Spatial Audio Approaches for Embedded Sound Art Installations with Loudspeaker Line Arrays

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ABSTRACT

The concept of embedded acoustic systems for diffusing spatial audio is considered. This paradigm is enabled by advancements in floating-point hardware on inexpensive embedded Linux systems. Examples are presented using line array configurations for electroacoustic music and for making interactive kiosk and poster systems.

Author Keywords

Embedded acoustic systems, sound diffusion, spatial audio, Pure Data (Pd), Raspberry Pi, Satellite CCRMA, embedded Linux, wave field synthesis, embedded sound art installations

1. INTRODUCTION

Some spatial aspects of sound can be simulated using only two loudspeakers, like with stereophonic sound, but then the reproduced sound field is only accurate at a single point in space called the *sweet spot*. In contrast, wave field synthesis (WFS) is an important technique that can be used to widen the sweet spot of sound reproduction systems [6]. It achieves this by controlling the wave field along a boundary in space instead of at only two points [5].

In electroacoustic music, WFS and related techniques are important because they expand the palette of sound perceptions that a composer can create by elaborately controlling arrays of loudspeakers [2].

2. EIGHT-CHANNEL EMBEDDED ACOUSTIC SYSTEMS

2.1 Overview

Already the benefits of WFS can be observed in some configurations using arrays of only eight (8) loudspeakers. Accordingly, the authors have created a series of eight-channel embedded loudspeaker arrays, for exploring applications in electroacoustic music and data sonification.

Each of these arrays has been built to include an embedded computer. Recent advancements have enabled these embedded computers to rapidly make complex calculations using floating-point numbers. For convenience in enabling a wide range of users to program these systems, the embedded computers have been realized using the Satellite CCRMA platform [4].

A *line array* is an array of loudspeakers placed in a line (e.g. see the 8 black loudspeaker grilles placed across the top of Figure 5). Simulations were performed in order to objectively investigate in what contexts WFS may be valid using line arrays of as few as eight loudspeakers.

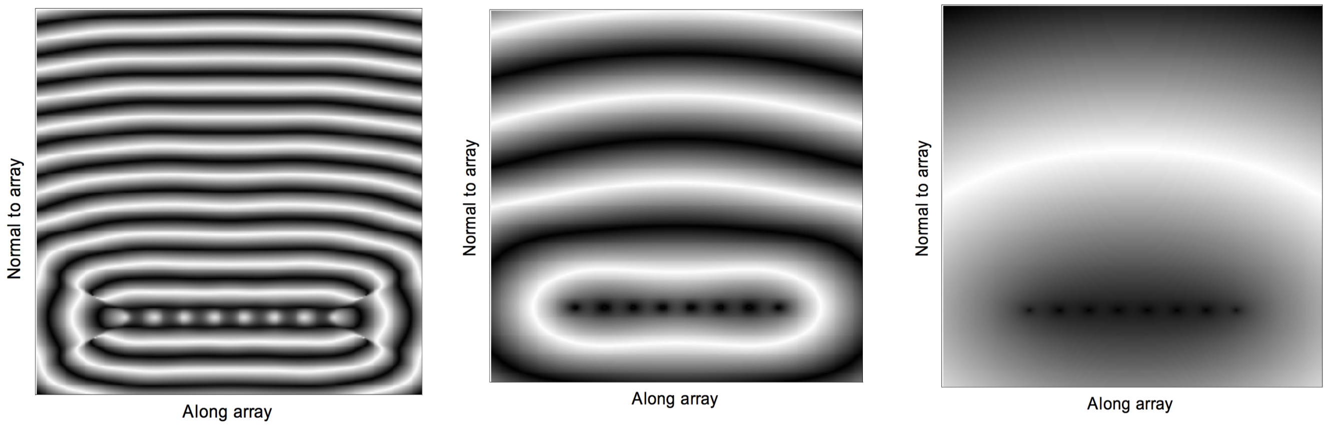
It tends to be easier for such arrays to synthesize spherical waves, as those are the kind of waves radiated by most common loudspeakers. In contrast, plane waves tend to be more challenging to synthesize because many loudspeakers need to be operated in concert in order to synthesize a real wave resembling a plane wave. Figure 1 shows that for line arrays with eight ideal monopole loudspeaker drivers, WFS can produce relatively planar wave fronts over an approximately square region in front of the array across a wide range of frequencies. Figure 2 illustrates how such an array can also approximately simulate relatively planar waves arriving from other angles as well. To a limited extent, plane waves may theoretically be simulated as arriving from some directions other than from behind the array, by relying on the perceptual phenomenon that the human auditory system cannot reliably detect which direction waves are traveling in (i.e. whether time is running forward or backward) [2]; however, this technique was not experimented with in the context of this interdisciplinary work in embedded acoustic systems for diffusing spatial audio.

