FURTHER DEVELOPMENTS IN THE DESIGN, IMPLEMENTATION, AND PERFORMANCE OF TIME VARIANT ACOUSTIC ENHANCEMENT SYSTEMS

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INTRODUCTION

There are a number of very good concert halls in the world that are documented as possessing good acoustics, and artists in residence who have developed patron and audience participation that is sufficient to support both the musicians and the venue. While these venues number in the hundreds, there are tens of thousands of venues that must support a wider variety of programming in order to remain viable. More than ever, the success of a such venues depends on their ability to adequately provide for the requirements of various groups or function. The acoustical requirements for many of these functions are often conflicting. This makes implementing acoustical treatments that will substantially satisfy all of these requirements a somewhat daunting, and expensive undertaking.

Electronic sound systems used to increase speech intelligibility in both dry and reverberant spaces have been used for some time, with largely successful results. Electronic systems that increase reverb time or level have been installed mainly in European halls, but the success of these systems is in doubt. The cause of problems relating to these systems is almost always acoustic feedback between loudspeakers and microphones which causes significant coloration and degradation in sound quality.

Recent developments in technology have produced a system that uses time variant reverberators to significantly reduce the effects of acoustic feedback (1). These systems have an enormous advantage in gain before coloration over previous methods, typically 12dB or more. Such systems are being installed throughout the world, and some of our experience in the course of planning, installing, and tuning these systems will be described here. Given the ability to use this new tool along with standard acoustical treatments, and good sound reinforcement practices, the desired and expected acoustical energy for a variety of performance types can be achieved without compromise to any other. The following is what we have determined to be the important parameters for successful installation of such a system.

DETERMINING THE INTENDED AND ULTIMATE USES FOR THE VENUES

Although there may be disagreement in the acoustical community over what constitutes “good” acoustics, and how one quantifies this, we must identify acoustical requirements for differing types of performances that the venue will undertake which may be directly conflicting. Distinguish as best you can the requirements for speech and clarity from the requirements for envelopment and spaciousness.

Examples of these applications are as follows:

Speech - Clarity and Articulation

- Lecture
- Film
- Dramatic Performance
- Amplified Music

Music - Envelopment and Spaciousness

- Solo musical performance
- Quartet
- Chamber Orchestra
- Symphony

Both Clarity and Envelopment

- Opera
- Broadway Productions
- Chamber Singing
- Orchestra and Chorus

WHAT YOU NEED TO KNOW ABOUT THE SPACE.

Before an enhancement system can be considered, one must take measures to identify and solve blatant acoustical problems within the environment. Noise intrusion from any source - street noise, HVAC, adjacent corridor spaces, etc., should be identified and eliminated. When using electro-acoustic enhancement there is a likelihood that these noise problems, if not resolved, will be picked up by the microphones and amplified uniformly throughout the space.

Our research and work by others has shown that strong specular reflections are detrimental to clarity for both speech and music. These problems should be treated with absorbing or diffusing materials. Since electro-acoustic enhancement systems only add energy to a space, corrective measures must be applied to the venue that produce natural acoustics with energy level and time that are appropriate for those goals requiring the most dry acoustical characteristics. This means that where utmost clarity is desired for the spoken word, much of the reverberant energy in the space must be absorbed. While such treatment is not usually associated with improving the natural acoustics for music, it is also an important consideration for emulating the increased room volume in a larger space.

Room Volume and Critical Distance are linearly related.

Consider a cube with a volume of one million cubic ft. This is slightly more volume than Boston Symphony Hall, although the surface area is less due to the cubic geometry. If we assume an absorption coefficient of .3 for the surfaces of this room, we can predict the reverb radius (critical distance). Since this is a measure of the direct to reverberant ratio of the environment, it can be used as a means of determining room gain. If we were to reduce the volume of the cube by half to 500,000 cubic feet while maintaining the same absorption coefficient, the critical distance will decrease, indicating that the level of the reverberation has increased, while reverb time will decrease. (Figure 1).

