Fast Fourier-based DSP algorithm for auditory motion experiments

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A digital signal-processing (DSP) technique for rapid generation of complex auditory motion stimuli based on dynamic linear changes in interaural delay is described. In this technique, a pair of complementary discrete Fourier transforms (DFTs) for which the component spacing in one series is different than that of the other is used. The appeal of this technique is its wide applicability, since it can generate real-time motion stimuli of any velocity and starting interaural delay for complex broadband or filtered noise waveforms and nonstationary sounds such as speech, music, and other natural sounds.

The mechanisms that underlie auditory motion perception have been investigated for over a century (Dove, 1839; Peterson, 1916; Rayleigh, 1876; Thompson, 1877, 1878; Valentine, 1928). In the last few decades, motion detection has been examined in psychophysical studies as a function of dynamic variation in interaural cues (Blauert, 1972; Grantham, 1984; Grantham & Wightman, 1978), the role of velocity (Altman & Viskov, 1977; Grantham, 1986; Perrott & Musicant, 1977a; Waugh, Strybel, & Perrott, 1979), movement trajectory (Saber & Perrott, 1990), stimulus spectrum (Chandler & Grantham, 1992; Perrott & Tucker, 1988; Saberi, 1996), as well as other dynamic stimulus features (Grantham, 1986; Perrott & Musicant, 1977b; Perrott & Nelson, 1969; Strybel & Fujimoto, 2000; Strybel, Manligas, & Perrott, 1989; Strybel, Span, & Witty, 1998). In physiological studies, brainstem and cortical neurons have been identified in a variety of species that respond preferentially to a specific direction of auditory-source movement but are nonresponsive to stationary sounds (Ahissar, Ahissar, Bergman, & Vaadia, 1992; Moiseff & Haresign, 1992; Rauschecker & Harris, 1989; Spitzer & Semple, 1991; Stumpf, Toronchuk, & Cynader, 1992). The study of auditory motion continues to be a main topic of research in traditional areas of the hearing sciences (psychophysics and neurophysiology) and is rapidly expanding to new areas, such as human brain imaging. Recent human neuroimaging experiments (Pavani, Macaluso, Warren, Driver, & Griffiths, 2002; Warren, Zielinski, Green, Rauschecker, & Griffiths, 2002) have implicated the parietal lobe and planum temporale in auditory motion processing, although this finding remains in dispute (Smith, Okada, Saberi, & Hickok, 2004).

In spite of its continued importance, the study of auditory motion perception has been limited by a number of difficulties. For example, constructing real-moving sound-source hardware at accurate velocities either in quiet or without extraneous acoustic cues correlated with direction and velocity of source movement has been challenging (Perrott, Saberi, Brown, & Strybel, 1990). Another difficulty, which has become more significant in recent years, is the problem of generating and presenting spatially dynamic auditory stimuli in fMRI neuroimaging studies in which use of real-moving sound sources is impractical (Smith et al., 2004). In addition to these problems, a major complication involves the complexity of simulating motion of wideband, filtered, or natural (e.g., speech or music) waveforms in real time. This is particularly true for motion based on dynamic changes in interaural delay, a major cue to sound localization along the azimuth (Haft, 1984; Rayleigh, 1907; Yost & Haft, 1987). In previous studies, a variety of signal-processing techniques have been used to simulate interaural delay-based motion. Among these are binaural beat waveforms that use pure tones of slightly different frequencies presented to the two ears (Perrott & Nelson, 1969; Saberi, Tirtabudi, Petrosyan, Perrott, & Strybel, 2002, 2003), FM noises that produce nonlinear (sinusoidal) motion (Grantham & Wightman, 1978), and summing time-domain-generated sinusoids combined with a binaural beat paradigm to simulate motion of broadband sounds (Smith et al., 2004). The latter technique, although successful in generating linear motion,¹ is extremely time-consuming for standard stimulus sampling rates and durations (e.g., 0.1–1 sec at 44.1 kHz) and would require calculation of hundreds of sinusoids with independent amplitudes and phases. This would necessitate the generation and storage of numerous independent samples of such waveforms prior to an experimental run, since generating even a single sample of such stimuli between trials is impractical. A further limitation of these techniques is that they generally cannot be applied to nonstationary waveforms, such as speech or other natural sounds.

