Modification of Generalized Side-lobe Canceller with an Adaptive Filter and Compensator

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Abstract

This paper proposes a modified generalized side-lobe canceller with a summed adaptive filter and an adaptive compensator. A summed adaptive filter reduces computational loads and the adaptive compensator minimizes the misadjustment of the adaptive filter coefficients. Computer simulations explain the performance improvement of the proposed method and the conventional generalized side-lobe canceller.

I. Introduction

Since an adaptive beamforming algorithm was proposed in the 1970's ,researches and applications in this area have attracted till now[1]. Frost et al., proposed the idea of the side-lobe canceller[2]. The Griffiths and Jim proposed a generalized side-lobe canceller(GSC) in the 1980s[3]. Possible GSC applications include hearing aids[4], speech recognition enhancement for hands-free telephones[5 ,6], sonar, communication of the underwater and high frequency communications[7]. In many adaptive filter applications, the normalized least-mean-square(NLMS) algorithm has been used popularly since it is simple and has an asymptotic convergence behavior. However, the fluctuating output causes to misadjust the filter coefficients when was adapted to the optimal value[8].

This paper proposes a modified GSC with an adaptive filter and a compensator. Each of the output of block matrix is summed and one adaptive filter is used. It reduces the number of computational loads because of the using only one adaptive filter. A linear predictor is used as a compensator and the compensator is cascade-connected to the GSC using an adaptive filter. The estimated error whitened by this adaptive compensator will converge to zero and then the adaptive filter coefficients can be adjusted adaptively to the optimal values[4].

II. Generalized Side-lobe Canceller

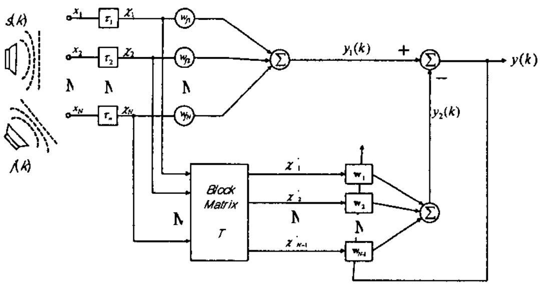

Fig. 1. Generalized side-lobe canceller structure of linearly constrained adaptive processing algorithm

This system studied here is based on the adaptive beamformer described by Griffiths and Jim[3]. The block matrix T is set up such that it prevents a target signal from received signal.

Each of the adaptive filters update using NLMS algorithm such that is to be minimized the power of the beamformer output:

$$y(k) = y_1(k) - y_2(k)$$

$$= s(k) + j_1(k) - W^T j_2(k) \qquad (1)$$

where $s(k)$ is the target signal, $j_1(k)$ is the upper channel jammer signal, and the lower channel jammer signal is $j_2(k) = [\, \boldsymbol{x}'^T_1(k),\ \boldsymbol{x}'^T_2(k), \ldots, \boldsymbol{x}'^T_{N-1}(k)]^T$. The $\boldsymbol{W}$ is the adaptive filter weight vector. The input signal vector to each adaptive filter are $\boldsymbol{x}'^T_n = [x'_n(k), x'_n(k-1), \ldots, x'_n(k-L+1)]^T$.

The adaptive filter weights, $\boldsymbol{W} = [\boldsymbol{w}_0\ \boldsymbol{w}_1, \ldots, \boldsymbol{w}_n]$, are adjusted to minimize the total output power, which, under ideal conditions, preserves the target signal and minimizes the jammer output[3].

If perfect adaptation occurs, only target signal remains at the beamformer output $y(k)$.

III. Proposed Generalized Side-lobe Canceller

The conventional GSC uses the $N-1$ adaptive filters and total multiplication operation in conventional GSC is $L \times (N-1)$ where the L is the each adaptive filter length. Besides the fluctuating output of GSC causes the filter coefficients to misadjust in adapting to the optimal value[8].

So we propose a modified GSC. Proposed GSC consists of an adaptive filter and a compensator. The compensator uses a forward linear predictor and it is cascade-connected to the GSC using an adaptive filter. An adaptive filter coefficients are updated NLMS algorithm, that algorithm uses the output signal as an estimated error signal. The estimated error whitened by this adaptive compensator will converge to zero and then the adaptive filter coefficients can be adjusted adaptively to the optimal values[8].

