherTracks
Proscenium		7	
Side fills		8	
Surrounds		8	
Delay Speakers		3	
Room Reverb speakers		6	Split into 12
Remote Monitoring		8	
Room Microphones	12		For Reverb enhancement
DVD Audio inputs	6		
Podium mics	2		
PFL (monitor)		1	
Sends/returns AES/EBU	8	8	to external processors
Spare inputs	4		
Stage monitor		7	
Control Surface			Matrix3: 1 each CueConsole module
Touch panel interface			RS-422 (Comm/Sync card)
Control Lighting system			MIDI (Comm/Sync card)
Electro-acoustic enhancement	16		VRAS processor
Programming computers (2)			Including backup, TCP/IP
Grand Total	72	56	

Figure 3: Example Cruise Ship Theater Requirements

With 1990's technology this system would require four major separate components: a traditional mixing console providing a minimum of 16 mix buses, one or more dedicated room reverb processors, an 8-track multitrack playback deck, an 8-output sample player, and an external DSP matrix for interfacing between the console, reverb system, and speaker system. Presets would be kept to a minimum to reduce the complexity in changing the systems.

With Matrix3, this system could be configured with five LX–300 frames configured as follows:

Frame 1: 16 analog inputs, 8 analog outputs, Comm/Sync card
Frame 2: 8 analog inputs, 16 analog outputs, EtherTracks card
Frame 3: 8 analog in 8 analog out, 8 AES/EBU I/O
Frame 4: 8 analog in, 8 analog out, 8 CobraNet out
Frame 5: VRAS processor

3.2 Mixer Configuration

The total number of system inputs available on the inter-frame digital bus is 40 external sources, plus 16 VRAS microphones, plus 16 Wild Tracks channels for a total of 72. These 72 signals are all available for mixing to any of the frame outputs.

However, direct matrix mixing with so many inputs can be a daunting task. Therefore, Matrix3 provides the designer with a mixer model based on a traditional theatrical sound reinforcement console having an input bus console and output matrix. The number of mix buses may be specified, up to 256. Each mix bus can be matrixed uniquely to the outputs. Any output may be configured either as a matrix output or as an auxiliary output. Auxiliaries are essentially an "input matrix" and are used for outputs dedicated to monitors or effects sends.

Mix buses could be allocated as follows: VRAS (16), Podium Mics (2), Music (8), Sound Effects (8), Vocal Mics (8), Aux Returns (8), DVD (6), for a total of 60 mix buses.

Eight outputs each could be used for effects and monitor sends, for a total of 16 auxiliary outputs.

3.3 Automation Configuration

The automation system uses a hierarchical management scheme. A single project file can contain multiple cue lists. A cue list provides a sequential list of cues, or "dynamic presets". The cue entries in each cue list refer to a pool of common cues in a cue library. Each cue in turn contains references to one or more subcues, which perform the specific automation tasks. This hierarchical system encourages re-use of information and an organizational style that promotes efficient, flexible programming. As a result, a cue can be configured to change every control point in the system, or a subset of these. In our case study four cue lists are set up—one for each show, plus a fourth for a systematic system check to exercise all the equipment in the system.

3.4 Control Systems

The mix position utilizes a modular CueConsole control system, with one each of Faders, Meters+, Editor, and Transporter modules. This provides the operator with a compact arrangement of essential tactile controls to run each function throughout the show. Additionally, an external touch panel is provided at the podium so that during lectures the system may be operated by the presenter. A minimum set of buttons to trigger cues is provided. A PC is used to program cues, and a RIF-108 is available as an additional controller.

3.5 Wiring and Equipment Layout

Most of the Matrix3 frames can be located in an equipment rack outside of the main theater. One module is located at FOH to interface with the effects units in the operator's rack, and linked with the main set of frames using the native digital link. This link transmits audio and control data. The operator uses the CueConsole modules and PC for all control and monitoring functions. The 5.1 remote feed is transmitted via CobraNet to a dedicated CobraNet-to-analog audio box in the monitoring room.

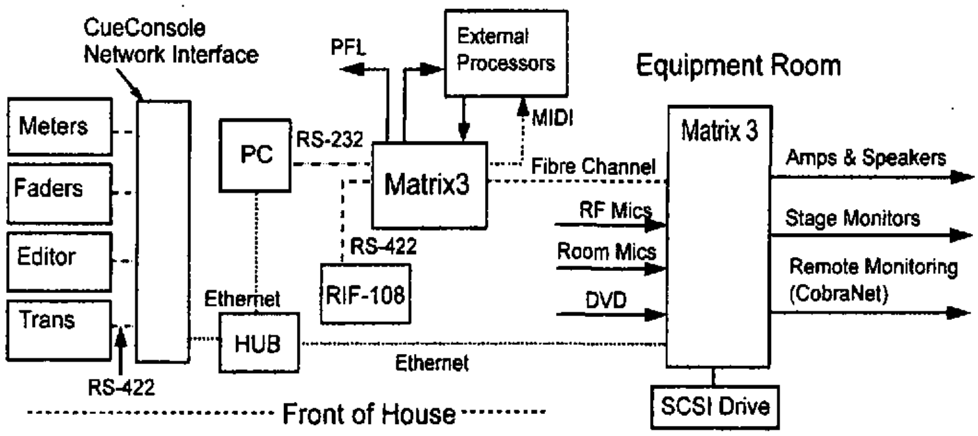

Figure 4: System Block Diagram

4. Conclusion

Multichannel audio system design goals have been established through the evolution of the public's demand for higher quality audio entertainment experiences. To meet these goals the Matrix3 audio control system has been developed. For the first time an integrated multichannel system is available that encompasses the full spectrum of audio requirements, including electroacoustic enhancement, sound reinforcement, live theater, and virtual reality. Future work will involve expanding the palette of available signal processing algorithms, expanding processing capability, and facilitating third party development of custom solutions using this new platform.

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