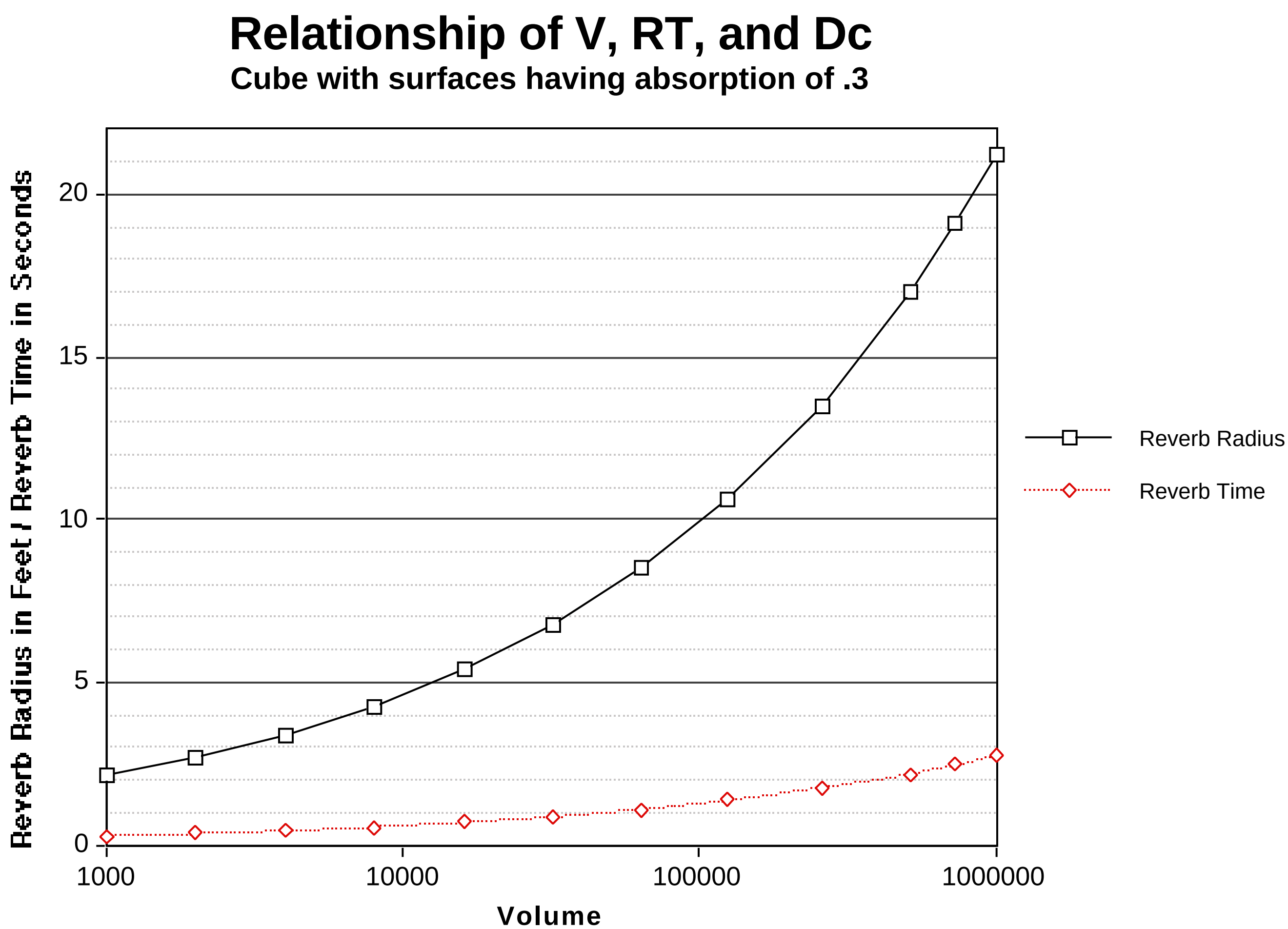

Figure 2 depicts measurements of impulse responses made in the Kulas Auditorium (500 seats) and Severance Hall in Cleveland, Ohio (1890 seats). Note that in the Kulas Auditorium the reverberant level has increased nearly 3dB; while the reverb time in both spaces is similar.

