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UNCOLORED ACOUSTIC ENHANCEMENT THROUGH MULTIPLE TIME VARIANT PROCESSORS

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1. INTRODUCTION

Finding the completely natural sounding electroacoustic system, one which raises the reverberant level or reverberation time with no artificial artifacts - has been like a quest for the Grail. The goal is clear, always just in sight, but never completely achieved. In the company of musicians electroacoustics has a deservedly poor reputation. Invariably the cause is acoustic feedback between the microphones and the loudspeakers. Even if it is possible to design a system which is (mostly) stable, it can be shown (10) that in the presence of acoustic feedback the reverberation time of an electroacoustic system will be different for different frequencies - such that the frequency density of reverberant modes thins out as the sound decays. The result is a metallic clang. The mathematics of this problem has been understood since the work of M. Schroeder (1), and is common to both sound reinforcement and acoustic enhancement systems. A conclusion from this analysis - that if coloration is to be avoided a cardioid microphone in a single channel system must be placed within 1 meter of the sound source - is unfortunately inescapable. (2, 3, 4)

Many partial solutions have been found and utilized. One of the best is MCR - Multi Channel Reverberation - which reduces feedback by using an enormous number of channels, each with a very low loop gain. The effectiveness of MCR at reducing the problems of feedback is proportional to the square root of the number of channels, and often 50 to 100 are used. The most common MCR system uses no digital processing in each channel, but if you add a reverberator of some kind the multichannel approach still works. You only have to find some way of positioning 50 or more microphones in the vicinity of the stage. Another idea - Schroeder's pitch shifter - increases gain before feedback by shifting the pitch of the sound a fraction of a semitone between the microphone and the loudspeaker. Mathematically this induces a phase modulation inside the loudspeaker microphone transfer function, which reduces the height of the peaks in the response (and thus increases the gain before feedback) by about 6dB. Pitch shifting works great for speech, but is a disaster for music.

We have developed a method of digital electronic processing which allows us to combine the advantages of both these techniques. The result - multiple time variant processing - allows us to make an acoustic enhancement system with unprecedented freedom from acoustic feedback and coloration - typically 18dB better than a single channel system. We call our system LARES - Lexicon Acoustic Reinforcement and Enhancement System. The Grail has not been exactly found - you do not find this grail, you build it. But with LARES equipment the building job can be surprisingly successful, and less painful than you might expect.

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Unlike several other recent electroacoustic enhancement systems, LARES is open. We are equipment manufacturers. We are happy to show you how a LARES system works, how to build one, and will help you plan and adjust it. But the fascinating task of figuring out what the customer wants and needs, of design, specification, and installation, can be done by a competent independent acoustician or contractor. We need you.

2. A LARES SYSTEM

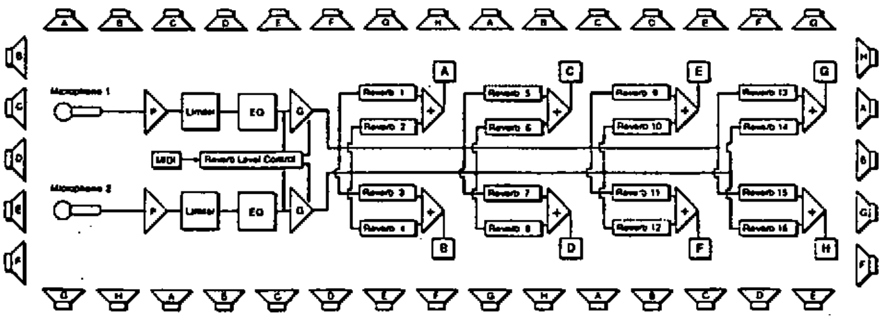

Figure 1. Block diagram of a LARES system

A typical LARES installation consists of a small number of directional microphones (2 to 4) hidden in the forward part of a hall, one or two LARES processing frames, and a distributed loudspeaker system with at least four independent channels. Because of the enormous improvement in gain before feedback provided by LARES the microphones can be far enough into the hall that the entire stage can be covered by a small number. The microphones can be up to a factor of two further from the sound sources than the critical distance (the hall radius). The microphones connect through the usual preamplifiers, eq and limiters to one or more Lexicon LARES processors.

