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AN ATTEMPT OF DISTANCE MEASUREMENT FOR A HUMAN BASED ON PHASE INTERFERENCE USING BEAM-STEERING AT EACH CHANNEL OF PARAMETRIC ARRAY LOUDSPEAKER

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Basically, it is important to know the distance and position to the elderly and the patient in order to realize improvement of home health care. The authors have already proposed a method to estimate the distance between a microphone and a target using audible sounds, which is based on the phase interference between transmitted and reflected waves. We call this method Acoustic Distance Measurement (abbr. ADM) method and have studied it from various viewpoints. This ADM method is a practical method just to apply Fourier transform to the power spectrum. However, in the distance measurement using the audible sound, there was a problem that when a person spoke in the surroundings, private conversation was recorded etc. In addition, for example, a human face or body has a relatively small reflection coefficient, and is not a flat plate but a complicated aspect, and the reflected wave is likely to be scattered in various directions. This means that we can only observe very small reflected waves or no reflected waves at all. On the other hand, there is a reproduction method of sound field by parametric array loudspeaker using ultrasonic sound. This paper describes a new trial of ADM from sound source to the human as a target using ultrasonic sound. More concretely, as a method for sound enhancement, we attempt to control a phase of the transmitted ultrasonic sound at each channel using the delay-and-sum beamforming technique. In addition, we try to detect the distance to the target, expanding the original distance estimation method by introducing synchronous addition to the power spectra. Finally, the validity of our method is also confirmed through experiment by applying our method to an actual sound field.

Distance measurement based on phase interference, parametric array loudspeaker, beamforming technique, range spectrum

1. Introduction

Japan has entered the super aging society ahead of the world and the medical expenses of elderly people put pressure on the national finances. It is getting harder to care for these elderly people only with medical institutions such as hospitals and medical clinics. Therefore, from the viewpoint of improving the quality of life of the patient, as well as from the viewpoint of lowering the medical expenses, the nursing or health care at home is an important issue in the future. This means that the technologies related to life innovation, namely, nursing care / welfare equipment / robot technology, home and health monitoring technology using information network, medical equipment / devices are required. Among them, since enrichment of home health care is an urgent issue in a super aging society, biological signal monitoring method and signal analysis method are very important.

Basically, it is important to know the distance and position to the elderly and the patient in order to realize improvement of home health care. The authors have already proposed a method to estimate the distance between a microphone and a target using audible sounds, which is based on the phase interference between transmitted and reflected waves [1, 2]. We call this method Acoustic Distance Measurement (abbr. ADM) method and have studied it from various viewpoints. This ADM method is a practical method just to apply Fourier transform to the power spectrum. However, in the distance measurement using the audible sound, there was a problem that when a person spoke in the surroundings, information related to privacy was recorded etc. In addition, for example, a human face or body has a relatively small reflection coefficient, and is not a flat plate but a complicated aspect, and the reflected wave is likely to be scattered in various directions. This means that we can only observe very small reflected waves or no reflected waves at all. So, in order to prevent it, we have introduced ADM based on phase interference for ultrasonic sound[3] as well as audible sound.

On the other hand, there is a reproduction method of sound field by parametric array loudspeaker (or simply parametric loudspeaker) using ultrasonic sound. ADM using parametric loudspeakers has the higher directivity than ordinary loudspeakers using self-demodulation of ultrasonic waves in the air[4]. Also, using parametric loudspeakers, it is possible to perform beam steering to emphasize the sound in a desired direction by giving a delay to a plurality of inputs to the parametric loudspeaker so as to control the phase of the radiated signal[5].

This paper describes a new trial of ADM from sound source to the human as a target using ultrasonic sound. In this research, the distance is obtained by ultrasonic wave as the first step, so the parametric loudspeaker is used only as an ultrasonic array. We use the parametric speaker because a parametric loudspeaker can also present an audible sound. More concretely, as a method for sound enhancement, we attempt to control a phase of the transmitted ultrasonic sound at each channel using the delay-and-sum beamforming technique. In addition, we try to detect the distance to the target, expanding the original distance estimation method by introducing synchronous addition to the power spectra. Finally, the validity of our method is also confirmed through experiment by applying our method to an actual sound field.