Proposed structure shown in Fig. 2. Just an adaptive filter is used after the block matrix. The adaptive compensator makes up for the reduced

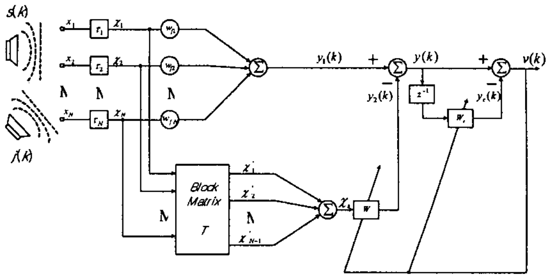

Fig. 2. Structure of proposed side-lobe canceller with an adaptive compensator

performance and minimizes the misadjustment of the adaptive filter coefficients.

The output of proposed structure is used as the estimated error signal for the adaptive filter. So the updated NLMS equation of the proposed GSC is the following equation(1).

$$\begin{aligned} &\boldsymbol{W}_c(k+1) \\ &= \boldsymbol{W}_c(k) + \frac{\alpha(\, y(k) - y_c(k))(\, \sum_{i=1}^{n} \boldsymbol{x}_i')}{((\, \sum_{i=1}^{n} \boldsymbol{x}_i')^T (\, \sum_{i=1}^{n} \boldsymbol{x}_i') + c)} \\ &= \boldsymbol{W}_c(k) + \frac{\alpha(\, y_1(k) - y_2(k) - y_c(k))(\, \sum_{i=1}^{n} \boldsymbol{x}_i')}{((\, \sum_{i=1}^{n} \boldsymbol{x}_i')^T (\, \sum_{i=1}^{n} \boldsymbol{x}_i') + c)} \\ &= \boldsymbol{W}_c(k) + \frac{\alpha v(k)(\, \sum_{i=1}^{n} \boldsymbol{x}_i')}{((\, \sum_{i=1}^{n} \boldsymbol{x}_i')^T (\, \sum_{i=1}^{n} \boldsymbol{x}_i') + c)} \qquad (2) \end{aligned}$$

The bold letter represents vector notation and the letter T represents a transpose. The compensator output $y_c(k)$ can be divided by the target signal component $s_c(k)$ and the jammer signal component $j_c(k)$. Then the error signal in compensator $v(k)$ can be interpreted the following equation(3).

$$\begin{aligned} v(k) &= y_1(k) - y_2(k) - y_c(k) \\ &= s(k) + j_1(k) - y_2(k) - s_c(k) - j_c(k) \\ &= [\, (s(k) - s_c(k)) + (j_1(k) - y_2(k) - j_c(k))] \qquad (3) \end{aligned}$$

where $s_c(k)$ is the involved target signal component of compensator output, $j_c(k)$ is the involved jammer signal component of compensator output and $v(k)$ is the estimated error of proposed structure.

In the equation(3), the target signal effect was minimized by this term $s(k)-s_c(k)$ and the estimated error whitened by cascade-connected adaptive compensator behavior will finally converge to zero.

IV. Simulation Result

In order to demonstrate the performance of the proposed GSC beamformer which consists of four sensor. We assumed that the propagation velocity of the signal in air environment is 340 m/s and the inter-sensor distance is 0.04 m to avoid a grating lobe effect[1]. The weights of conventional delay-sum beamformer are all one . The block matrix T was used :

$$T=\begin{bmatrix} 1 & -1 & 0 & 0 \\ 0 & 1 & -1 & 0 \\ 0 & 0 & 1 & -1 \end{bmatrix} \tag{2}$$

The sampling rate is $8KHz$ and the number of adaptive and compensator filter length is 16. The step size of GSC is $\alpha=0.1$ and step size of compensator is $\alpha'=0.01$.

