A LOW-COST SPHERICAL LOUDSPEAKER ARRAY FOR ELECTROACOUSTIC MUSIC

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ABSTRACT

This describes design and function of a loudspeaker array for diffusing sound outward in all directions from a given point in performance space. This diffusion is similar to that of acoustic musical instruments. Its purpose is to allow acoustic blending of electronically and acoustically produced sounds in music combining the two. Perceptually integrating these is a problem for composers and performers of electroacoustic music. There are various approaches to the problem, each with costs and benefits. A device was constructed with multiple outward facing loudspeakers arrayed around a single point. From an empirical standpoint it provides source point identification and emulates the emanation of sound from an acoustic instrument or human voice when deployed in a space used for acoustic musical performances and without recorded reverberation or post-production reverb effects added. Its use should allow acoustic blending of electronically produced sounds and un-amplified voices or instruments. Tests show a functional frequency response that may or may not require equalization. Performance experiences must be observed and evaluated.

1. INTRODUCTION

This project was greatly enhanced by the work of Curtis Bahn, Perry Cook, Stephan Moore, Daniel Trueman and others on spherical speaker arrays, but it is not concerned with diffusion of sound from electric or digital musical instruments *per se.* Other recent work with spherical arrays has focused on multi-channel devices controlled through digital signal processing for the express purpose of imitating the field directivity of acoustic instruments to accurately simulate them, as in the work of Misdariis, Warusfel and Caussé at IRCAM on bandwidth improvement and optimization of directivity [1].

A digitally controlled, multi-channel array was developed at CNMAT for the same general purpose—more accurately imitating sounds as if they were created acoustically in the immediate space [2]. Work has also been undertaken to control near- and far-field equalization for accurate synthesis of directivity patterns [3]. All these efforts are aimed at synthesis of known acoustic phenomena.

The purpose of this project is to create a low-cost diffusion device for composed or improvised sounds as played back or processed live in the performance of electroacoustic music, so those sounds radiate in a manner analogous to the radiation of un-amplified acoustic instruments used in the performance. Without the attempt to imitate radiation fields of acoustic instruments, this can be accomplished with a one-channel device. The ultimate aim is creation of acoustic *ensemble* without subverting spatial presence of acoustic instruments by their electronic amplification.

In acoustic performance settings, sounds produced by instruments or human voices emanate outward from central areas of focus—the media of vibration and resonation—to diffuse in and interact with the physical limits of performance and listening spaces. For diffusion of amplified sound, loudspeakers are typically placed beyond listening spaces and directed toward listeners within [4]. With respect to source and focus, sounds of acoustic instruments and voices proceed inside-out, while sounds of amplified audio proceed outside-in [5]. Moreover, strength and focus of some acoustic sources notwithstanding, their sonic diffusion is multi-directional, while diffusion of amplified sound is focused along axes on which driver components are centered.

Inasmuch as electroacoustic music represents an attempt to integrate acoustic and electronic sounds, their differing properties and perceptions can be problematic. Typical means of integration can bring undesirable costs to the personal dynamics of performing and listening [6].

This project concerns integration of acoustically and electronically produced sound in a way that preserves acoustic spatial presence and static localization for both types of phenomena. In this way, psychoacoustic perception of electronically produced sound depends on arrangement and placement in space rather than the manipulation of electronic signals according to established techniques of computer music spatialization and localization [7]. This proposes to take nothing from established procedures of diffusion and spatialization, rather to offer an additional alternative for integrating electronic and acoustic parts in electroacoustic music, effecting an inside-out acoustic model for both parts rather than an outside-in loudspeaker model.

2. PROBLEMS

The challenges behind developing this mode of integrating acoustic and electronic sound were threefold: 1) Shape one sonic component of electroacoustic music so both are better integrated and less perceptually individuated; yet, 2) Create or preserve spatial acoustic presence and localization for both parts of the sonic texture; and, 3) Develop a system applicable no matter how small or large the acoustic part(s)—from solo instrument to large ensemble.

