A Modified Robust Adaptive Beamformer for Microphone Arrays

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Abstract

The conventional GSC is inappropriate in real situation when the target signal is present. The steering vector error cancels the target signal and the target signal misadjusts the weight of the adaptive filter. To prevent the target signal cancellation, the robust GSC using the constrained adaptive filters was already proposed. However, the adaptive weight misadjustment is not settled in robust GSC. This paper proposes a revised robust sidelobe canceller with adaptive compensator. To compensate the influence of target signal, the adaptive compensator is used in cascade. In computer simulation, we show the performance improvement by comparing the robust GSC with the proposed GSC.

1. Introduction

Microphone arrays have been widely used for teleconferencing, speech recognition, speech enhancement, and hearing aids. The microphone array can reduce the interference signal to achieve the higher output signal-to-noise ratio(SNR). The generalized sidelobe canceler(GSC) uses the adaptive beamforming to yield the target signal more clearly[1-3].

However, the classical adaptive beamformer based on GSC cannot be directly used in real situation because the target signal cancellation and weight misadjustment.

The target signal cancellation occurs in the presence of steering vector errors. The steering vector error caused by errors in microphone positions, microphone gains, reverberation, and target direction. Several approaches to inhibit target-signal cancellation have been proposed[4-6]. Hoshuyama et al., proposed the robust adaptive beamformer with an adaptive blocking matrix using CCAF's(coefficients-constrained adaptive filter) and a multiple input canceller using NCAF's(norm-constrained adaptive filter)[7].

The adaptive weight misadjustment occurs in the presence of target signal. The GSC use the output signal to update the adaptive filter coefficients. The output signal of GSC estimates the target speech signal. Accordingly, when the adaptive filter converges to optimal value, the fluctuation of the target signal makes the filter coefficients misconverge to the optimal weight.

In this paper, we propose the new adaptive beamformer to apply in real situation. The proposed beamformer use the Hoshuyama et al.,'s GSC structure to prevent the target signal cancellation and the adaptive compensator in cascade to reduce the adaptive filter weight misadjustment. The linear predictor is used as the adaptive compensator. The performance of proposed adaptive beamformer is demonstrated by computer simulation.

2. Conventional GSC

Fig. 1 shows the block diagram of robust GSC proposed by Hoshuyama et al.

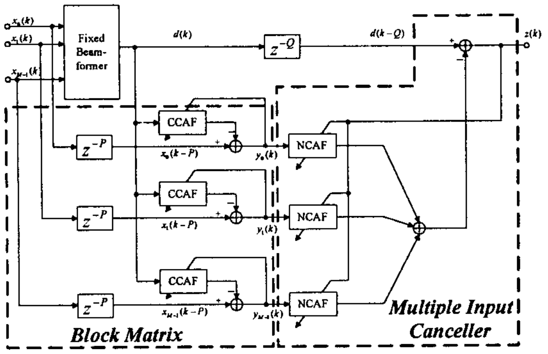

Fig. 1. The block diagram of robust GSC

The robust beamformer uses a new blocking matrix(BM) with CCAF's and an multiple input canceller(MC) with NCAF's. The CCAF's in the BM minimizes the output signals of the BM using the reference signal from the FBF. Because of the constraints in the CCAF's, this leads to minimization of the target-signal leakage at the BM output. The NCAF's in the MC prevent the target signal cancellation when the target signal minimization at the BM is incomplete, and there is some residual target signal at the outputs of BM.

A. Signal Processing in BM

$$y_m(k) = x_m(k-P) - H_m^T(k) D(k) \tag{1}$$

$$H_m(k) \equiv [h_{m,0}(k), h_{m,1}(k), \ldots, h_{m,N-1}(k)]^T \tag{2}$$

$$D(k) \equiv [d(k), d(k-1), \ldots, d(k-N+1)]^T \quad (m=0,1,\ldots,M-1) \tag{3}$$

The signal $y_m(k)$ is the mth output of the BM, $x_m(k)$ is the mth microphone signal, and P is the number of delay samples for causality. $H_m(k)$ is the coefficient vector of the mth CCAF, and $D(k)$ is the signal vector consisting of delayed signals of $d(k)$. The CCAF coefficients $h_m(k)$ are adapted with coefficients constraints. For adaptation, the NLMS algorithm is used.

$$h'_{m,n} = h_{m,n}(k) + \alpha \frac{y_m(k)}{\|D(k)\|} d(k-n) \tag{4}$$

$$h_{m,n}(k+1) = \begin{cases} \phi_{m,n}, & \text{for } h'_{m,n} > \phi_{m,n} \\ \psi_{m,n}, & \text{for } h'_{m,n} < \psi_{m,n} \\ h'_{m,n}, & \text{otherwise} \end{cases} \quad (m=0,1,\ldots,M-1), (n=0,1,\ldots,N-1) \tag{5}$$

where $\|\cdot\|$ denotes the Euclid norm. The terms $h'_{m,n}$ are temporal coefficients for limiting functions, α is the step size, and $\phi_{m,n}$ and $\psi_{m,n}$ are the upper and lower limits for each coefficients.

