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Virtual auditory scenes created by time reversal mirror technique

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ABSTRACT

The present paper describes the process of generating virtual auditory scenes exhibiting multiple virtual sources in different locations, which is accomplished through the application of the time reversal mirror (TRM) method. This technique, developed by Mathias Fink, can be used to focus an acoustic signal at a particular point in space. Time-reversing the transfer function between a TRM array and an acoustic source generates an acoustic spatiotemporal focus at said source's original location. Thus, this time-reversed focus behaves as a "virtual" source in the outbound direction with respect to the TRM. Provided that an acoustic impulse is previously registered by the TRM device, a "virtual" audio source can be generated at the impulse's location by convolving the TRM impulse response with an audio signal. Since the system is linear, it allows the addition of impulse responses belonging to different locations, which can be convolved with audio signals in order to shape the sound field of the auditory scene. The numerical simulations implemented to explore this method, locate arbitrary audio signals in selected positions of an auditory scene.

Keywords: Virtual sources, Time reversal mirrors, Impulse response.

1. INTRODUCTION

The Huygens-Fresnel principle establishes that each point of a wavefront can be replaced by an infinite set of point sources with same frequency and phase to the original wavefront. One way to implement this principle is to record a wavefront emitted by a point source using a finite array of transducers and then re-emit the record (1). This generates a wavefront similar to that of the point source within a frequency range related to the size of the array and the separation between elements. Mathias Fink (2, 3) proposed a modification to this process that takes advantage of the reversibility of the differential equations of acoustic propagation to make the wavefront converge into a focus at spatial and time domains. The time reversal mirror technique (TRM) allows an incident acoustic field to be re-focused on the position of the original source, based on the reciprocity and temporal symmetry of the wave equation and the coherent recombination of the acoustic fields by taking advantage of the reversibility of acoustic propagation to create wavefronts that propagate as if time had been reversed.

Fink (4) expresses it very graphically by stating that if a person says "Hello" in front of a TRM then will be heard "Olleh" at the point where his mouth is.

Consider the case of an acoustic field sampled by a TRM, with a source in location r=rn. For a source signal time dependence described by function F(t−tn), where tn represents a reference delay, the time-reversed field (TRF) is defined by the Eq. (1)

$$\Psi_u(r,t) = \frac{1}{4\pi R_n}\cdot\left(F\left(t_n - T_n - t\right) - F\left(t_n + T_n - t\right)\right) \tag{1}$$

(with Rn≡ |r-rn| and T≡Rn/c, where c is the sound speed), describes a focusing event where waves converge to (for t < tn) then diverge from (for t > tn) the focal point at location r = rn. In this sense, the TRF focus event can be considered as a virtual source in front of the array (5), as shown in Figure 1.

The time-reversal of the multi-track recording of the array is a fundamental principle for the successful TRM performance, so it seems necessary that the convergent signal will be temporarily inverted.

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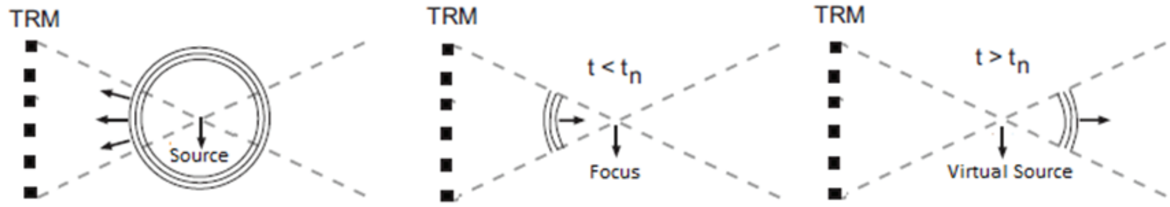

Figure 1 – Virtual source scheme

The time development of speech is critical in acoustic applications in the audible range, however in biomedical, geological or meteorological uses reported in the state of the art (6, 7, 8, 9) work with signals without a discursive or musical content in which obtaining a reversed time development is not considered a problem.

Since the system is linear and time-invariant (3), it is possible to convolve the TRM impulse response with a signal before re-emitting it. From now on, this modification will be called "Convoluted Time Reversal Mirror" (CTRM). In this way a source can be located with the right time order.

By the principle of superposition, using several TRM impulse responses convolved with different audio signals from different locations allows to achieve a full virtual sound scene (10). For example, you could select four positions for the musicians in a string quartet, obtain the impulse responses independently from each of the desired positions, and then process this information to locate the audio signals at any of those points.