2.2 Virtual Monopole Sources

Sound spatialization using such arrays can be programmed using a virtual acoustic representation. Other techniques and more advanced software packages are available [1, 8, 7, 3], but the current setup is simply implemented directly in `Pd`, to enhance pedagogical applications and customization within `Pd`.

Each virtual sound source is considered to be placed in virtual space at a certain position (x_v, z_v), from which it radiates sound equally in all directions. It is considered to be a *monopole source*.[1]

[1]In the following, all sources and loudspeakers are assumed to be at the same height "y"-position since a two-dimensional array instead of a one-dimensional loudspeaker



(a) Wave field of high frequency sine wave (b) Wave field of mid-frequency sine wave (c) Wave field of low-frequency sine wave

Figure 1: Simulations of the wave field emanating from the array when aiming to recreate a plane wave (e.g. a spherical wave from a very far-away source). The locations of the drivers are indicated by the eight alternatively displayed circles in a line. The wave field is approximately accurate in a square-shaped in front of the array.

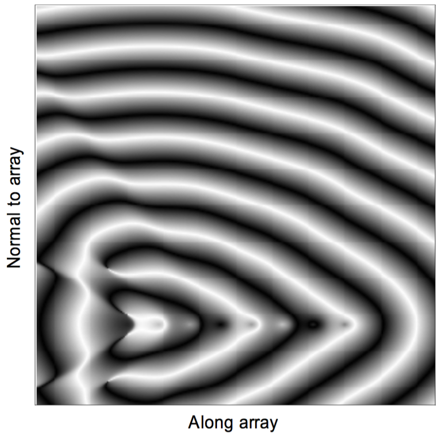


Figure 2: The array can also be employed to simulate wave fronts arriving from some other angles.

Accordingly, the following equation derives the filter that should be applied to a real loudspeaker placed at position (x_l, z_l), where θ_l is the angle of incidence for the virtual spherical wave arriving at (x_l, z_l) [6]:

$$Q(x_l, z_l, f) = \frac{jf \cdot cos(\theta_l)}{cd} \cdot exp(-j2\pi fd/c). \quad (1)$$

The distance $d = \sqrt{(x_v^2 - x_l^2) + (z_v^2 - z_l^2)}$ in meters, the speed of sound $c = 343$ m/s, and f is the frequency in Hz.

To improve the efficiency of practical simulation, the term $exp(-j2\pi fd/c)$ is realized using a delay line with time delay d/c seconds. The term $\frac{cos\theta_l}{cd}$ is realized using a simple gain. The first and last loudspeaker signals are additionally attenuated slightly in order to realize a spatial windowing function, as is common with wave field synthesis [6].

Although the additional jf term implies a 6dB/octave highpass filter, it is currently neglected as noise at high frequencies in the input sound sources could potentially be disadvantageously amplified; however, if some users desire to most accurately synthesize a wave field for a virtual monopolar sound source, they might wish to pre-process their audio using the `hip~` object in `Pd`.

array would be required in order to simulate height.

2.3 Implementation in Pd

Figure 3 illustrates how users can easily prototype their own real-time wave field synthesis simulations according to (1) using the `wfs_monopole~` abstraction in `Pd`. This abstraction is configured especially for these line arrays, but it could be internally adapted to suit other geometries.

The left inlet brings in an audio signal, which is to be spatialized using the array. The second inlet receives the floating-point number z_v that describes how far behind the array the virtual source should be placed (0.1m to 10m away). The third (e.g. rightmost) inlet receives the floating-point number x_v that describes where along the array the virtual source should be placed (-0.4m to 0.4m).