2.1. Mixing Acoustic and Amplified Sound

It is a primary challenge in electroacoustic music to mix the sonic output of acoustic instruments with the audio output of loudspeakers. As these two phenomena are so different in shape, directionality and focus, they tend to remain separate in the psychoacoustic perceptions of listeners unless some intentional means of coordinating them is employed so they are perceived by performers and listeners alike as integral to the same process. Mixing acoustic and electronic sound is central to the composition, performance, and improvisation of electroacoustic music.

2.2. Amplification and Dissociation

One way integration of acoustic and electronic sound is addressed is by electronically amplifying acoustic instruments or voices. This creates a mixture wherein both elements are diffused through the same outside-in loudspeaker system, but at the cost of what may be considered important elements of acoustic music, in terms of both performer presence and listener reception. Foremost is the immediate material, temporal and spatial association of performer activity with the sounds being heard in performance. In short, the problem is dissociation of performer actions from sonic phenomena [8]. The psychoacoustic and aesthetic costs of amplifying acoustic instruments or voices also include immediate localization, and the spatial presence in performance and listening spaces characteristic of acoustic instruments. These impact the physical, social, psychological and cultural dynamics of music making itself.

2.3. Limits of Amplification

There are limits to the amplification of acoustic instruments. The greater the number of acoustic instruments paired with electronic sounds, the more expensive and technically problematic it is to amplify them. Sounds from a large ensemble can fill a space, per listener perception, without amplification; but, the characteristics of those sounds *vis-à-vis* those from loudspeakers makes it just as hard to combine the two without creating the perception of two distinct and unrelated phenomena. In fact, the more of both is produced, the more perceptually distinct they become. In short, sheer forces of acoustic instruments do not solve the problem of combining acoustically and electronically generated sounds.

2.4. A Reverse Solution

Rather than conforming acoustic sounds to those from conventional loudspeakers by amplifying them through the same system, it was proposed to conform electronically generated audio signals to the properties of acoustically generated sounds by reversing their source-focus relationship from the outside-in orientation and acoustic properties of typical loudspeaker placements to the inside-out orientation and acoustic properties of acoustically generated sounds.

In this way, challenges behind developing such a mode of electronic and acoustic combination were met: 1) One sonic part (the electronic one) is shaped to better integrate both and lessen their perceptual individuation; 2) Localization and spatialization are developed for electronic sounds and preserved for acoustic sounds; and, 3) The system is adaptable to a variety of forces.

3. DESIGN, CONSTRUCTION, AND TESTING

Design elements included shape, material and configuration of a frame or enclosure, selection and wiring of loudspeakers, support in the performance space, power handling, audition for artistic assessment, and testing for technical evaluation.

3.1. Structure

Support and/or enclosure had to be created for the loudspeakers. Considerations included shape, audio production parameters, material, and function.

3.1.1. Shape and Configuration

The way speakers would be arrayed determined the shape of their support. A spherical or polyhedral structure or enclosure was an intuitive choice, as with the work of others on spherical speakers [9].

Given the purpose was never to faithfully reproduce recordings of past performances, it was determined that tuning and frequency response related to dimension and symmetry would be less important, and could be corrected with equalization if necessary.

3.1.2. Plasticity

Adapting suitable bodies manufactured for other purposes was chosen over custom fabrication. Plasticity was important for layperson work as time and expense for professional specialists could be avoided. The final choice was a pair of 37-liter polycarbonate bowls purchased from a commercial kitchen supply. These proved practically effortless to measure, mark, cut, drill and fasten. Fastened and sealed at the rims, and sprayed with an opaque vinyl flecking and clear sealant, the bowls provided a flattened spherical shape for support and enclosure of the loudspeakers.