This paper examines with three-dimensional simulations the possibility of keeping the convergence of wavefronts without reversing their time development; and also the possibility of overlapping convoluted impulse responses to get virtual auditory scenes.

Section 2 describes the k-space pseudo-spectral method, describing its adequacy for simulating wavefront propagation for TRM. Section 3 tests the correct time development of the signal after convolving the impulse responses, generating a virtual audio source. Section 4 shows the possibility of overlapping TRM impulse responses to create an auditory virtual scene.

2. SIMULATION METHOD

2.1 K-space Pseudo-Spectral Method

Simulate TRM requires specific features that need to be covered by the numerical method used to resolve the differential equations. It must be possible to operate with arbitrary waveforms, ensuring that the signal is not distorted while propagating. This is a limitation for conventional wave propagation simulation methods. Both the finite element and finite difference methods have an intrinsic difficulty called numerical dispersion. When resolving differential equations it is obtained as an unwanted effect a propagation rate that depends on the frequency (11).

On the contrary, the pseudo-spectral method performs the spectral calculation of the gradient using Fourier series which avoids the dispersion produced by spatial discretization. In addition, it allows applying a correction in the error committed by the discrete-time jump when working in homogeneous media.

There is an open source acoustic toolbox for MATLAB called k-Wave that implements the k-space pseudo-spectral method (12, 13). k-Wave allows developing time domain simulations of acoustic waves in one, two and three dimensions using sources with arbitrary time development. This can be generated by mathematical expressions (sines, complex tones, noises, etc.), or even emit audio files that are previously stored in the computer. The validity of the use of k-Wave in the propagation of ultrasonic waves in general and TRM in particular is supported by multiple papers published in the area of biomedicine and medical imagery (6, 7), field where Fink developed his principal contributions regarding this technique (14, 15).

The k-space pseudo-spectral method requires the definition of a rectangular grid of points representing the simulation space. The minimum spacing between grid elements will be determined by the limits of the wavelengths to be simulated (16). By conveniently choosing these values it is possible to adequately represent the wavefronts corresponding to the audible range.

To ensure the stability of the solutions in the pseudo-spectral method it is necessary to consider the CFL (Courant - Friedrichs - Lewy) coefficient described in Eq. (2), which corresponds to the ratio between the distance that a wave will advance in an elementary time step Δt and the grid step Δx (17).

$$\mathrm{CFL} = c\cdot\frac{\Delta t}{\Delta x} \tag{2}$$

2.2 Simulation Settings

The simulations were performed using a CFL value of 0.3 (12) to present an appropriate compromise between precision and computational speed for homogeneous or slightly heterogeneous media. The highest frequency valid for the time functions of the simulation corresponding to the sources or registers is obtained by Eq. (3), where c is the velocity of propagation. Values of $\Delta x = 20$ mm and $c = 343$ m/s were considered, obtaining a $\Delta t = 17.5$ µs and a maximum frequency of 8575 Hz.

$$f_{max} = \frac{c}{2\cdot\Delta x} \tag{3}$$

The rectangular grid used as the simulation space was 128 mm x 128 mm x 64 mm, with 20 mm steps between grid points. The transducer array consisted of thirty-two sensors, with a total length of 2 m. The sensors were evenly spaced with a distance of 64.5 mm from one another.

Based on the classical linear array theory, the frequency limits of the near-field array behavior were determined. The distance between elements relates to the upper cut-off frequency while the total length of the array determines the lower cut-off frequency. The specified values indicate that the array can generate signals between 170 Hz and 2600 Hz.

The sources used in simulations were located in different locations depending on the test to be performed. Impulses were emitted from these sources to capture the response of the system. In order to keep the impulse components within the frequency limits of the simulation (fmax=8575 Hz), the impulse was processed using a Butterworth filter of order 15 with a cut-off frequency of 8 kHz. The pulse duration used was 7 ms to ensure enough propagation time through the simulation space, thus guaranteeing the arrival to all transducers (Figure 2).

