

A TWO-CHANNEL ADAPTIVE MICROPHONE ARRAY WITH TARGET TRACKING

Y. Nagata and H. Tsuboi

Kansai Research Laboratories, Toshiba Corporation
8-6-26 Motoyama-Minami-cho, Higashi-Nada-ku, Kobe 658, Japan
e-mail: nagata@krl.toshiba.co.jp

ABSTRACT

This paper proposes a new robust adaptive beamformer suitable for a two-channel microphone array. The proposed beamformer is a combination of two Griffith-Jim generalized sidelobe cancellers (GSC) in which the look direction of each GSC is determined by the other GSC's directional response. Moreover, to reduce the degradation of interference suppression performance caused by spatial aliasing, a new arrangement of directional microphones is proposed. The arrangement is simple and effectively reduces the degradation. Simulations demonstrate the effectiveness of the proposed two-channel two-beamformer microphone array.

1. INTRODUCTION

An adaptive microphone array is a promising technique for reducing interference using a small number of microphones. A two-channel array is particularly attractive because stereo audio input is standard for personal computers, which are used for many audio input applications such as speech recognition, computer telephony, and so on.

However, a simple adaptive microphone array using GSC [1] or Frost's beamformer (FBF) [2] suffers from target-signal cancellation caused mainly by look-direction error arising from the movements of the target speaker. Several techniques have been proposed to avoid signal cancellation due to look-direction error.

Correlation-based inhibition (CBI) [3] is a simple method that makes the array insensitive to look-direction error and is applicable to arrays with any number of channels, but the convergence of the adaptive filter becomes slow when the specification allows a large look-direction error. On the other hand, spatial filters with derivative constraints or multiple directional constraints [4,5] do not affect the convergence speed, but these

techniques require a sufficient number of input channels to allow large look-direction error. The method using constrained adaptive filters as a blocking matrix [6] can allow large look-direction error with a small number of channels, but it is difficult to apply this method to a two-channel array because the fixed beamformer output used as a pseudo-target signal in this method may cause degradation in the case of a two-channel array.

Target tracking [7] also requires a sufficient number of channels to estimate the initial direction of arrival (DOA), and it is difficult to estimate the DOA using a two-channel array if interference is present. However, this technique can reduce large look-direction error with a small number of channels if the target DOA is available. To exploit this advantage, we used another DOA estimation technique for a two-channel array. In addition to the problem of target-signal cancellation, we should anticipate the problem of spatial aliasing in the case of a two-channel array. Conventionally, this problem is avoided by setting the intermicrophone distance a value less than the wavelength of the highest frequency of interest.

However, a small intermicrophone distance results in lower resolution in target DOA estimation. To reduce the degradation caused by spatial aliasing, we propose a new arrangement using directional microphones.

2. TARGET TRACKING USING TWO BEAMFORMERS

To estimate the DOAs of two signal sources using a two-channel array, we propose a structure using two beamformers as shown in Fig. 1. One of the two beamformers is the main beamformer, which suppresses interference and extracts the target signal. The other is a sub-beamformer, which suppresses the target signal arriving from the look direction. The main beamformer is a 2-ch GSC with CBI and the sub-beamformer is one without CBI. The structure of the 2-ch GSC is shown in

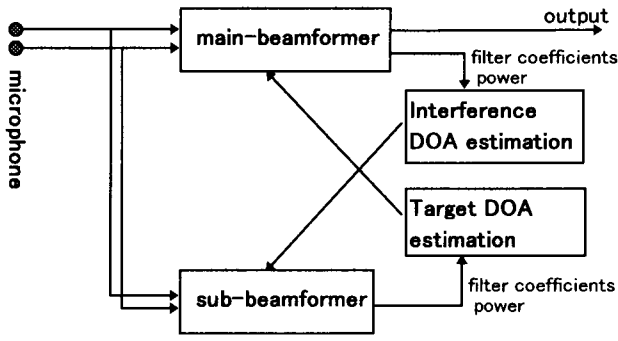


Figure 1. Proposed beamformer

Fig. 2.