3.1.3. Sealed Enclosure vs. Ports or Radiators

While introduction of bass reflex ports or passive radiators into speaker enclosures increases bass response, there are tradeoffs in precision, cost and tuning. It was decided to construct a sealed enclosure for optimal diaphragm suspension and precision of transduction. Components were fastened tightly through the enclosure against foam rubber weather stripping to prevent pressure equalization and neutralize vibration. Fiberglass insulation was fastened inside and between the two halves of the enclosure.

3.2. Drivers

Factors in the selection of drivers included power handling, frequency response, sensitivity, design for intended application, circuit limitations related to phase, wattage and system impedance, and cost. Automotive speakers were selected.

3.2.1. Wattage and System Impedance

It was desirable to limit power handling, as total wattage would be a multiple of the number of speakers. Keeping power handling down limited the cost of amplification. Automotive speakers typically have an impedance of 4 Ω, while pro audio speakers are typically 8 Ω. Selecting car speakers effectively doubled the wattage available from a given amplifier.

3.2.2. Design Adaptation

Speakers for car audio are designed for spaces of a few cubic meters. Pro audio speakers are designed to drive sonic output comparatively greater distances. Given localized sound radiating in a way analogous to acoustic sources was desired, automotive speakers were preferred. Pro audio speakers would require matching sensitivities between speaker types (woofers, tweeters, etc.), plus calculating and wiring crossover components. Automotive speakers are typically designed as coaxial modules combining speaker types and crossover components. Car speakers optimized internal component compatibility and integrity, increased modularity, and decreased the overall design burden.

Automotive speakers tend to deliver relatively broad frequency responses, while individual pro audio speaker components small enough for this application tend to deliver frequency responses narrow enough to require one or more sub-woofers for the lower end of the frequency spectrum.

3.2.3. Speaker Specifications

Chosen for this application was a coaxial 2-way model with a 10 cm highly oriented polyolefin cone woofer and a 2.8 cm aluminum cone tweeter. Power handling is rated at 30 W RMS, impedance at 4 Ω, sensitivity at 88 dB/W/m, and frequency response at 45–22,000 Hz. Higher sensitivity models were cost-prohibitive, but the frequency response of the model chosen is a positive.

3.2.4. Wiring Topology

Given limitations in relation to impedance loads, and needing a sufficient number of speakers for omni-directional array, it was determined that 16 speakers would be used. These were wired in 2 parallel sets of 4 parallel pairs in series (see Figure 1) for immediate isolation of any failed speaker. The result was a 1-channel unit with a sustained power rating of 480 W RMS and an impedance load of 4 Ω. A Speakon jack was provided for connectivity. An amplifier rated 800 W at 4 Ω in bridged mode was acquired to drive the unit.

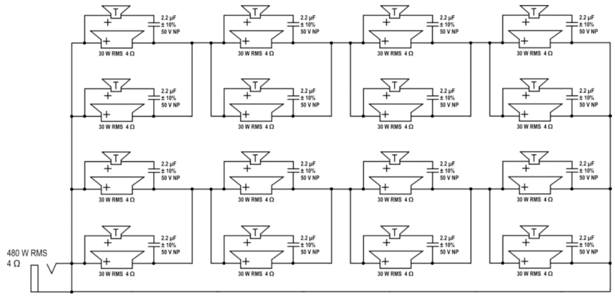

Figure 1. Schematic diagram

3.3. Support

The unit was fitted with 6 chrome-plated lugs and wire legs designed for a floor tom. These support it at about chair-height (see Figure 2). Legs could be re-fabricated for adaptation to commercially manufactured percussion or speaker stands, or for suspending from above.

Support or suspension is important, given the omni-directional design. Hemispherical speakers designed by Stephan Moore [10] and used in a variety of installation and performance applications including the Princeton Laptop Orchestra [11] would certainly produce the desired effect, perhaps with less resonance without downward radiation and acoustic interaction with floors and lower spatial contours (including an audience seated below a stage). It is proposed that for electroacoustic music, dry signals radiating in all directions provide the best acoustic blend with un-amplified acoustic sources.