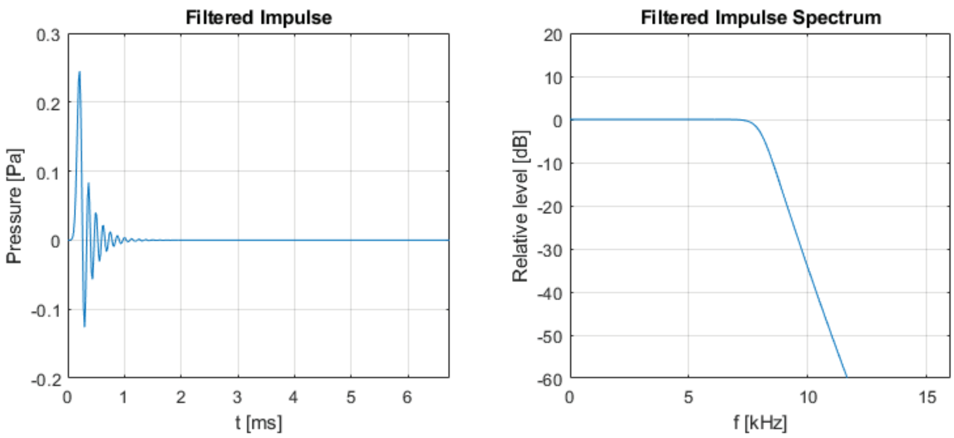

Figure 2 - Time and spectral development of the filtered impulse

3. CTRM WAVEFRONT CONVERGENCE

In this section, a TRM impulse response is convolved with a signal to probe convergence. For this, a source and the array described in section 2 were used, with the distribution shown in Figure 3.

The source emits an impulse and a TRM impulse response is recorded by the array. The response recorded by each transducer is reversed and convoluted with a signal with three harmonic components (300 Hz, 600 Hz, and 900 Hz).

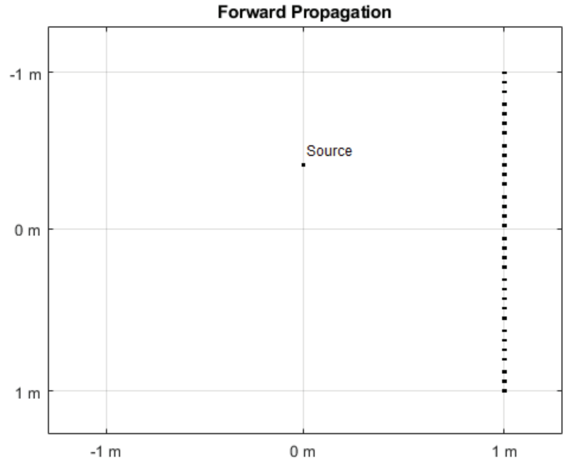

Figure 3 - Cut view of the x-y plane with source distribution and transducers array

To simulate the backward propagation stage of the TRM, the functionality of the array's transducers was modified to behave as sources and a sensor was placed where convergence is expected. In addition, sensors were located at three other points to analyze the differences when the wavefront is recorded at other points where convergence is not expected, as shown in Figure 4.

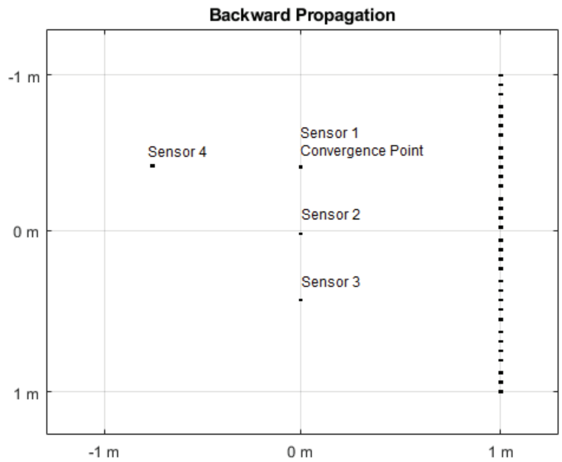

Figure 4 - Cut view of the x-y plane with the distribution of the sensors and the transducers array

Simulation results show a convergent wave at the focal point with the right time order (Figure 5a). The recording of the signal at the convergence point maintains the desired waveform. The signal recorded by sensor 4, which is shifted back from the focal point, shows a decrease in level as shown in Figure 5a, as it is expected considering the convergence point as a virtual source.

On the other hand, in locations far from the focal point but at the same distance from the array, the signal is expected to lose phase coherence altering its shape and level as can be seen in Figure 5b.