Since the directional response of the main beamformer is constrained to have unity gain in the look direction, the directional response shows a minimum value at the interference DOA. Therefore, the interference DOA is estimated by searching the minimum value of the directional response of the main beamformer. The target DOA is estimated similarly from the directional response of the sub-beamformer. The main beamformer and the sub-beamformer are steered to the estimated DOAs of the target and of the interference, respectively. It is possible to track the target by performing the above process continuously.

3. MICROPHONE ARRANGEMENT TO REDUCE SPATIAL ALIASING

Spatial aliasing occurs when the subtraction process in GSC eliminates not only the target signal but also the components of the interference signal that coincide in phase between the channels. As shown in Fig. 2, the coincidence occurs when $2\pi\tau f = 2\pi n$, where τ is the propagation time difference, f is frequency, and n is an integer. Because the adaptive filter employs the subtracted signal as a reference signal, the lack of frequency components in the reference signal results in less reduction of interference at the frequency.

To reduce the elimination of the interference signal, directional microphones are selected and arranged as shown in Fig. 3. In Fig. 3, the angle between the look direction and the front direction of the microphone is

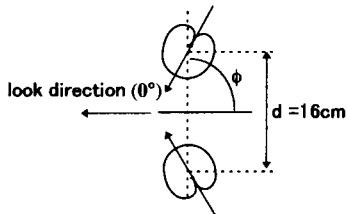


Figure 3. Arrangement of the microphone array

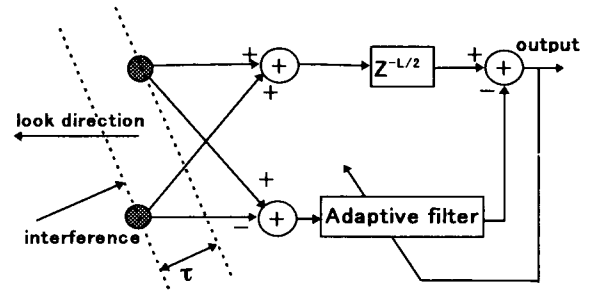


Figure 2. 2-ch Griffith-Jim generalized sidelobe canceller

denoted as ϕ . If the angle ϕ is not equal to 0° , the amplitude of the interference signal in each channel differs from the other, while the amplitude of the target signal is the same.

4. SIMULATIONS

To evaluate the performance of the proposed technique, computer simulations were performed. Throughout the simulations, we assume that the microphones are hypercardioid with an intermicrophone distance of 16 cm as shown in Fig. 3, the filter length of the beamformers is 50, and the sampling frequency is 11 kHz.

4.1 Reduction of Spatial Aliasing

In the first simulation, we calculated the frequency response of the beamformer at the interference DOA to investigate the relationship between spatial aliasing and microphone arrangement. Figure 4 shows the results obtained by changing the angle ϕ to 0° , 30° , 60° , and 90° for the case of single white noise arriving at 50° . As shown in Fig. 4, although no interference reduction is achieved at frequencies near 2.5 kHz and 5 kHz when $\phi = 0^\circ$, the reduction becomes larger as ϕ increases.

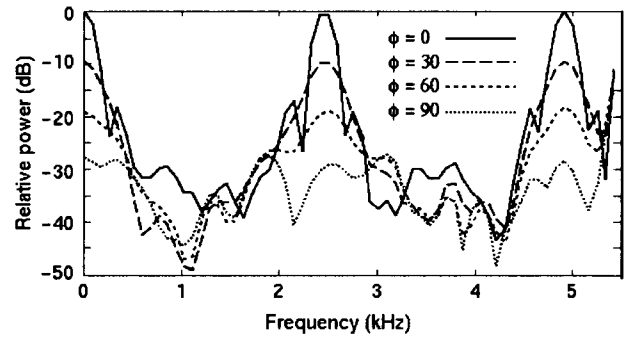


Figure 4. Frequency response of the beamformer at interference DOA.