B. Signal Processing in MC

$$z(k) = d(k-Q) - \sum_{m=0}^{M-1} W_m^T(k) Y_m(k) \tag{6}$$

$$W_m(k) \equiv [w_{m,0}(k), w_{m,1}(k), \ldots, w_{m,L-1}(k)]^T \tag{7}$$

$$Y_m(k) \equiv [y_m(k), y_m(k-1), \ldots, y_m(k-L+1)]^T \quad (m=0,1,\ldots,M-1) \tag{8}$$

The coefficients of the NCAF's are updated by an adaptive algorithm a norm constraints.

$$W'_m(k) = W_m(k) + \beta \frac{z(k)}{\sum_{j=0}^{N-1} \|Y_j(k)\|^2} Y_m(k) \tag{9}$$

$$\Omega = \sum_{m=0}^{M-1} \|W'_m\|^2 \tag{10}$$

$$W_m(k+1) = \begin{cases} \sqrt{\frac{K}{\Omega}}\, W'_m, & \text{for } \Omega > K \\ W'_m, & \text{otherwise} \end{cases} \quad (m=0,1,\ldots,M-1) \tag{11}$$

where β is the step size, W'_m is the temporal vector for the constraints, and Ω and K are total squared-norm of $W_m(k)$ and a threshold.

The structure of Hoshuyama et al.,'s GSC shows the better performance in steering vector error than the conventional GSC. However, the misadjustment of adaptive coefficients is not considered. In the proposed algorithm, the adaptive compensator is used to reduce the weight misadjustment.

3. Proposed GSC

The structure of the proposed GSC is shown in Fig. 2.

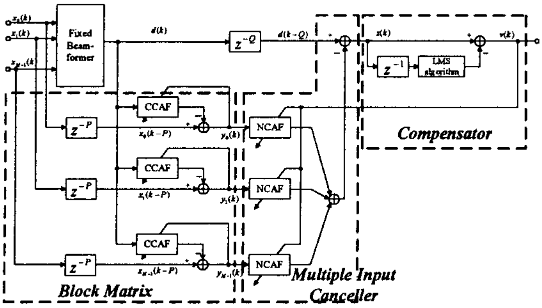

Fig. 2. A robust GSC with a adaptive compensator

A linear predictor is used as the compensator to prevent the weight misadjustment by the target signal. The adaptive compensator is cascade-connected to the GSC, then the output of compensator is used as an estimated error signal which is converged to zero, so the fluctuation of the estimated error signal is reduced.

$$v(k) = z(k) - z_c(k) = z(k) - W_c^T(k)Z(k-1) \quad (12)$$

$$W_c(k+1) = W_c(k) + \gamma v(k)Z(k-1) \quad (13)$$

The $v(k)$ is the compensator output signal, $W_c(k)$ is a linear predictor coefficient vector, $Z(k-1)$ is the delayed output vector of GSC, and γ is the step-size. If the adaptive compensator estimates well the target signal, $v(k)$ will only contain the component of interference signal. Then, the NCAF's in MC minimizes the interference signal in $z(k)$. The final output of this structure is the output signal of GSC, $z(k)$.

The total output powers after convergence normalized by the power of the assumed target direction are plotted in Fig. 3. The bandlimited(0.3-3.7 ㎑) Gaussian signal is used. The maximum allowable target direction is set to 20° and the number of microphone is set to 4 with an equal spacing 4.1 ㎝. First, the CCAF's are adapted for 50,000 iterations, and then, the NCAF's are adapted for 150,000 iterations. The number of coefficients of the CCAF's and NCAF's is 16. The parameters are $P=4, Q=7, K=10, \alpha=0.1, \beta=0.2,$ and $\gamma=0.1$.

In Fig. 3, the robust GSC and the proposed adaptive beamformer accepts the signal incident within 0° and 20° . In robust GSC, The signal incident above 20° is attenuated over 25 dB. Because of the influence of the compensator, the interference attenuation is degraded in the proposed GSC. However, the proposed GSC shows the interference reduction performance over 15 dB.