2. Theoretical considerations

2.1 Directivity control based on phase control (beam steering)[5]

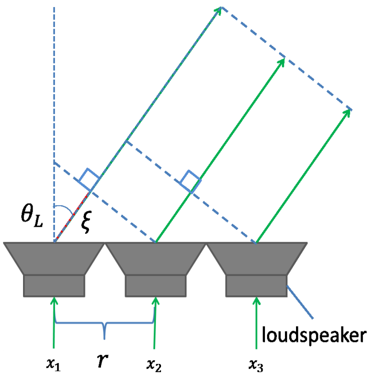
Beam steering is a signal processing technique that controls directivity using array speakers etc. When the sound is radiated in the direction θ_L of interest with multiple speakers as shown in Fig. 1(a), a difference ξ [m] occurs between one speaker element and the adjacent speaker element. Letting c [m /s] be the propagation speed and r [m] be the speaker array interval, the delay time τ_s [s] is expressed as follows:

$$\tau_s = \xi/c = (r \sin\theta_L)/c \tag{1}$$

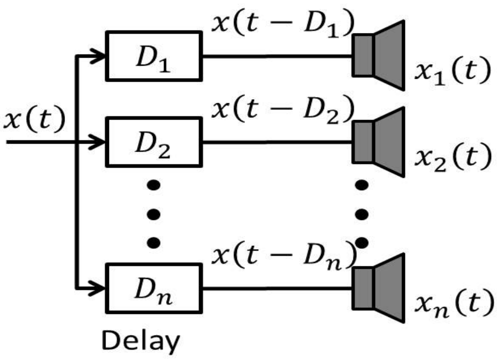
As shown in Fig. 1(b), the beam can be formed in the desired direction by setting the delay D_i to each speaker input. Therefore, using the signal $x(t)$ of the reference speaker, the delayed signal $x_n(t)$ of the n-th speaker is given by the following equation.

$$x_n(t) = x(t - D_n), \qquad D_n = (n-1)\tau_s + D_0, \tag{2}$$

where t is a time [s] and D_0 is a fixed delay [s] to satisfy the causality. From Eq.(2), the output $x_n(t)$ from each delay is the same signal regardless of the channel number n. This means that the time



(a) Path of the transmitted wave in the desired direction θ_L.



(b) Delay-and-sum array.

Figure 1: Principle of beam-steering.

differences of the signals (corresponding to the phases) are adjusted so that sounds propagating in the desired direction θ_L are enhanced by adding the signals with coherent phase, while sound propagating in a direction different from θ_L is not enhanced, even if Eq.(2) is applied, since the phases are not coherent (i.e., signals are temporally shifted). This approach is called a delay-and-sum array which has high directivity to the desired direction.

2.2 Acoustic Distance Measurement (ADM) method based on the interference[2]

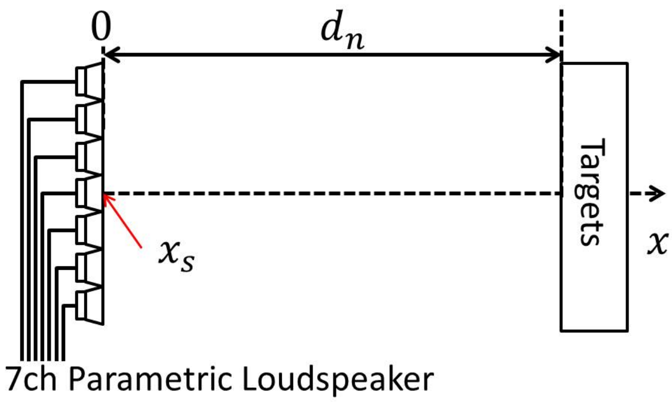


Figure 2: Geometrical position of target and parametric loudspeaker.

As shown in Fig. 2, letting x [m] be an arbitrary position, the wave transmitted toward the target is defined as follows;

$$v_{\mathrm{T}}(t,x) = \int_{f_1}^{f_N} A(f) e^{j(2\pi f t - \frac{2\pi f x}{c} + \theta(f))} df, \tag{3}$$

where $A(f)$ and $\theta(f)$ [rad] are the amplitude and initial phase at frequency f [Hz], respectively. f_1 [Hz] and f_N [Hz] represent the lowest and highest frequency, respectively.