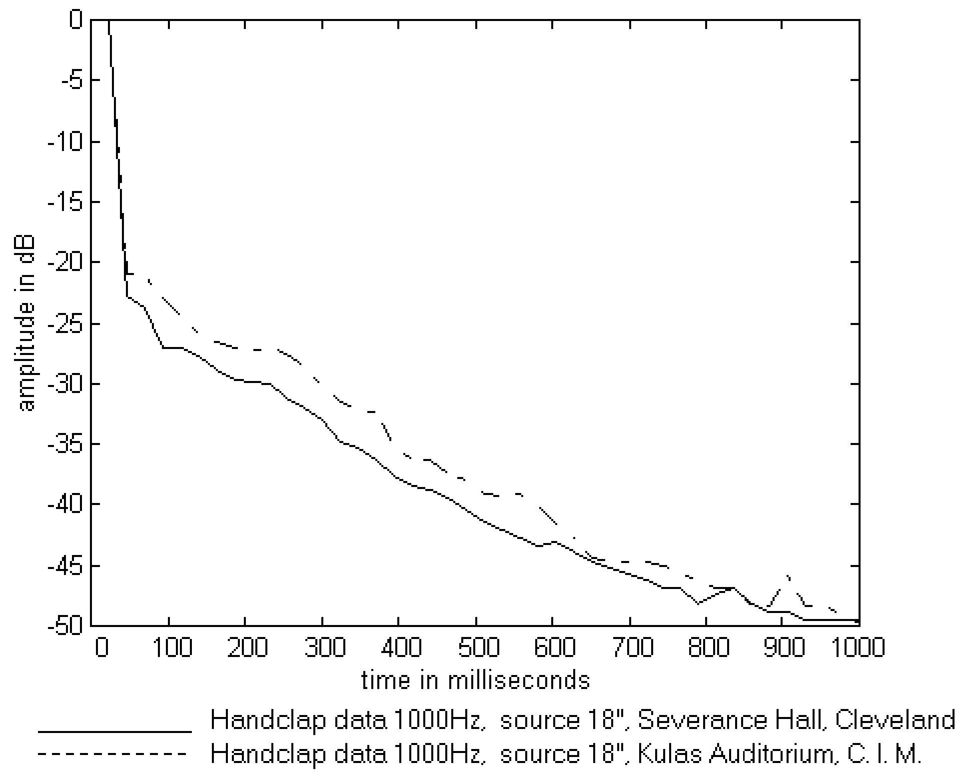

Figure 2.

In smaller rooms (<400 seats) , the audience is potentially closer to the surfaces of the space, and the direct sound. This means that unless the space has a significant amount of absorptive materials other than the audience, the direct and early energy will be loud. This can be detrimental for goals that require very high clarity, and for musical goals that require the acoustical character of spaces with larger volume.

The implications for electronic enhancement of small and medium sized performance venues are that if you wish to successfully emulate the reverberant characteristics of a larger space in such a venue, one must absorb a significant amount of the direct and early energy. A logical place to start is the stage house. Adding absorption and diffusing surfaces will both reduce the early energy contributing to the loudness of an orchestra on a stage. It will also provide the impression of a greater hall return. Consequently, the level of early energy in the space will necessarily be lower as well.

Smaller spaces have audience seating that can be described as "intimate", or closer to the musicians. Reducing the level of direct and early energy in such spaces allows the enhancement system to increase the ratio of direct to reverberant energy for seats closer to musicians, providing a greater sense of envelopment. This can be achieved without raising reverberant levels so high as to sound unnatural. Such a system would create greater uniformity in the levels of direct and reverberant energy throughout the venue, providing optimum spaciousness and envelopment for a greater number of seats.

In medium to large spaces (>500 and less than 1000 seats) there is most often sufficient volume to create reverberation times greater than 1.3 sec. The surfaces of the venue and direct sound sources are farther from most audience members. This means that the direct to reverberant ratio is higher for most seats in the venue, and that the (G) of the room is usually lower. If problems with specular reflections do not exist, it might be possible for such a venue to support several of the goals defined earlier for musical performances, but it would likely have trouble with the extreme requirements for clarity and music, As the size and volume of the space increase further, it is possible for the space to be deficient in both direct, reverberant level, or both dependent on the surface treatments and your position in the room.