Due to linearity and the principle of superposition [9], multiple virtual sound sources can be simulated simultaneously by instantiating multiple instances of `wfs_monopole~`. A related second abstraction is alternatively available for simulating virtual monopole sources with reverberation.

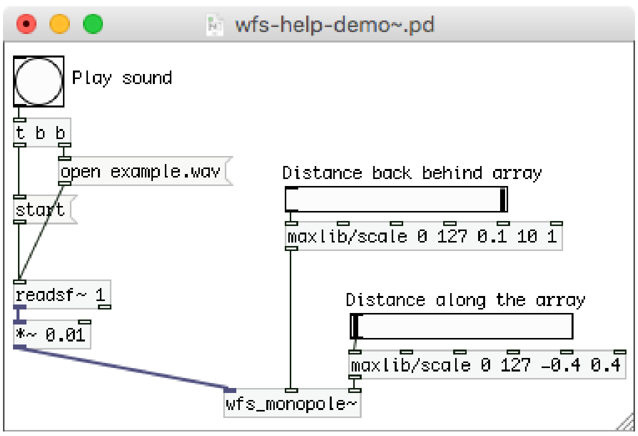


Figure 3: Help patch for the `wfs_monopole~` object.

3. REPORT ON EMBEDDED HARDWARE

This system was implemented using a Raspberry Pi 2. No benchmarks were conducted, but the simple implementation using delays and gains implies that it should be possible to realize a significant number of virtual sound sources in real time.

Eight channels are sufficient to realize wave field synthesis over a limited space, and accordingly eight channels of output were selected due to the widespread availability of Linux compatible-7.1 USB sound interfaces that can be straightforwardly used. Using this kind of inexpensive sound interface, the authors are not aware of any simple way to retain synchronous playback of all channels and to extend the channel count.

4. ARRAY DESIGNS

A series of arrays are designed for the project. They are all based on the concept of line arrays with eight identical loudspeakers.

4.1 Interactive Posters & Kiosks

The dimensions of the arrays are chosen to be appropriate for integrating the array into kiosks and/or posters for data sonification. While this application has not yet been tested, the authors believe that it could be fruitful, providing for high fidelity interactive and immersive sound, enhancing the presence of an interactive poster and providing an extended eyes-free potential for communicating data.

Figure 4: Concept of an interactive poster/kiosk for building inhabited by the authors. Each time a visitor changes the digital poster that is being viewed, a 5-second wave-field-synthesis earcon is played that represents the focus area.

A conceptual drawing of a computer science poster is shown in Figure 4. A working prototype panel (see Figure 5) has been created for this concept. In this version, signs representing the various focus areas in the authors' research group can be illuminated using an RGB LED strip.

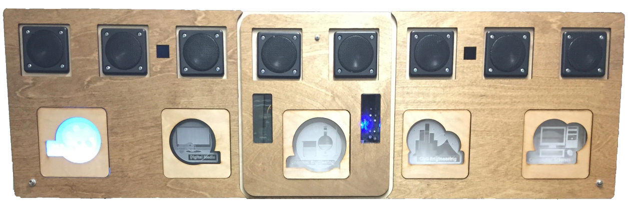


Figure 5: A wave-field synthesis panel representing five focus areas in the authors' research group.

4.2 A Portable Microphone Array for Recording Wave Fields

In the opinion of the authors, employing virtual sound sources (see Section 2) is one of the most straightforward way to employ arrays of loudspeakers such as these. However, it can also be useful, particularly for applications in electroacoustic music composition and sound art, to be able to record sound wave fields. For example, if a sound wave field is measured using a series of figure-of-eight microphones placed analogously to those shown in Figure 4, then the wave field can be approximately reproduced by sending the recorded signals to the loudspeakers in the array shown in Figure 4 [5]. For this reason, a portable realization was created as shown in Figure 6.

Figure 6: Portable eight-channel microphone array being used to record the sound wave field created by a car driving by.

4.3 The Wedge Loudspeaker Array

The Wedge (see Fig. 7) is an eight-channel embedded loudspeaker array, whose enclosure is made of white acrylic plastic, which is *matte* on the outside surfaces. Arrays of multicolor LEDs are installed inside, and the light emanating from them is diffused by the white acrylic plastic, enabling the creation of various kinds of coordinated light displays. Demonstrating a pedagogical example, Fig. 8 shows how an array of visual "dB-meters" illustrates the amplitude of sound being transmitted to the respective loudspeaker drivers.