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Spectral Component Spacing in Complementary Discrete Fourier Transforms

A novel DSP technique is described here that allows calculation of broadband or filtered waveforms between trials, eliminating the need for storage or computational delays. The technique allows generation of a motion stimulus of any bandwidth and motion velocity based on a linear shift in interaural delay.

A discrete time-domain waveform, $x(n)$, $n = 0, 1, \dots, N - 1$, may be represented in the frequency domain by its discrete Fourier transform (DFT):

$$X(k\Delta\omega) = \frac{1}{N} \sum_{n=0}^{N-1} x(n) e^{-j \frac{2\pi nk}{N}}; k = 0, 1, \dots, N - 1, \quad (1)$$

where

$$\Delta\omega = \frac{\Omega}{N}.$$

N is the number of samples, Ω is the sampling frequency, and $\Delta\omega$ is the frequency resolution or bin spacing of the DFT output in hertz (i.e., how far apart the samples are in the frequency domain). Note that the frequency components $k\Delta\omega$ are equally spaced starting at 0 (the dc value) and ending at $N - 1$ ($\Delta\omega$). The spacing $\Delta\omega$ depends on N , which is directly related to the waveform duration. For a fixed sampling frequency, $\Delta\omega$ is inversely proportional to the waveform duration—that is, stimulus duration in seconds ($T = N/\Omega = 1/\Delta\omega$). Thus, spectral energy is measured only at the harmonics of $1/T$.

We use this concept to generate acoustic stimuli at the two ears that simulate a dynamic linear shift in interaural delay (i.e., motion). In its simplest form, the concept utilizes the idea of binaural beats. If the frequency of a pure tone stimulus presented to one ear is f and that of the stimulus to the other ear is $f + \Delta f$, a single auditory image in motion is usually perceived for Δf values of less than 3 Hz as the waveforms at the two ears modulate in relative phase (Licklider, Webster, & Hedlun, 1950). At greater Δf values, motion is replaced by loudness fluctuation, a roughness quality, or two separate images. Motion velocity in this paradigm may be specified as the rate of change in interaural delay, $\mu\text{sec/sec}$, which depends on f and Δf (i.e., for $\Delta f = 1$ Hz, the interaural delay shifts by one period of f each second). Because a component that is higher in frequency than f has a shorter period, in order for a higher frequency component to maintain the same rate of change in interaural delay as the lower component, Δf at the contralateral ear must be proportionately larger and, thus, must have a higher beat frequency. For example, a frequency component of 500 Hz with a 1-Hz beat frequency (500 Hz to one ear and 501 Hz to the other ear) simulates a motion velocity of 2,000 $\mu\text{sec/sec}$. In order for a 1000-Hz component to generate a dynamic shift in interaural delay of the same velocity, it must be matched at the contralateral ear by a beat frequency of 2 Hz (1000 and 1002 Hz, respectively).

The DFT components of a complex waveform are separated by the inverse of the waveform duration—that is, $\Delta\omega = 1/T$ Hz. For a stimulus duration of 100 msec, this spacing is 10 Hz. A 100-msec broadband waveform will thus have energy at $\{0, \dots, 500, 510, \dots, 1000, 1010, \dots\}$ Hz, effectively up to the Nyquist limit. Simulation of linear motion would require component frequencies at one ear that increase proportionately relative to the corresponding component at the other ear. In the present example, a velocity of 2,000 $\mu\text{sec/sec}$ would require component frequencies to the contralateral ear equal to $\{0, \dots, 501, 511.02, \dots, 1002, 1012.02, \dots\}$ Hz. To generate such a dichotic waveform, one simply has to select the appropriate component spacing, which in this case is $\Delta\omega_2 = \Delta\omega_1(1 + V) = 10.02$ Hz, where V is velocity in sec/sec. A time-domain array may then be generated, representing the contralateral waveform with a duration of $T_2 = \Delta\omega_2^{-1} = [\Delta\omega_1(1 + V)]^{-1}$. This shorter duration waveform ($N_2 = \Omega T_2$ points) will result in an appropriately wider component spacing and create exactly the type of matched components at the left and right ears that simulate the desired linear change in interaural delay. This array's bins are then filled with the matched frequency components (amplitude and phase spectrum) of the waveform to Channel 1 obtained from its DFT:

$$X_2(k\Delta\omega_2) = X_1(k\Delta\omega_1) \quad \text{for } k = 0, 1, \dots, N_2 - 1, \quad (2)$$

with nonmatching bins (due to the different number of samples) zeroed out.² Inverse Fourier transform of X_2 will then provide the time-domain signal for this channel.