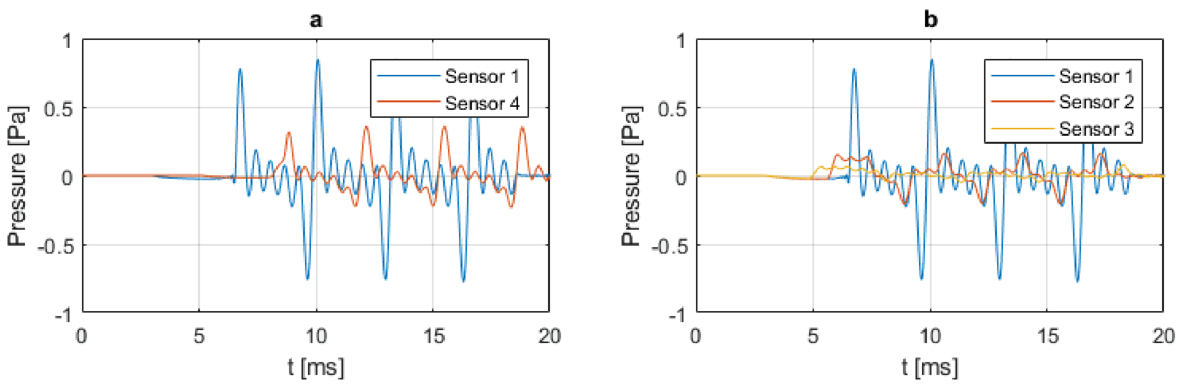

Figure 5 a - Recording sensor 1 and sensor 4. Figure 5 b - Comparison of records at three different locations at equal distances from the array as shown in Figure 4.

4. AUDITORY SCENE BY CTRM

The convoluted time-reversal mirror generates a "virtual" source in the outbound direction with respect to the array. Provided that an acoustic impulse is previously registered by the TRM device, a "virtual" audio source can be generated at the impulse's location by convolving the set of reversed impulse responses recorded by the array (TRM impulse response) with an audio signal. Since the system is linear, it allows the addition of impulse responses belonging to different locations, which can be reversed and convolved with audio signals in order to shape the sound field of the auditory scene.

Figure 6 illustrates the whole process that involves two stages. The first corresponds to the recording of impulse responses for the desired locations, and their subsequent time reversal process. The second requires the convolution of each audio to be located with the corresponding TRM impulse response. The sum of the convoluted results emitted simultaneously gives rise to the appearance of several virtual sources located at different points.

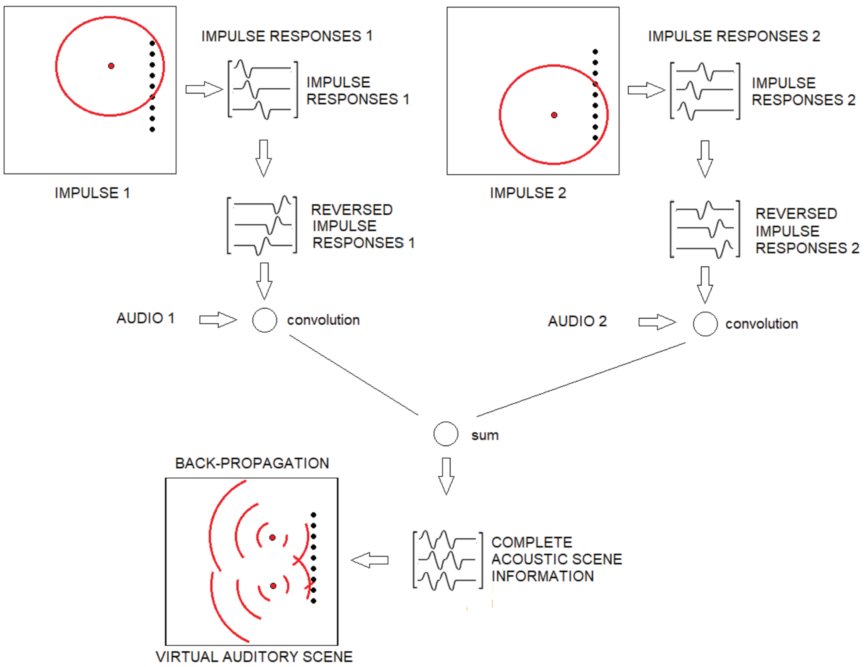

Figure 6 – CTRM process diagram

An auditory scene was simulated using four audios corresponding to a string quartet of 1second in length. Figure 7 shows the selected positions for the musical instruments (for which four sets of pre-recorded impulse responses were available).