Figure 2. Completed unit

3.4. Testing

Sonic output from the unit is immediately identifiable as having clear source focus and reverberation when played from the stage of a recital hall seating 275. Playing back recordings made in spaces with high levels of reverberation, or playing sounds with reverb added as a digital effect, disrupts acoustic localization and physical reverberation; however, these become apparent when sounds with a dry signal are played. This suggests composing electronic parts for the unit without post-production reverb or other spatial effects added, so natural reverberation can occur along with that of acoustic sources.

The array was tested for frequency response using a flat-response testing mic and a 1,000 ms sine-sweep. Near-field, far-field (1m) and ground-plane (2m) measurements were taken. The near-field measurement is shown in Figure 3, the far-field measurement in Figure 4, and the ground-plane measurement in Figure 5. These tests show a functional frequency response between 20 and 20,000 Hz. For those aiming at highly accurate reproduction of studio results, the near-field measurement (most accurate for the lower end of the spectrum) suggests boosting the low extremity of the spectrum with an equalizer, applying a smooth downward curve to reach unity gain at about 150 Hz. Another possibility would be to supplement the unit with a subwoofer.

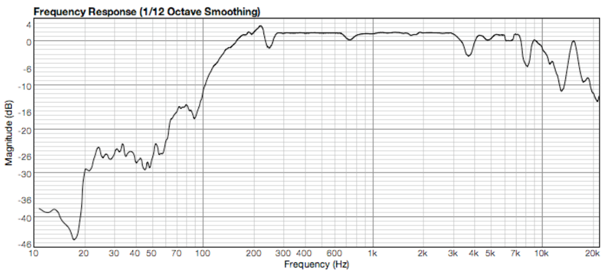

Figure 3. Near-field frequency response

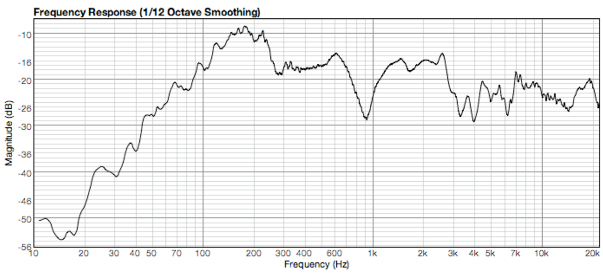

Figure 4. Far-field frequency response

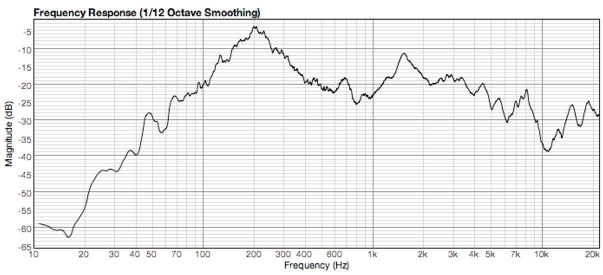

Figure 5. Ground-plane frequency response

4. Conclusion

The unit on empirical observation seems useful for creating satisfactory acoustic spatialization, in terms of both localization and reverberation, especially for signals without recorded or post-production reverb that disrupt sound cutoffs and subsequent acoustic decay. This suggests potential for blending with un-amplified acoustic instruments or human voices. Acoustic tests show a functional frequency response that may or may not require equalization depending on user sensibilities.

It remains for the array to be heard under actual performance conditions. As of this writing, two pieces: *Virtual Duet* for solo bassoon and electronic sound from digitally edited bassoon samples, and *Two Irish Dances* for uilleann bagpipes and automated digital synthesizers have been composed by the author for the device and forwarded to performers for preparation. Yet to be assessed from empirical and aesthetic standpoints are the relative achievement of *ensemble* and overall artistic success of the unit's application.

5. References

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