

# A NEW ADAPTIVE BLOCKING MATRIX WITH EXACT FIR STRUCTURE FOR ROBUST GENERALIZED SIDELOBE CANCELLER

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## ABSTRACT

To improve the robustness of the generalized sidelobe canceller (GSC) against steering error and inevitable misdetection of adaptation mode control (AMC) in real applications, the conventional adaptive blocking matrix (ABM) consists of coefficient-constrained adaptive filters (CCAFs) which use the output of delay-and-sum fixed beamformer (FBF) as input, in order to limit the tracking range of the ABM. Since the output of FBF is a FIR-like response for directional signal, the attempt of inverse filtering with a FIR-based CCAF in the conventional ABM will result in irregular spread and oscillation of coefficients. As results, the actual tracking range varies with the signal frequency, and the rejection of low frequency interference will be deteriorated in the case of misdetection of AMC. This paper proposes a new ABM with exact FIR structure, in which the output of FBF is used as desired signal, thus the coefficients oscillation caused by inverse filtering is eliminated. Coherent tracking range can be achieved across full frequency band by simply constraining the pulse width of the ABM filter response. Experiment in real environment shows that the loss of interference rejection caused by misdetection of AMC is reduced significantly.

## 1. INTRODUCTIONS

The generalized sidelobe canceller (GSC) is an effective approach to enhance the interference reduction of microphone array beamforming [1]-[4]. An adaptive blocking matrix (ABM) have been proposed by Hoshuyama *et al.* in [5] to take the place of the fixed blocking matrix in Griffiths-Jim's GSC [1] so that the target signal cancellation caused by steering error or slight movement of desired talker can be reduced. The adaptation of the ABM is controlled by an SNR estimation based AMC (adaptation mode controller) in [6]. However, due to the double-talk scenario [7] and the SNR misdetection caused by high correlation of directional interference in low frequency band, the mistracking of target are inevitable in real applications. As a remediation

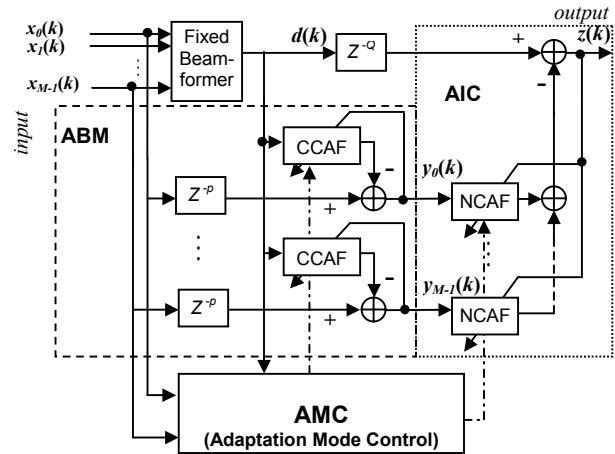


Fig.1 Diagram of GSC with conventional ABM

of the misjudgment of AMC, the coefficients of the ABM filters are constrained in order to leak the interference into the sidelobe canceller. Nevertheless, as the DOA of interference increases, the tap coefficients of the conventional ABM in [5] spread less for low frequency input [8], but spread more and even tend to oscillate at some particular frequencies for high frequency input. As results, the actual tracking range varies with the signal frequency, and the rejection of low frequency interference will be deteriorated in the case of AMC misdetection.

To improve the robustness against double-talk and SNR misdetection in low frequency band, two approaches, a frequency domain GSC with frequency-dependent DFT-bin-wise adaptation control [7] and a subband-GSC in which the ABM and the adaptive noise canceller are processed in multiple subbands with different coefficients constraint vectors [8], have been proposed by W. Herbordt *et al.* and W. H. Neo *et al.* respectively. However, the irregular spread of ABM tap coefficients have not been addressed.

In this paper, the causation of the irregular spread and oscillation of the ABM tap coefficients is analyzed from the point of view of FIR filter structure, and then a new ABM with exact FIR structure is presented to eliminate the irregular coefficients spread and oscillation. Simulations on five testing signals show that coherent

tracking range can be achieved across full frequency band by simply constraining the pulse width of the ABM filter response, and experiment in real environment shows that the loss of interference rejection caused by misdetection of AMC is reduced significantly.