We will test several cases. The first case is that the target signal $s(k)$ is a mono-chromatic wave signal with a $0.085m$ wave length and was received from broad-side. The jammer signal $j(k)$ is a narrow-band random signal which has a frequency band from $1000Hz$ to $2000Hz$. And $j(k)$'s incidence degree is assumed from 45^o direction. The SNR of the whole simulation is 3dB. Fig. 3 illustrates the learning curves between the tracking signal and the target signal. The curves are averaged over 100 times simulations.

The second case is that the target signal $s(k)$ is narrow-band random signal which has a frequency band from $1500Hz$ to $2500Hz$. The jammer signal $j(k)$ is a narrow-band random signal which has a frequency band from $500Hz$ to $1500Hz$. And $j(k)$'s incidence degree is assumed from 45^o direction.

Namely target signal frequency band and jammer signal frequency band are different in this case.

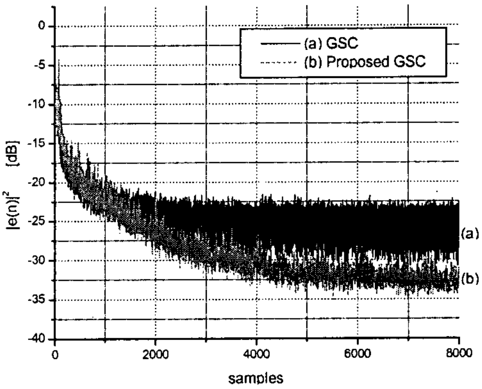

Fig. 3. Learning curves of error signals with a mono-chromatic sinusoidal signal as a target signal and with a narrow-band random signal as a jammer

In Fig. 4, the proposed system has better performance than conventional GSC about 10dB in output error.

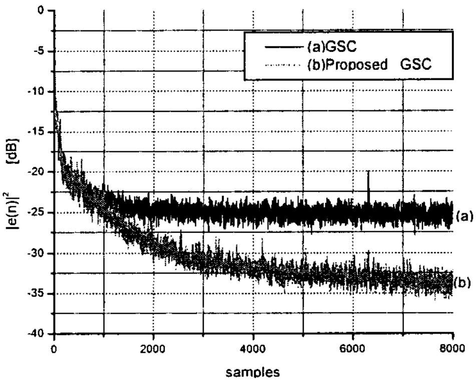

Fig. 4. Learning curves of error signals with a narrow-band random signal as a target signal (1500Hz~2500Hz) and a jammer(500Hz~1500Hz)

The Third case is that the target signal $s(k)$ is narrow-band random signal which has a frequency band from $1500Hz$ to $2500Hz$. The jammer signal $j(k)$ is a narrow-band random signal which has a frequency band from $2000Hz$ to $3000Hz$. And $j(k)$'s

incidence degree is assumed from 45° direction.
Namely target signal frequency band and jammer signal frequency band was overlapped in this case.

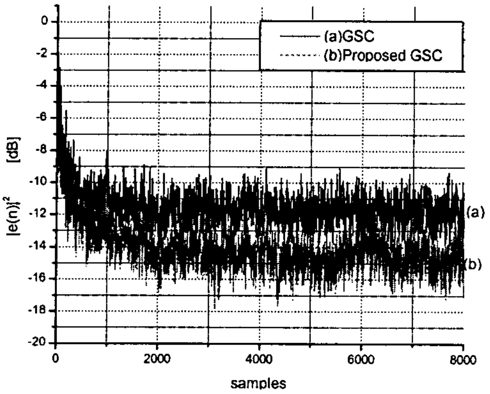

Fig. 5. Learning curves of error signals with a narrow-band random signal as a target signal (1500Hz~2500Hz) and a jammer(2000Hz~3000Hz)

In Fig. 5, the proposed system has better performance than GSC about 2dB in output error.
Due to the adaptive compensator, the convergence speed is slow in the proposed adaptive beamformer. However, the proposed structure shows the improvement in steady-state performance.

V. Conclusion

In this paper is proposed a modified GSC which is compared with conventional GSC. The proposed system decreased in quantity of adaptive filters, although has better performance than conventional GSC in output error because of the compensator. Accordingly the proposed GSC will be using hearing aids, speech recognition enhancement for hands-free telephones, sonar, underwater communication and high frequency communications.
Further research will concentrate on developing of the applications in a real-world environment.

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