4.2 Sensitivity After Convergence

In the second simulation, we investigated the sensitivity after convergence vs. signal DOA, assuming a single

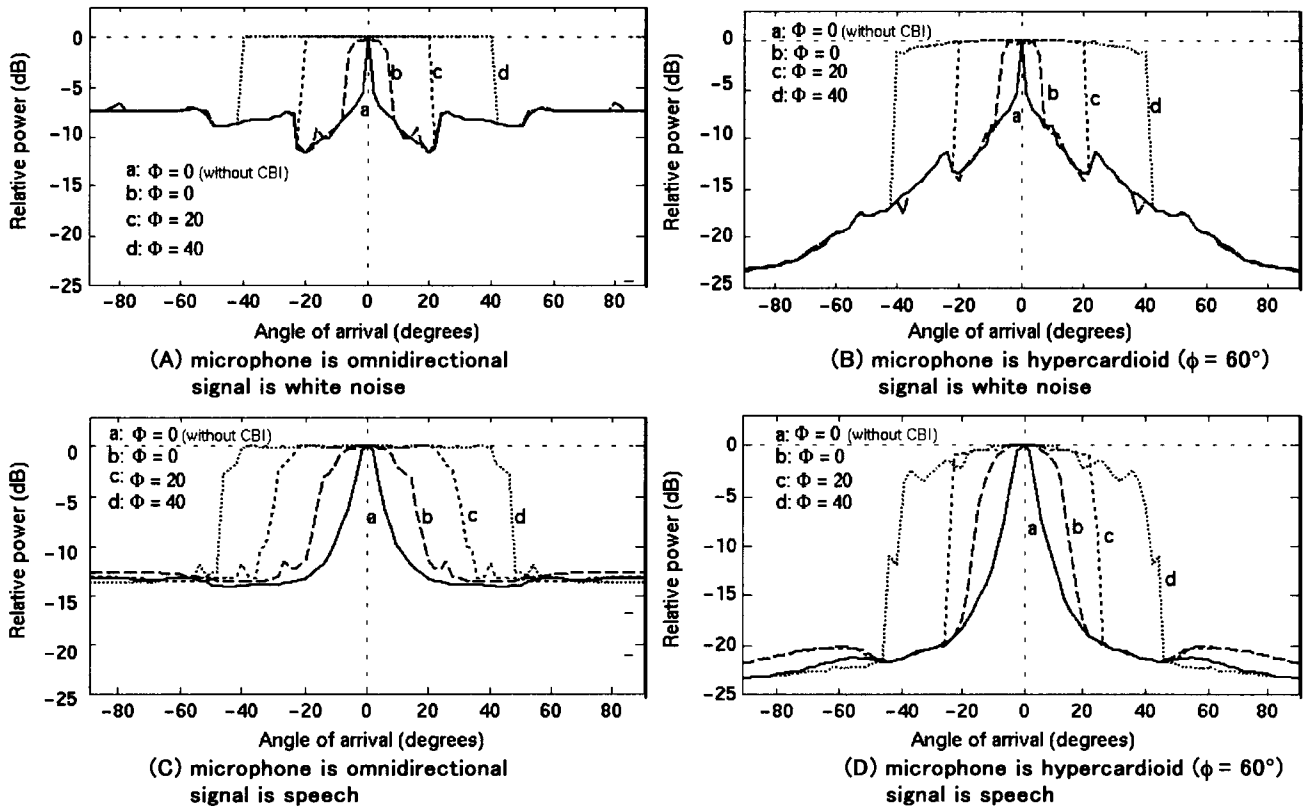


Figure 5. Output power after convergence vs. DOA

- a: without CBI, without tracking
- b: with CBI, without tracking
- c: with CBI, with tracking $\Phi = \pm 20^\circ$
- d: with CBI, with tracking $\Phi = \pm 40^\circ$

signal arrives. We used white noise (0-5.5 kHz) and speech as source signals, and added random noise to each channel. The signal-to-noise ratio (SNR) is 25 dB vs. random noise. The microphone arrangement is $\phi = 60^\circ$ (Fig. 3).

The output power of the main beamformer is summed through 200,000 iterations and plotted at each DOA. We set the angular range for tracking the target to $\pm 0^\circ$, $\pm 20^\circ$, and $\pm 40^\circ$. The range of tracking is denoted by Φ .

Results are shown in Fig. 5(A),(B) for the case of white noise and in Fig. 5(C),(D) for speech. In Fig. 5, (A) and (C) are the results obtained with omnidirectional microphones, and (B) and (D) are those for hypercardioid microphones. We observe that the sensitivity is constant over the tracking range in all cases, and interference reduction performance obtained using directional microphones ((B),(D)) is at least 10 dB higher than that obtained using omnidirectional microphones ((A),(C)).