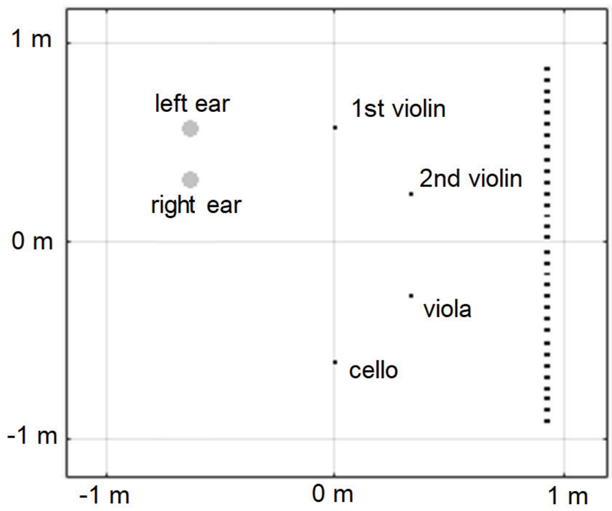

Figure 7 – Musical instruments positions and listening points of virtual sources

To choose the location point of each virtual musical instrument, the audio was convoluted with the appropriate TRM impulse response. As a result of back-propagation of wavefront, virtual sources are generated. Figure 8 shows the records obtained at two listening points.

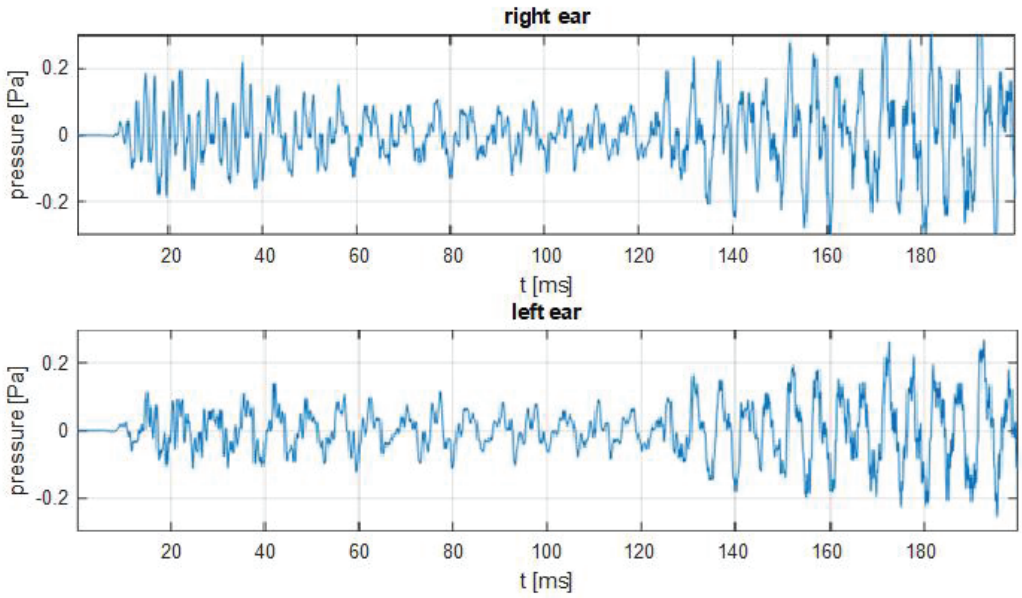

Figure 8 – Results of CTRM listening points

5. CONCLUSIONS.

The results suggest that the use of the TRM technique can be adapted for applications that require recreating an auditory scene in a flexible way. The proposed adaptation (CTRM) requires separating the process into a recording stage, to have sets of TRM impulse responses related to specific locations, and a playback stage that convolves an audio signal with the time-reversal of impulse responses to obtain a virtual source.

Tests show that a wavefront converging at a point using the CTRM technique produces a signal keeping its waveform in focus. Each virtual source generates an auditory field within which the signal maintains phase coherence as if there was a real source at the point of convergence. Outside the auditory field, the contribution of the various elements of the array tends to cancel each other out. There the waves lose phase coherence resulting in an alteration of their shape and a decrease in the level (Figure 9).

The CTRM technique brings another advantage, due to the ability to conserve the source location information without knowing what will be the signal to be convoluted. This allows having previously-stored different registers of locations to process them freely with new arbitrary signals, giving rise to a process of "spatial audio mixing" in which the location and level of the different virtual sources that conform the auditory scene can be freely chosen.

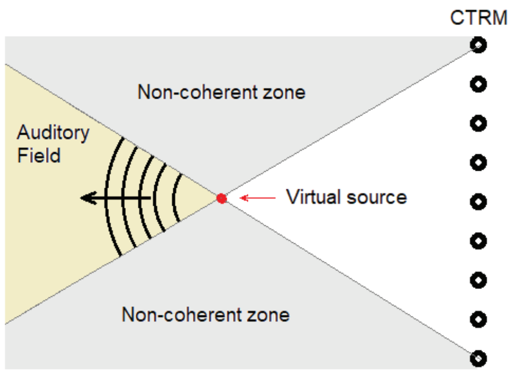

Figure 9 – Impact area of virtual source

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