Assuming that there are m targets, the wave reflected by the n-th target is given as follows;

$$v_{\mathrm{R}_n}(t,x) = \int_{f_1}^{f_N} A(f)\gamma_n e^{j(2\pi f t - \frac{2\pi f}{c}(2d_n - x) + \theta(f) + \phi_n)} df, \tag{4}$$

where d_n [m] represents the position of the n-th target and $\gamma_n e^{j\phi_n}$ is the reflection coefficient of the n-th target. For convenience, supposing that the transmitter is set at the origin ($x = 0$) and the receiver at the position $x_s(x = x_s)$, the composite wave (observation wave) at the receiver position is formulated as follows;

$$v_{\mathrm{C}}(t, x_s) = v_{\mathrm{T}}(t, x_s) + \sum_{n=1}^{m} v_{\mathrm{R}_n}(t, x_s) \tag{5}$$

From Eq.(3), applying the Fourier transform to the transmitted wave $v_{\mathrm{T}}(t, x_s)$ at observation position $x = x_s$ yields

$$V_{\mathrm{T}}(f, x_s) = Ae^{-j\{\frac{2\pi f}{c}x_s - \theta(f)\}} \tag{6}$$

with $A(f)$ constant($= A$). Similarly, applying the Fourier transform to the composite wave $v_{\mathrm{C}}(t, x_s)$ yields[2]

$$V_{\mathrm{C}}(f, x_s) = Ae^{-j\{\frac{2\pi f}{c}x_s - \theta(f)\}} \left[1 + \sum_{n=1}^{m} \gamma_n e^{j\left\{-\frac{2\pi f}{c}(2d_n - 2x_s) + \phi_n\right\}}\right] \tag{7}$$

When the magnitude γ_n of the reflection coefficient is sufficiently small ($\gamma_n \ll 1$), by subtracting the power spectrum $p_{\mathrm{T}}(f, x_s)(= |V_{\mathrm{T}}(f, x_s)|^2)$ of transmitted wave from the power spectrum $p_{\mathrm{C}}(f, x_s)(= |V_{\mathrm{C}}(f, x_s)|^2)$ of composite wave and dividing the subtracted power spectrum by twice the power spectrum $p_{\mathrm{T}}(f, x_s)$ of the transmitted wave, we have

$$\begin{aligned} p(f, x_s) &= \frac{p_{\mathrm{C}}(f, x_s) - p_{\mathrm{T}}(f, x_s)}{2p_{\mathrm{T}}(f, x_s)} \\ &\approx \sum_{n=1}^{m} \gamma_n \cos\left(\frac{4\pi f}{c}(d_n - x_s) - \phi_n\right) \end{aligned} \tag{8}$$

This $p(f, x_s)$ in Eq.(8) is a periodic function with respect to f, whose period is inversely proportional to the distance $d_n - x_s$ between microphone and target. Thus, assuming that the receiver is placed at $x_s = 0$ and applying Fourier transform to $p(f, 0)$, we can get

$$P(x) = \int_{f_1}^{f_N} p(f, 0)e^{-j2\pi\frac{2x}{c}f} df \tag{9}$$

The absolute value $|P(x)|$ of $P(x)$ is called a range spectrum, and the peak position of $|P(x)|$ corresponds to the distance $d_n - x_s$ between microphone and target.

2.3 Acoustic distance measurement method using phase control of parametric loudspeaker[6]

In order to estimate the distance and direction to the target (human), we use the acoustic distance measurement method in section 2.2. This time, as shown in Fig. 1(b), the directivity is controlled by giving a delay to the transmitted wave for each column of the parametric loudspeaker, and the range spectrum is obtained based on the signal obtained in the emphasized direction.

3. Distance estimation experiment in actual sound field

We perform distance measurement using ultrasonic sound in the room sound field and verify the effectiveness of the proposed method experimentally.

3.1 Experimental conditions

The band-limited impulse signal is reproduced from the ultrasonic element (transmitter) and recorded with a sampling rate of 96 kHz and a quantization bit rate of 24 bits using an ultrasonic element (receiver). Equipment for recording and playback is shown in Table 1, and experimental conditions are shown in Table 2. As shown in Fig. 3, the parametric loudspeaker has 7 column ultrasonic elements, each of which forms array structure composed of 13 ultrasonic elements , and 6 columns other than the center column act as the transmitter(loudspeaker), while the center column acts as the receiver(microphone). The parametric loudspeaker radiates the same sound source from each column and arranges rows in a close-packed structure. We steered beams in the 30° direction (Fig. 4) and radiated the sound towards the target (human face) 1.5 meters away. We prepared three sets of transmitted signal, each of them is composed of 50 band-limited impulse waves. One of the transmitted waves is shown in Fig. 5. Measurement is carried out by radiating these three sets of transmitted waves (repeating 50 measurements three times). In this experiment, the synchronous addition is applied 50 times over the power spectra and distance estimation is performed with $p_{\mathrm{C}}(f, x_s)$. This is because when the target is a human, since only a small reflected wave is recorded at the receiver, distance estimation cannot be achieved in some cases, and it is necessary to emphasize reflected waves buried in noise.