Each LARES processing frame has two independent inputs and four independent outputs. Each output must be connected to a separate loudspeaker bank. The output banks are interleaved in the hall, such that no adjacent neighbor of a particular speaker is driven from the same output channel. The inputs are usually driven by one microphone each. Each frame contains 8 independent time variant processors, one for each connection path between the two inputs and the two outputs. When two LARES frames are used the two inputs of each are usually driven in parallel, giving a 2 input 8 output system with 16 time varying processors in total. The processors are all flexibly controlled through a single small MIDI controller. The hardware used for LARES is also used in a Lexicon standard product, the 480 reverberator. Parts and service are available worldwide.

The adjustment of the LARES frames and the arrangement of the loudspeakers depends on the acoustic requirements. Ideally the loudspeakers are arranged to create a diffuse sound ield. In the Elgin Theater, Toronto, (1500 seats) one frame controls 53 loudspeakers in the eiling of the hall, and another frame controls 53 loudspeakers under the balcony. Both rames are set to the same reverberation time, but different reverberant levels. In a small ~400 seat) concert hall in Concord MA one frame produces a short bright early reflection ield through 8 speakers over the orchestra, and the other creates a concert hall acoustic ver 16 audience loudspeakers.

hus the essence of LARES is:

- A small number of microphones placed up to 2 times the hall radius from the sound sources
- One or more LARES processing frames
- Four or more interleaved loudspeaker banks with enough loudspeakers to provide a diffuse sound field throughout the audience area.

he cost for a LARES system can vary from about $50,000 complete to over $250,000.

3. Prior Systems

evious types of acoustic enhancement system are covered in (10). I will give only a mmary here.

Closely Miked Systems (CMS)

ese systems consist of one or two channels of artificial reverberation fed by a microphone ray mixed into mono or stereo. Many loudspeakers are used, driven in parallel. Stability these systems is low, with the maximum source to microphone distance given by 0.17 nes the enhanced hall radius divided by the square root of the number of microphones xed together. In general cardioid microphones must be within 1 meter of the source. th greater microphone distances the reverberant level created by the system will be too all to be heard while the music is playing, coloration will be high, or both.

Assisted Resonance (AR)

R utilizes 50 to 100+ narrow band channels, each one of which is carefully adjusted to be t within the margin of stability. Systems are expensive, complicated, and are limited in nbre by the number of channels used.

Multi Channel Reverberation (MCR)

CR uses a 50 to 100 broad band channels, each one of which is limited to about a loop in of -20dB. If the channels can be arraigned so they operate without correlation each annel is capable of raising the reverberation time of the room by about 1%. These tems are expensive, and tend to coloration at the end of decays.

RODS - Peter Barnett's system as marketed by Jaffe

DS uses a tapped delay line and a set of switches (variable attenuators). While the input sical level is rising or constant the input to the delay line is open, and the outputs from delay line are off. When the music is decreasing in level or ceases, the input is adually) cut off and the outputs are turned on. The switches are effective at increasing

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stability and reducing coloration, but the system is inherently incapable of increasing the reverberant level or lateral energy while the music is running.

5. Acoustic Control Systems (ACS)

With ACS we get to a more modern system. Many input microphones are used, and many loudspeakers. In theory these are connected together through a delay matrix the elements of which are calculated from an image model of an ideal hall drawn around the real hall. In published papers on this system some type of time variance is claimed in at least some of the matrix elements - how this can be consistent with the method of calculating delays is not explained. In practice ACS could be quite close to LARES - except we think our reverberation algorithms are superior, as is our careful implementation of time variance to both minimize pitch artifacts and maximize gain before feedback. Because LARES lacks some of the theoretical trappings it is likely to be easier and less expensive to install.

6. System for Improved Acoustic Performance (SIAP)

The author does not know enough about SIAP to comment on it. Perhaps we will learn more at this conference. The system appears to be similar to both LARES and ACS.