Our research has shown that good spatial impression and envelopment occur when the direct and reverberant energy in the room are about equal, when both the early and later energy are diffuse, and when most of the reverberant energy arrives from the side and rear of the listener, and not in the medial plane. In the rear of a large hall where direct energy is weak compared to later energy, the perceived sound may not in fact be become more enveloping. Because energy arriving from the front at this seating location is primarily reflected or reverberant, there is little distinction between early and later energy. Seats in the middle of the house often yield best spatial impression and envelopment. Perceptually this is because where the direct and the reverberant fields can be separately distinguished the directional properties of the reflected field are masked by the direct field, and the reflected field is perceived as maximally wide. Where the direct sound is weak and the total reflected energy comes mostly from the front the envelopment is less.

For enhancement system design in larger spaces, it is important to consider using two separate processors - each receiving input from the same microphones; with one system producing early energy and one producing later energy. These would then be delayed, and mixed accordingly to provide optimum impact and envelopment throughout the space.

DESIGNING THE SYSTEM

Once you determine that there are problems with the natural acoustics of the venue, treat the space accordingly so that it will meet the objectives for goals that require the driest acoustical character.

Hang Microphones

The goals described earlier give you some idea of the sound sources and their locations. Microphones need to be hung near these sources. In most systems installed to date, two or four good quality cardioid microphones are suspended over the stage area as so not to cause sight obstructions (1). The microphones should be positioned as so to pick up a uniform blend of the instruments or performers. WE are just beginning to install systems that have 8 microphones installed and a switching network that enables or disables microphone channels in order to accommodate widely different micing requirements. The advantage to this is that the microphones positions can remain fixed, and gain can be calibrated and reset in a repeatable manner.

Where to put Speakers

There has been significant controversy over the importance of early lateral energy as it pertains to the impression of spaciousness. Our recent work has shown that the identification of spaciousness with only the early lateral reflections having delays ranging from 10ms to 80ms is erroneous, and that it is the total lateral energy with delays greater than 2ms which contributes to spaciousness (5,6). This is particularly important for the fundamental design of an electronic enhancement system. It means that if increased envelopment and reverberance is desired, it is neither essential or desirable that that all of the lateral energy from the enhancement system be delivered in the first 80ms. Because of the frequency dependence of the optimum angle for spaciousness (5), the most pleasant sounding reverberant fields are diffuse, with sound arriving from many different directions. It is not a good idea that all the energy should arrive from the side, hence, it is not a good idea to have most of the speakers in an enhancement system placed on side walls. Our experience has shown that an array of loudspeakers located overhead capable of producing uniform coverage, complemented by a lateral array where necessary, yields best results.

In situations where the low frequency energy of the space is deficient, an array of subwoofers can be incorporated into the enhacement system. Systems installed at the Church of St Michael and St George and the Tsai Center for the Performing Arts at Boston University use subwoofers to augment low frequency energy. Although fewer of these devices are required than with the overhead array, the number should be sufficient as to minimize localization to any one device.

Type of Speaker

The device used should provide uniform power response; specifically, it should not have noticeable aberrations in its off axis frequency response. Mendel Kleiner has stated to me in the recent past that these systems should be built using the best possible components, and here I agree. Fortunately, some the best components for this task are not extraordinarily expensive. Most of our systems utilize a two-way bookshelf type speaker manufactured in Canada and developed in accordance with research generated by the NRC.