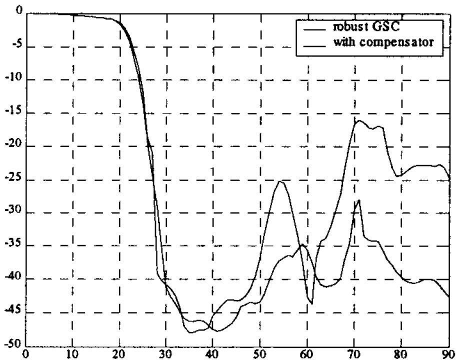

Fig. 3. Normalized output power after convergence as a function of DOA

4. Computer Simulation

The simulations using a real speech signal and bandlimited signal is performed. For target signal, the male speech in Korean is used and for interference signal, the bandlimited (0.3-3.7㎑) Gaussian signal is used. It is assumed that four microphones are located with equal spacing 4.1 ㎝. The maximum allowable target direction is set to 20° . The target DOA is set to 0° and the interference DOA is set to 45° . The number of coefficients of all the CCAF's, NCAF's and compensator are 16. The parameters are $P=4, Q=7, K=10, \alpha=0.02, \beta=0.004,$ and $\gamma=0.05$. The sampling frequecy is 8 ㎑.

In Fig. 4. (a) is the original target speech signal, (b) is the output of robust GSC, and (c) is the output of the proposed GSC. In this figure, the output of proposed GSC is more similar to the original signal than the output of robust GSC when the voice is active. To illustrate the effect of the compensator, the output power is shown in Fig. 5. To get the output power, the following first-order difference equation is used.

$$P(k) = 0.99P(k-1) + 0.01|z(k)|^2 \qquad (14)$$

In Fig. 5, (a) is the original target speech signal, (b) is the output of robust GSC, and (c) is the output of the proposed GSC. When the voice is present, the proposed structure shows the performance improvement approximately 3 dB and the output of proposed structure coincide with the original signal because of the compensator.

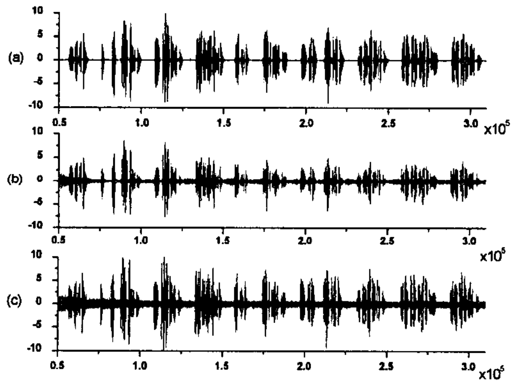

Fig. 4. Ouput signal for a male speech and a white noise

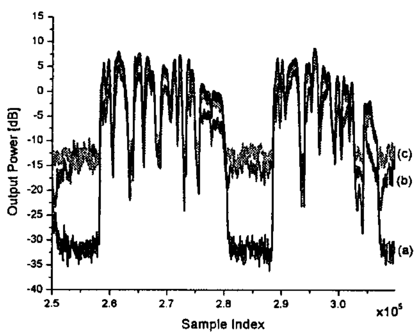

Fig. 5. Ouput powers for a male speech and a white noise

5. Conclusion

In this paper, the modified robust adaptive beamformer is proposed. The proposed beamformer shows the better performance than the conventional robust adaptive beamformer about 3 dB. Using the conventional robust adaptive beamformer, the signal power decrease to the half power. But, the proposed adaptive beamformer compensates the power attenuation. Therefore, we can obtain the output signal close to the original speech signal.

For the further research, the compensator applicable to the colored noise is necessary and the experiment using the multi-DSP board is also needed.

[References]

[1] L. J. Griffiths and C. W. Jim, "An alternative approach to linearly constrained adaptive beamforming," IEEE Trans. Antennas Propaget., vol. AP-30, pp. 27-34, Jan. 1982.

[2] P. M. Clarkson, Optimal and adaptive signal processing. CRC Press, 1993.

[3] H. L. Van Trees, Optimum Array Processing. John Wiley & Sons, Inc., 2002.

[4] I. Claesson and S. Nordholm, "A spatial filtering approach to robust adaptive beamforming," IEEE Trans. Antennas Propagat., vol. 40, pp. 1093-1096, Sept. 1992.

[5] N. K. Jablon, "Adaptive beamforming with the generalized sidelobe canceller in the presence of array imperfections," IEEE Trans. Antennas Propaget., vol. AP-34, pp. 996-1012, Aug. 1986.

[6] H. Cox, R. M. Zeskind, and M. M. Owen, "Robust adaptive beamforming," IEEE Trans. Acoust., Speech, Signal Processing, vol. ASSP-35, pp. 1365-1376, Oct. 1987.

[7] O. Hoshuyama, A. Sugiyama, A. Hirano, "A robust beamformer for microphone arrays with a blocking matrix using constrained adaptive filters," IEEE Trans. Signal Processing, vol. 47, pp. 2677-2684, Oct. 1999.