## 2. THE INVERSE FILTERING OF THE CONVENTIONAL ABM

The diagram of the robust GSC with conventional ABM is illustrated in Fig.1. The robust GSC consists of a fixed beamformer (FBF), an ABM for rejecting target signal and leaking interference, an adaptive interference canceller (AIC) for subtracting the signals correlated to the output  $y_m(k)$  of ABM from the output  $d(k)$  of FBF, and an AMC (adaptation mode controller) for stalling the adaptation of ABM and AIC according to signal spatial information. From the point of view of adaptive FIR filter, in this conventional ABM, the steered input signals  $x_m(k)$  are used as the desired signal, while the FBF output  $d(k)$  is used as FIR input.

The issue of inverse filtering arises in the case of directional interference comes from  $\theta \neq 0^\circ$ . For ideal linear equi-spacing array, the relationship of input signals between channel 0 and channel  $k$  follows

$$x_k(t) = x_0 \left( t - k \frac{d \cdot \sin \theta}{c} \right), \quad (1)$$

where  $c$  is the sound speed, and  $d$  is the microphone spacing,  $t$  is the time index. Thus the BF output signal follows

$$d(t) = x_0(t) * h_{FBF}(\theta, t) \quad (2)$$

where  $*$  denotes convolution, and

$$h_{FBF}(\theta, t) = \frac{1}{M} \sum_{k=0}^{M-1} \delta \left( t - k \frac{d \cdot \sin \theta}{c} \right) \quad (3)$$

Therefore in the case of  $\theta \neq 0^\circ$ , the FBF can be regarded as a equivalent lowpass FIR filter  $h_{FBF}(\theta, t)$  with single input  $x_0(t)$  and single output  $d(t)$ . For the conventional ABM, since  $x_k(t)$  are used as the desired signal, and  $d(t)$  is used as FIR input, the essential function of the ABM filters are inverse filtering of the equivalent filter  $h_{FBF}(\theta, t)$ . As we know, the inverse filtering of a FIR response cannot be implemented exactly through a FIR structure. Furthermore, the frequency-domain zero-points of  $h_{FBF}(\theta, t)$  will lead to irregular spread and even oscillation of the tap coefficients of the ABM filters. As in Fig. 2 and Fig. 3, the filter coefficients spread less for low frequency input (cutoff at 500Hz), but spread much more and even tend to oscillate for high frequency input (cutoff at 1kHz). Consequently it is difficult to design the ABM constraint vectors that can effectively control the tracking range at low frequency while blocking high frequency target signal.

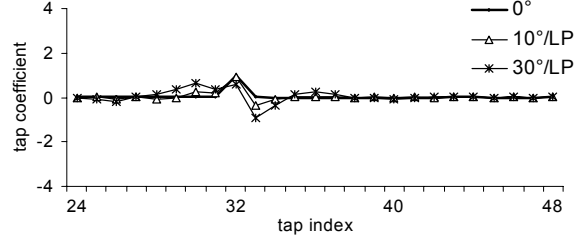


Fig. 2 Coefficients spread of conventional ABM for lowpass Gaussian noise input from  $0^\circ$ ,  $10^\circ$  and  $30^\circ$

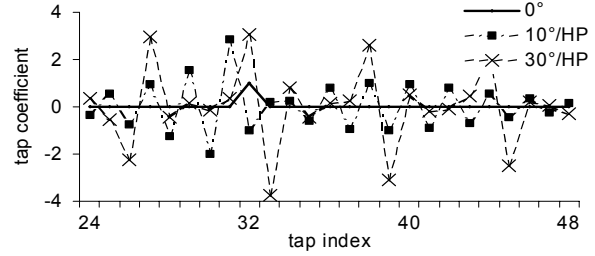


Fig. 3 Coefficients oscillation of conventional ABM for highpass Gaussian noise input from  $0^\circ$ ,  $10^\circ$  and  $30^\circ$