Table 1: Experimental equipment.

Audio Interface	M-AUDIO, ProFire610
Ultrasonic transducer	SPL (Hong Kong) Limited, UT1007-Z325R
Power amplifier	VICTOR, PS-A1004

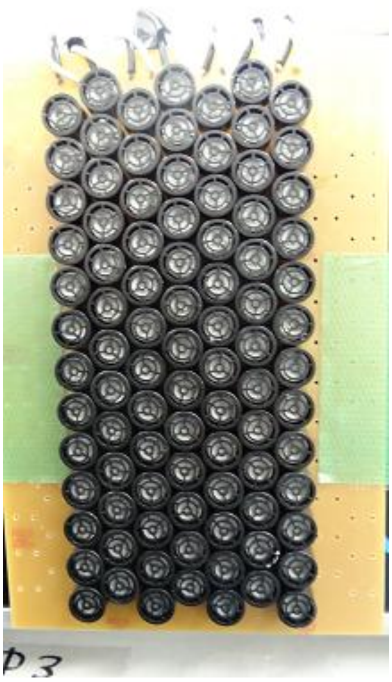


Figure 3: 7ch parametric loudspeaker.

3.2 Experimental results

Power spectra obtained by 50 times addition for 3 sets are shown in Figs. 6(a), 7(a) and 8(a), and corresponding range spectra are shown in Figs. 6(b), 7(b) and 8(b). Here, the minimum measurable

Table 2: Experimental conditions.

Sampling frequency		96 kHz
Sound speed		340 m/s
Frequency bandwidth		6 kHz (37 kHz $\sim$ 43 kHz)
Position of receiver		0 m
Desired direction		30°
Distance		1.5 m
Number of specker array		7
Number of data points	in time domain	4096
	in frequency domain (before 0-padding)	256
	in frequency domain (after 0-padding)	4096

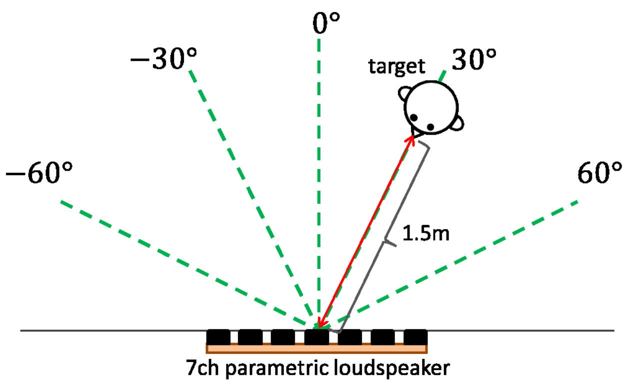


Figure 4: Geometrical position of 7ch parametric loudspeaker and target.

distance (step size of the distance axis) is determined by the bandwidth of the transmitted wave. The data length of $p(f, 0)$ in the frequency region is 256, but since it is expanded to 4096 by 0-padding, the step width on the distance axis is 0.002 m. From the range spectra in Figs. 6(a) and 7(a), we can find the largest peak near 1.5 m which is the true distance to the target and it might be a good estimation. However, in Fig. 8(a), although there is a peak at the position of the target, the peak at the other position becomes larger, and it can be seen that the estimation is not well done.

Table 3 shows the rate of the number of samples with good estimation to total number of samples, at which the distance estimation can be made in each of 50 measurements of three sets without taking synchronous additions. From Table 3, it can be seen that the rates at which the distance estimation can be performed are greatly different, depending on the 1st, 2nd set or 3rd set. Also, in the case where measurement cannot be performed, peaks are around 3.5 m in all Figs. 6(b), 7(b) and 8(b). This is thought to be because the signal reflected from the human face was not directly recorded, but the signal reflected once to a wall etc. were recorded.

Table 3: Percentages of the number of successful trials to the number of all trials.