4.3 Target Tracking

In the third simulation, we investigated tracking

properties assuming that the target is moving. The locations of the target, interference, and microphones are shown in Fig. 6. We used a speech signal of isolated word utterances as the target signal and a speech signal of sentence utterances as the interference signal.

The direction θ of the target source was determined by the expression

$$\theta = D \sin(k \cdot v),$$

where D is the maximum displacement, v is the angular velocity, and k is the number of the data segment. The data segment is defined as three words of speech.

We set D to 20° , v to 30° , target and interference SNR values to 25 dB, and angle ϕ to 60° . Figure 7(A) shows the resulting tracked angle of the target source, and Fig.

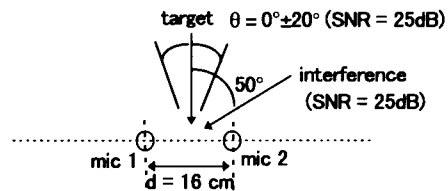


Fig. 6 Microphones and signal DOA locations

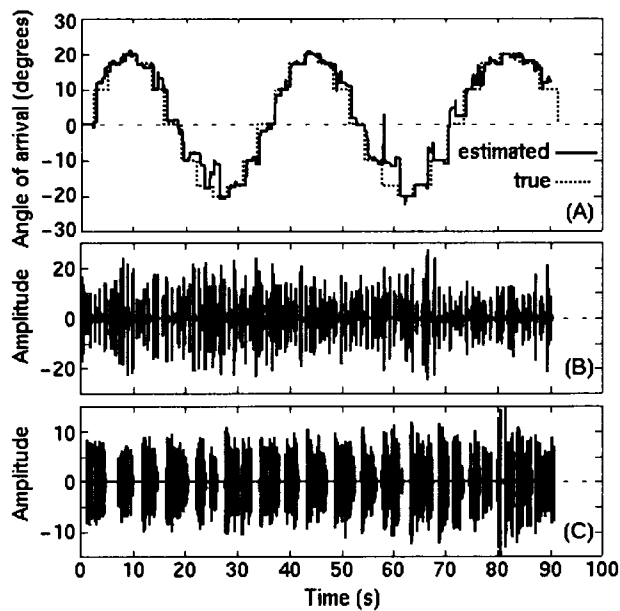


Figure 7. Results of target tracking
(A) track of the target (SINR = -1.4dB)
(B) target signal
(C) interference signal

7(B) and (C) show the target and interference signals, respectively. The signal-to-interference-and-noise ratio (SINR) is -1.4 dB. We observe that the tracking error is sufficiently small.

4.4 Performance of Target Signal Extraction

In this simulation, we investigated the performance of target signal extraction, changing the angle ϕ from 0° to 90° for the case of a moving target. Figure 8 shows the results for target SNR vs. ϕ . In Fig. 8, curve A is obtained for the case where the tracking range $\phi = 0 \pm 30^\circ$, curve B is obtained for the case without tracking, and curves C and D are the SNR values of each microphone output. While the SNR obtained without tracking is lower than 5 dB, the SNR with tracking is higher than 9 dB. The highest SNR of 14.7 dB is obtained with target tracking at $\phi = 60^\circ$.

5. CONCLUSION

A two-channel adaptive microphone array with target tracking using two-beamformers and a new arrangement of directional microphones is proposed for robust interference reduction. Simulation results show that the proposed microphone arrangement using hypercardioid

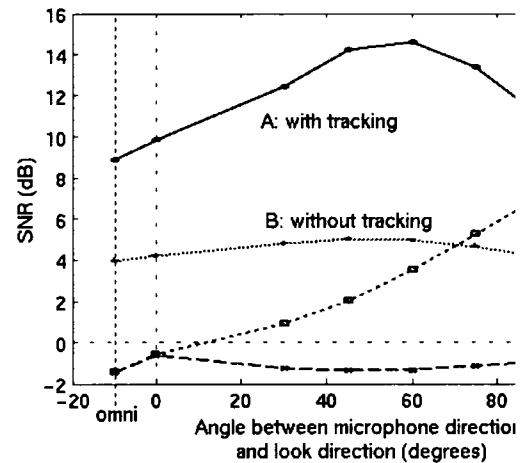


Figure 8. Performance of target signal extr

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