The shape of the wedge allows it to be installed in many useful environments such as beneath large monitors on tables, on the floor flush against the wall in large array installations, on top of a table, etc.

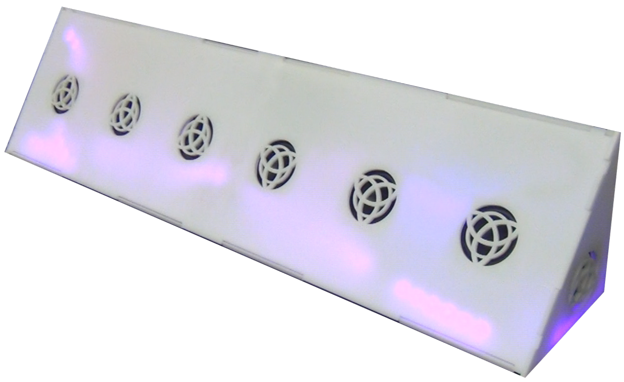


Figure 7: The wedge in action in a bright environment.

Figure 8: The wedge in action in a dark environment.

Careful readers will note that *The Wedge* differs from the other loudspeaker arrays in that the first and last loudspeakers are placed orthogonally on the ends (see Figure 8). This arrangement helps reduce the length of the physical array enclosure. Although it no longer matches the microphone array's geometry, playback of sounds from the microphone array still seems to be quite spatially effective. The amplitude for the first and last loudspeakers is attenuated a little bit anyway as is common in wave field synthesis practice [6].

5. MUSIC FOR THE WEDGE

Matt Williams has composed a musical work for the wedge. The microphone array was employed to record the wave fields emitted by various traditional musical instruments with meditative-type sounds. Then these sounds were processed and rearranged in order to realize the music composition *medit8*.

> *medit8* is meant to help induce a meditative trance, and it explores technology's ability to assist in not only mental endeavors, but also physical and spiritual. The goal is for the listener to synchronize her or his respiration with that of the composition. *medit8* utilizes wavefield synthesis to immerse listeners in a bath of traditional meditative instruments. This process of synthesis involves the use of an array of eight microphones that very accurately captures the spatial qualities of that which is being recorded. This recording is then played back through an array of eight speakers, which are arranged exactly as the microphones were. This will produce very accurate spatial interpretation of what the microphones recorded.

The speaker system, an LED strip, and the Raspberry Pi 2 which contains the sound files, are all contained within an acrylic enclosure which we call *The Wedge*. The LEDs are responsive to the amplitude of the sound going through the speakers.

6. CONCLUSIONS

Informal listening tests show that the more precise control of the wave field seems to enable broader exploration of spatial aspects in electroacoustic music. When listeners stand in the square space in front of the array, the sound no longer seems to emanate from specific "speakers" located around the space, as is often the case for 5.1 or stereo standards – instead, the sound actually seems to be coming from behind the array.

Virtual sound sources can be animated in real-time, which can enhance the spatial effect. Even as a virtual sound source moves from one virtual location to another, the reverberation from the prior location continues. This is achieved by applying a digital waveguide network reverberator independently to each output channel of sound [9].[2]

In future work, the authors are planning to place multiple instances of the arrays adjacent to one another, in order to create a larger "sweet space." For instance, if the arrays are placed all the way around the perimeter of a room, then the whole room becomes the sweet space, under the assumption that passive room reflections are sufficiently dampened using carpet, etc.

Another future application could potentially include placing an embedded line array along a graphic display (as in Figure 4) for use in locally rendering 5.1 audio for film playback in an entire room. According to this paradigm, the 5.1 audio would be "upmixed" to the array using a beamforming strategy. The left, front, and center channels can be directly implemented using the array. The rear left and right channels from the 5.1 standard could potentially be beamformed into the back corners of a room. When the sound reflects from those corners, it would simulate the effect of remotely installed speakers, without needing to install them remotely and wire them up.

7. ACKNOWLEDGMENTS

Aspects of this research have been supported by NSF CNS-1126739.

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