First set	98%
Second set	100%
Third set	0%

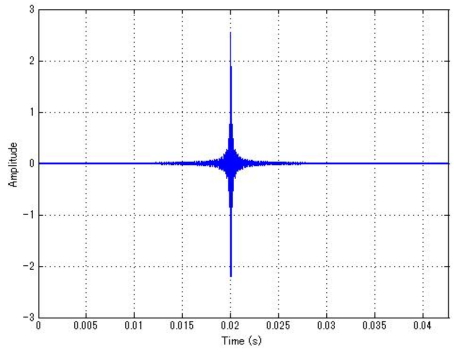
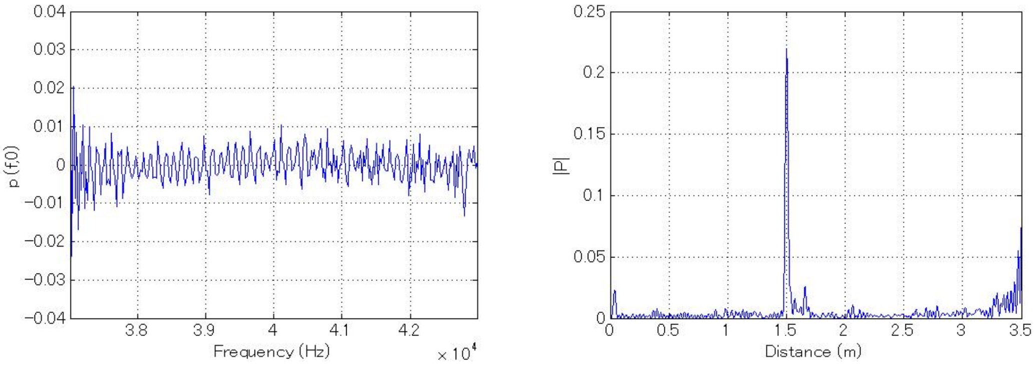


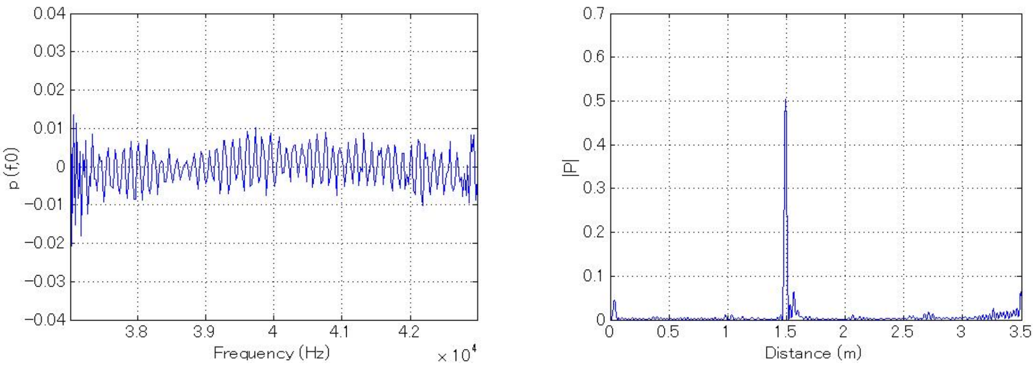
Figure 5: Transmitted wave.



(a) Average of power spectra.

(b) Range spectrum.

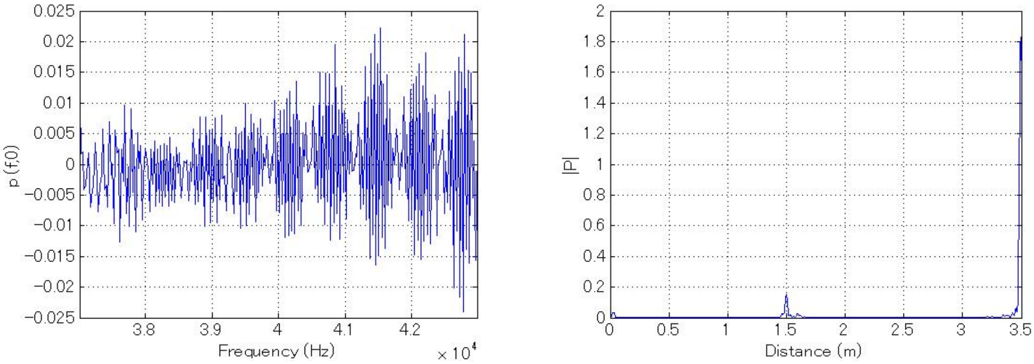
Figure 6: Experimetntal results for the 1st set of observations.



(a) Average of power spectra.

(b) Range spectrum.

Figure 7: Experimetntal results for the 2nd set of observations.



(a) Average of power spectra.

(b) Range spectrum.

Figure 8: Experimetntal results for the 3rd set of observations.

4. Concluding remark

In this study, we measured the distance to a person, especially using the acoustic distance measurement method based on phase interference with parametric array loudspeakers. Since only small reflected waves are recorded at the receiver, estimation was made by introducing synchronous addition to the power spectra. As a result, it was possible to estimate the position of the person, though not perfect. In the future, we will consider methods to improve accuracy, and examine the measurement under various conditions and in cases where there are two or more targets.

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