Sample Waveforms

An example is given here of how complementary DFTs may be generated for a typical auditory motion experiment. Waveform parameters for this example include a duration of 0.1 sec for Channel 1 (e.g., the left ear), a sampling frequency of 44.1 kHz, a velocity of 2,000 $\mu\text{sec/sec}$, and a "distance" traveled of 100 μsec starting at an interaural delay of 0 μsec (i.e., simulating a starting point directly in front of an observer). Assuming that a 100-msec waveform (e.g., Gaussian noise) is generated for Channel 1, the task, then, is to generate the appropriate complementary waveform for Channel 2 (e.g., the right ear).

As a first step, the DFT component spacing for Channel 2 must be determined. The component spacing for Channel 1 is 10 Hz ($1/T$)—that is, the first frequency component after the dc component is 10 Hz. To produce a velocity of 2,000 $\mu\text{sec/sec}$ (0.002 sec/sec), the frequency of the first component (after the dc component) for Channel 2 must be $10 + \Delta f$, where $\Delta f = V/\tau$ and τ is the first component's period (0.1 sec for 10 Hz). This results in $\Delta f = 0.02$ Hz, a frequency spacing of $\Delta\omega_2 = 10.02$ Hz for Channel 2, and a Channel 2 stimulus duration of $T_2 = 1/\Delta\omega_2 = 99.80039920160$ msec. Given a frequency spacing $\Delta\omega_2 = 10.02$ Hz, the 51st component at Channel 1 will

be 500 Hz and at Channel 2 it will be 501 Hz. The 101st component at Channel 1 will be 1000 Hz and at Channel 2 it will be 1002 Hz and so forth.³

To summarize, one should select a waveform duration for Channel 2 that results in the appropriate DFT component spacing at that channel. Having generated Channel 2's array, one may fill in this array's bins with complex numbers from the corresponding DFT of Channel 1 to generate complementary amplitude and phase spectra at Channel 2. Since the two channels have different numbers of points (fewer points for the shorter duration waveform), one must zero out the remaining bins of Channel 1 that do not have complementary bins at Channel 2. This procedure will thus produce exactly the same amplitude and phase spectra at the two ears, with the exception that each component of Channel 2 will be marginally shifted in frequency to simulate motion based on a linear shift in interaural delay. To generate motion that has a starting point other than zero (i.e., directly ahead), the components of one channel may be shifted in phase (linear phase-by-frequency shift; see the Appendix). To generate bandpass motion stimuli, one simply has to zero out those components above and below the passband. Figure 1 shows a sample noise waveform generated using this procedure. Note that Channel 2 contains a noise waveform that progressively shifts in position relative to Channel 1. We have generated and used sample broadband and band-limited audio waveforms using this procedure. These stimuli produce percepts of smooth intracranial motion of a punctate auditory image (see Archived Materials).

Finally, we should caution that the selection of appropriate motion parameters is critically important to generating a strong percept of motion. The starting interaural delay usually should not exceed 800 μ sec if one intends to produce motion within the natural range of interaural delays. This value corresponds to a sound source positioned opposite one ear (displaced horizontally by 90° from di-

rectly in front of the listener). Nonetheless, the technique described here does allow the selection of an arbitrarily large starting or ending interaural delay, which, although unnatural, may be of use in investigating the limits of binaural processing (Saber, Tirtabudi, et al., 2002). Furthermore, although this DSP technique can generate motion of very high velocity, it is not likely that stimuli with velocities greater than several thousand μ sec/sec (several hundred angular degrees per second) will be perceived as moving (Saber, Takahashi, et al., 2002).

Summary and Conclusion

A DSP technique has been described in which complementary DFTs with slightly different frequency component spacings are used for rapid generation of auditory motion stimuli based on a dynamic change in interaural delay. The advantage of the technique is that it eliminates the need for storage of large numbers of complex stimuli prior to an experimental run, and that it is able to generate motion of any velocity for any type of complex sound, including nonstationary waveforms such as speech.

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Figure 1. Gaussian noise waveforms generated using the complementary discrete Fourier transform algorithm. The two waveforms start in phase and gradually drift apart, simulating motion based on a dynamic change in interaural delay.