Number of Speakers

The ideal speaker for use in such a system is one that produces linear frequency response over a half spherical coverage pattern (a uniform power response), and that has high sensitivity. These requirements are conflicting, and with such a system it is better to use a larger number of devices. A system producing multiple channels of time variant energy will benefit from additional speakers separated by some distance when adjacent speakers have statistically different time variance. This produces a greater number of statistically differing paths between speaker and microphones thereby increasing decorrelation and gain before feedback. Where the ceiling is relatively low (less than 20 feet), larger array density can provide sufficient energy to meet overall power requirements. Where ceilings are higher (above 25 feet) the power uniformity requirements of the loudspeaker are not as stringent. Due to the increased distance between the listener to the speaker array, the listener hears more of the devices in the array, and the aberrations of any one device are diminished. Density usually cannot be compromised where ceilings are high, however; the power delivery requirements are significantly greater due to the increased distance, and coverage must be sufficient to negate artifacts of a more sensitive device. On should strive to maintain power uniformity in whatever device is utilized.

Power Requirements

Reverberance is the musical impression of the hall one hears while the music is playing - running reverberance (RR) is possibly a more descriptive name. We have studied reverberance extensively (3,4,6). We find that for soloists on stage, their impression of the support of the hall can be measured by comparing the energy in the first 160ms of an impulse response to the energy in the second 160ms. We use the measure RR160:

$$RR = RR160 = 10 * \log 10\left(\frac{\int_{160ms}^{320ms} p(t)^2 \, dt}{\int_{0}^{160ms} p(t)^2 \, dt}\right)$$

Note that it is only reflected energy which arrives after 160ms which contributes significantly to musician self support on stages. This result is highly significant for enhancement system design.

For the audience the running reverberance is strongly influenced by the tendency of music to mask its own reverberation. Musical masking pushes the time window for hearing musical reverberance even later - to 300ms and beyond. Our best measure for reverberance in the audience is based on the Schroeder integral of the impulse response. If S(t) is the schroeder integral, and S(0) is the peak, and S(350) is the value at 350ms, then:

$$EDT350 = \frac{60 * 350ms}{(S(0)-S(350)) * 1000ms/\sec}$$

We express this as the slope of an equivalent decay, scaled to correspond to the standard reverberation time. In practice EDT350 is often close to the Early Decay Time (EDT), a standard measure of running reverberance. EDT350 more accurately corresponds to perception over a wide range of listener distance and room decay. Once again notice that the behavior of the decay at times quite long after the direct sound dominate the perception.

This means that for the system to have adequate headroom, the array must be capable of achieving levels that are slightly higher than the point where the release of the direct energy unmasks the reverberant energy in the space. While this measure is dependent on a factors such as the type of music and number of musicians, our experience has shown that the enhancement system for an audience should provide energy to within 4dB of the direct signal for any seat in the defined coverage area. In a small space where the audience seating is very close to the conductor and musicians this requirement can prove problematic. If the performance involves an orchestra (115 dB direct energy), and the first rows of seats are ten feet from the conductor, the distance inverse squared yields a drop in level of 12dB to 103dB, minus masking factor, equals a required energy level of 100dB. If the speaker array generating this energy is 20' or more overhead, it will take considerable power to meet this requirement.

In several system designs, we have made provisions to utilize the enhancement speaker array for secondary sound reinforcement, film surround, and theatrical sound effects. These applications must be considered in calculating array power, but usually are met by enhancement system requirements.

Coverage

Reverberance may vary somewhat throughout a hall, but it is not localizable as a distinct sound source. Therefore an important consideration for any electro-acoustic enhancement system is that one should not be able to perceive the reverberant energy generated by the system as being localizable to an individual sound source. This means that the array density must be sufficiently large as to produce a uniform field, In our experience, the requirements are more stringent than for sound reinforcement. If a power uniform loudspeaker is used, one should strive for 1.5 dB uniformity in the energy generated by the system over the desired coverage area. We have successfully utilized prediction programs such as CADP2, EASE, Modular, etc., to confirm optimum speaker placements. This is accomplished by selecting a very low Q device with uniform frequency response and mapping only direct energy (from the speakers without contribution of room surfaces) using a power sum calculation method.

Our experience is that the number of speakers in the array will not change from the from to the rear of a space. In smaller spaces, the level of the direct energy does not drop significantly, hence the enhancement system array should not change much. Balconies with low ceilings require a greater number of speakers to insure uniformity. Larger spaces require the addition of early energy to after the direct level begins to drop, and coverage requirements for the enhancement system usually keeps speaker numbers consistent from front to rear.