4. HOW LARES WORKS

Note that LARES, ACS, and SIAP all depend on an extension of the idea first proposed by Franssen for MCR - that with multiple channels system performance can be improved. Both ACS and SIAP explicitly use a large number of input and output channels, and depend on the multichannel effect to achieve a realistic reverberant level with practical microphone placements. ACS uses a matrix to connect the microphones to the loudspeakers, so there is an electrical connection between each microphone and each loudspeaker channel. The matrix was developed to try to create the reverberant field of a different space, but it yields interesting results when it is analyzed for stability.

The analysis which led to LARES recognizes that mathematically there is a different transfer function created by the hall between each microphone and each loudspeaker bank. Lets assume we have M microphones and N loudspeaker banks. Then there are M*N acoustic transfer functions. If electronics are postulated which connect each microphone to each loudspeaker bank we will need M*N connections, or matrix elements. If these matrix elements are fixed - not time variant - then the N acoustic paths associated with each microphone are not independent. They add to form a single transfer function. The total stability of the system will then be determined by the smaller of the two numbers N or M.

However, if make the matrix elements vary in time sufficiently quickly that they have a completely different transfer function after a time short compared to the natural reverberation time of the hall, the N acoustic paths associated with each microphone behave independently. For system stability they behave as if there were a separate microphone driving each. In addition, the phase modulation which results from the time variance can boost the gain before feedback associated with each path.

In a LARES system we have M*N acoustic paths, and in series with each path is a time variant device which provides both statistical independence and sufficient phase modulation to increase the gain before feedback by 6dB. We can find the total improvement in stability over a single channel system by adding to this 6dB the additional gain we get from the multiple channels, the square root of M*N. For a system with 16 reverberators the result is an improvement of 18dB! (This is an enormous number - demonstrations of LARES appear magical to people familiar with electroacoustics or sound reinforcement.) Naturally we must have M*N independent matrix elements - but these can be cheaper than M*N microphones, amplifiers, and loudspeakers.

5. THE LARES REVERBERATORS

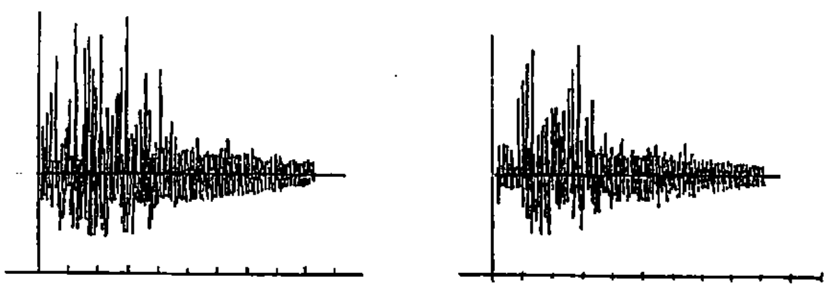

Figure 2. Response of a single time varying reverberator to a positive impulse. The first and second graphs were made 1 minute apart. 50ms/division

These advantages only occur if the matrix elements vary completely and quickly. Figure 2 shows two impulse responses from a LARES matrix element separated in time by about 1 minute. As you can see there is nothing in common. The actual rate of change is much faster. Typically after 1 second the autocorrelation of one reverberator with itself is zero. The trick is to create such a high degree of change with minimal pitch shift. Even with a special algorithm the adjustment is critical. When we adjust for best feedback performance there can be a slightly noticeable pitch uncertainty in solo piano. A compromise must be made. Other instruments appear to be fine.

High gain before feedback is not the only requirement for these matrix elements. You also want smooth, dense decays with no audible coloration. Here we are helped by our nearly 15 years as the world's primary source for high quality studio reverberation. The trick is to preserve this performance in the presence of acoustic feedback, and the time variation in the LARES algorithms does this.