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NOTES

1. Here, *linear motion* refers to a linear change in interaural delay approximating a constant velocity source at a fixed distance from the observer.
2. For this example, the highest frequency components of the original waveform will be eliminated. However, given a typical sampling frequency of 44.1 kHz, the eliminated components are above the highest frequency audible to humans (20 kHz) and, thus, inconsequential. To retain *all* points from the original waveform, one may set $\Delta\omega_2$ smaller than $\Delta\omega_1$, resulting in an array with a greater number of points than the original waveform [i.e., $\Delta\omega_2 = \Delta\omega_1(1 - \epsilon)$]. The effect is simply a change in the direction of motion (see the Appendix).
3. The duration is rounded to the nearest digital sampling period, which, for a sampling rate of 44.1 kHz, is $\sim 23 \mu\text{sec}$. Thus, the maximum quantization error in estimating duration is $11.5 \mu\text{sec}$, which, for a 1-sec stimulus, results in a maximum frequency shift of less than 0.00002 Hz.

ARCHIVED MATERIALS

The following materials associated with this article may be accessed through the Psychonomic Society's Norms, Stimuli, and Data archive, <http://www.psychonomic.org/archive/>.

To access these files, search the archive for this article using the journal (*Behavior Research Methods, Instruments, & Computers*), the author's name (*Saberi*), and the publication year (2004).

FILE: Saberi-BRMIC-2004.zip.

DESCRIPTION: The compressed archive file contains five files:

stationary_noise.wav, an audio file of spatially stationary burst of Gaussian noise.

stationary_speech.wav, an audio file of spatially stationary recorded speech.

noise_in_motion.wav, an audio file of spatially moving Gaussian noise, with a starting interaural delay of 800 μsec leading to one ear and an ending interaural delay of at 800 μsec leading to the contralateral ear, with a motion velocity of 800 $\mu\text{sec}/\text{sec}$.

speech_in_motion.wav, an audio file of spatially moving recorded speech with a velocity of 500 $\mu\text{sec}/\text{sec}$.

motion.m, an executable Matlab program that loads any mono file (labeled monostimulus.wav) and transforms it into a motion stimulus (labeled motionstimulus.wav). Requires the basic version of the Matlab programming application (a signal-processing toolbox is not needed). All sample waveforms were recorded at a rate of 44.1 kHz. Requires listening through stereo headphones.

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APPENDIX

Matlab Program for Generating Stereo Motion From Any Mono Stimulus

```

clear;
%***** Stimulus parameters *****
[y,fs]=wavread('monostimulus'); %load mono wav file (fs=sampling frequency)
ly=length(y); %Number of samples in y
dur1=(ly-1)/fs; %Duration of channel 1 waveform (in seconds)
t=0:1/fs:dur1 %Time vector
v=250; %Velocity in microseconds/seconds
v=v./1e6; %Convert to s/s
id=0; %Starting value of interaural delay in microseconds
% Usually takes value between -800 to 800
%***** Create Channel 2 waveform *****
if min(size(y))=2;y(:,2)=[];end; %Delete channel 2 if uploaded waveform is not mono
Dw1=1/dur1; %Frequency component spacing for channel 1
Dw2=Dw1*(1-v); %Frequency component spacing for channel 2
dur2=1/Dw2; %Duration of waveform to channel 2
hz1=(0:1/dur1:fs)'; %Array of Channel 1 component frequencies
hz2=(0:1/dur2:fs)'; %Array of Channel 2 component frequencies
lhz1=length(hz1); %Length of Array 1
lhz2=length(hz2); %Length of Array 2
x(lhz2,2)=0; %Create zero array based on longer stimulus
fy=fft(y); %Fast Fourier Transform of Channel 1 waveform
lyy=floor(ly/2)+1;
fy2=hz2.*0;
fy2(1:lyy)=fy(1:lyy); %Fill Channel 2 with Channel 1 components
fy2(lhz2-lyy+1:lhz2)=fy(ly-lyy+1:ly);
x(:,2)=real(ifft(fy2)); %Transform Channel 2 into time domain
x(1:ly,1)=y; %Channel 1 waveform
p1=angle(fy)-hz1.*2*pi*(id/1e6); %Linear phase shift (starting interaural delay)
r1=abs(fy); %Magnitude spectrum
z1=r1.*exp(i.*p1); %New complex numbers for waveform 2
x1=real(ifft(z1)); %Transform Channel 2 to time domain waveform
x(1:ly,1)=x1; %Channel 1 time waveform
wavwrite(x,fs,'motionstimulus'); %write the final waveform to a .wav file

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