Performance of other electronics used in the system

If we consider the primary use of the electronics in the system to be acoustical treatments, one can appreciate the need to make their operation transparent to both the casual and trained listener. Once inherent noise intrusion problems in a space have been addressed, and the acoustical treatments necessary for speech clarity have been installed, the space itself should not produce hiss or hum and neither should the enhancement system.

System Control

Another weakness in past attempts at electro-acoutic enhancement was adjustability. The system described herein provides control over a wide range of critical acoustic parameters. These changes can even be made during a performance. Our recent experience with multipurpose venues and houses of worship has proven that a programmable control surface should be a necessary consideration for these systems. It allows limits to be placed on the amount of control the user has over the acoustics - and it can accommodate multiple users with differing levels of security and control. System control can integrate well in the environment, both now and later. Furthermore, if the interface is carefully designed, it allows those most familiar with the space to make minor adjustments which ultimately become additional standard modes of system operation.

CORROBORATING THE RESULTS

All the above measures depend on having an impulse response of the space in question. There are many ways to obtain such a response. The recent availability of high quality small microphones and digital tape recorders has solved the problem of the receiver. We use two small microphones mounted on glasses bows, just above the pinnae. This is a simple way of obtaining a binaural response. The big problem is the source. The oldest sources are balloons, blank pistols or small cannon. The newest tend to use dodecahedral loudspeakers driven by rather nasty sounding signals. All of these have their disadvantages. Firearms are difficult to transport, and far to loud for use with unprotected audience. Balloons are very convenient, but they are too loud for enhancement systems, and they are seriously non omnidirectional. Handclaps are quite convenient, but are non omnidirectional above 1000Hz, not very repeatable, and have poor low frequency output. The methods involving loudspeakers are usually heavy, bulky, balky, and tedious for the audience.

The most important principle is that the source, whatever it is, should be omnidirectional. Omni sources are absolutely essential if data is to be compared between different researchers. However we feel it is more important to get some data than no data. For this reason we often use handclaps and balloons - popped in a standardized way. For more precise work we have developed a sweep tone technique which uses a small battery powered dodecahedral loudspeaker. A complete measurement can be made in less than 10 seconds, and the peak loudness is 88dB at one meter. The sweep is audience friendly, accurate, and works with time variant enhancement systems. The software is based on Matlab, and runs in any computer with a 16 bit sound card. The software (but not Matlab) is free.

Once the impulse response has been found, we find it most useful to integrate it with a 40ms window, and then analyze it in octave bands. Both functions are performed by a 1024 sample block FFT, with a 50% overlap. With a 25kHz sample rate this is a 40ms block, repeating every 20ms. The FFT is then displayed graphically, and analyzed into acoustic measures. The binaural data is useful for a number of these. This software is also free.

CONCLUSIONS

For the design of enhancement systems the primary acoustic properties are the intelligibility and the reverberance of the sound. This holds true for both purpose built systems such as a concert hall catering primarily to music, or genuine multipurpose venues whose success depends on the ability to blend both traits faithfully on demand for a variety of applications.

In small acoustic spaces there is almost always sufficient reflected energy from the existing room. Where there are no troublesome reflections in small spaces, intelligibility is often improved by adding absorption, especially around the stage house. Consequently in small spaces it is the reverberance which can be effectively enhanced by a time variant system.

If sufficient absorption is used in the stage house, separate enhancement hardware can be dedicated to adding controlled energy here. The benefits are that this energy can be controlled to accommodate differing performance types, numbers of musicians, and stage setups. In addition, such a system improves uniformity of the energy on stage.

In large spaces electro-acoustic systems can enhance both reverberance and intelligibility. They have the advantage over conventional acoustic systems of being able to do this where it is needed. Reverberance can be increased close to the orchestra, and intelligibility can be increased where the direct sound is low. With modern electronics this enhancement can be done without coloration due to feedback.

In any installation, the ultimate goal for the system is to achieve transparency - adequate care and consideration must be taken so that both acoustical and electronic treatments yield appropriate acoustical characteristics and an enjoyable listening experience for each type of performance the venue must support.

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