We have given the reverberators a great deal of flexibility. The rate of build-up of reverberation, the time over which the reverberation holds relatively constant, and the eventual decay can all be separately set. The reverb time can be adjusted in three frequency bands. The delay before the onset of reverberation can also be set. Recent theories of the perception of reverberation (11) predict that these adjustments should be sufficient to duplicate any desireable acoustic space - from hall to stage. Our experience so far indicates this is correct. Note we do NOT try to simulate strong fixed reflections, even though these exist in some halls. The software can do it - but a fixed delay in this system triggers feedback. We have theoretical and experimental evidence that strong fixed delays are both unnecessary and undesirable.

Reverberation is not the only trick the LARES software can do. There is also a program called "anti-feedback" which applies the LARES time variance to as few as two delays. This program can be used to reduce feedback in a sound reinforcement system with four or more output channels. It is currently being used on a soundstage in Disneyworld, Florida, and will soon be installed at Eurodisney.

6. DECAY COMPENSATION

Even with the LARES time variation we found at the Elgin Theater Toronto that high amounts of acoustic feedback in the system could cause an audible coloration on speech, and a non-exponential decay. To reduce these problems a dynamic compensation circuit was developed which works in a fashion similar to RODS, but with the opposite sign. When the system detects the input signal has dropped below the reverberant output of the system - that is when the system itself is producing the only audible sounds - the input to the reverberators is attenuated. Coloration is greatly reduced. The action is transparent - the circuit only operates when there is no direct input, which is the only time coloration on the tail of the decay is audible. As soon as any new sound appears full gain is restored.

7. VOICE DETECTION

An additional feature is the ability to detect the level patterns which differentiate human speech and music. A circuit can be activated which raises the reverberant level as much as 6dB when continuous music is present. This circuit works amazingly well in opera, and was used in the last season by the Opera Atelier in Toronto. They liked the full 6dB. I do not recommend more than 3dB for regular classical music.

8. WHERE CAN LARES BE USED?

LARES can be used whenever it is desired to INCREASE the reflected energy level or reverberant level in halls or rooms. There are many obvious applications in medium size (1500 seat) halls and theaters which serve multiple functions. The installation in Toronto is a fine example. Here the reverberation time and the reverberation level can be made similar to a shoebox concert hall, even in the under balcony seats. The effect is quite impressive. With the ability to instantly and silently change the acoustics the hall really can be utilized for Shakespeare or romantic symphonies.

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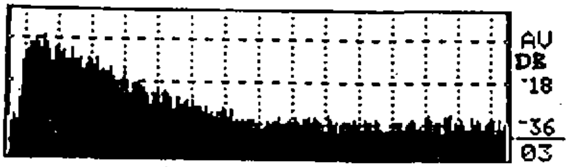

Figure 3. Lateral sound energy under the Elgin Theater Balcony
system off - 180ms per horizontal division - 1000Hz octave band

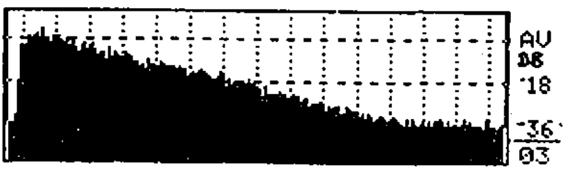

Figure 4. Lateral sound energy under the Elgin Balcony
system on - 180ms per horizontal division - 1000Hz octave band

In a large hall LARES can increase both lateral sound energy and loudness. Here the problem is likely to be loudspeaker placement. A shoebox hall has a strong prompt lateral field, and you cannot in general create such a field from surfaces 30 meters from the audience. Small speakers can be used, but they must be diffuse and close. We look forward to working on such a project, especially with a consultant and a customer who is willing to explore creative solutions to loudspeaker placement.

In a small hall - such as our installation in Concord - LARES must be used in concert with the addition of considerable absorption to the room. Small halls frequently have too much short time delay reverberation, which makes the sound loud and muddy (12). In Concord we added at least 1000 sq ft of absorption in the vicinity of the orchestra, which greatly increased the ability of the musicians to hear themselves. We were only able to do this because the LARES system supplied the needed reverberation. The sound now is clear, bright, and with the reverberant level and reverberation time typical of a much larger hall.

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