## 3. THE PROPOSED ABM WITH EXACT FIR STRUCTURE

To eliminate the coefficients oscillation, we propose a new ABM with exact FIR structure, in which the input signals are used as FIR input, while the FBF output is the desired signal for adaptation, as in Fig. 4. Compared with the conventional ABM structure, since the roles of  $x_k(t)$  and  $d(t)$  are swapped with each other, the function of the proposed ABM filters is no longer inverse filtering of  $h_{FBF}(\theta, t)$ . Therefore, the ideal time-domain filter response of ABM in channel  $l$  for target from  $\theta$  should be

$$w_{ABM,l}(\theta, t) = \frac{1}{M} \sum_{k=0}^{M-1} \delta \left( t - (k-l) \frac{d \cdot \sin \theta}{c} - \tau_p \right) \quad (4)$$

where  $\tau_p$  is the delay compensation for the ABM filters, which is realized by the delay line  $z^{-p}$  in Fig. 4.

Thus the ideal time-domain filter response is an impulse for  $\theta = 0$ , and equals to the sum of  $M$  sequent impulse for  $\theta \neq 0$ . Furthermore, the width of the total response is

$$\tau_w \approx (M-1) \frac{d \cdot |\sin \theta|}{c} \quad (5)$$

Fig. 5 shows the coefficients spread in the proposed ABM with two testing signals: lowpass Gaussian noise (LP) and highpass Gaussian noise (HP). The results are regular in time domain and less sensitive to signal spectrum.

Instead of constraint on coefficients magnitude as in [5,6], here constraining is performed on the pulse width of filter coefficients in time domain. All the filter coefficients that outside the allowable pulse width  $\tau_w$ , which is

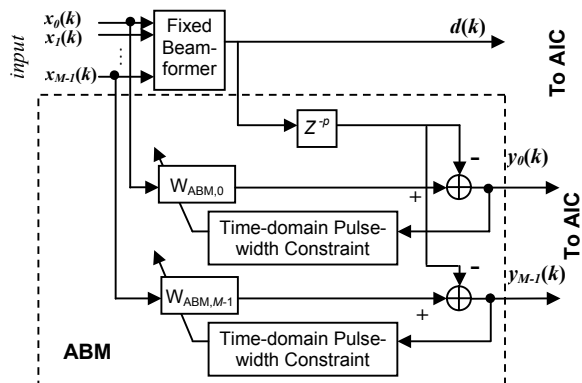


Fig. 4 Diagram of the proposed ABM with Exact FIR structure

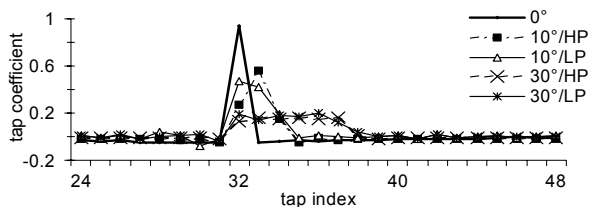


Fig. 5 Coefficients spread of the proposed ABM for lowpass and highpass Gaussian noise input from  $0^\circ$ ,  $10^\circ$  and  $30^\circ$

calculated according to (5), are limited by tolerance thresholds  $\pm\delta_p$ , which are for protection in reverberation environment.

#### 4. SIMULATIONS AND EXPERIMENTS

Simulations were conducted to measure the directional responses of two GSC beamformers with conventional ABM and the proposed ABM under mistracking. Five typical signals with different spectrums were used as input: white Gaussian noise, lowpass Gaussian noise (cutoff at 500Hz), highpass Gaussian noise (cutoff at 1kHz), female voice and male voice from NIST Speech Disc. The microphone array configurations are:  $M=8$ ,  $d=4\text{cm}$ ,  $f_s=12\text{kHz}$ . The allowable target direction error range was designed as  $\pm 15^\circ$ . To simulate the mistracking, the adaptation of both the ABM and the AIC were active for each direction.

Fig. 6 and 7 show the simulation results. Seen from Fig. 6, although the allowable range with Hoshuyama's ABM can be controlled as expected for 4 kinds of testing signals, there is obvious leakage of lowpass Gaussian noise outside the expected range. By using the proposed ABM, near-rectangle directional responses and coherent beam patterns for all testing signals are achieved under mistracking condition, as shown in Fig. 7.

Experiment based on real recordings was also conducted to measure the loss of performance caused by

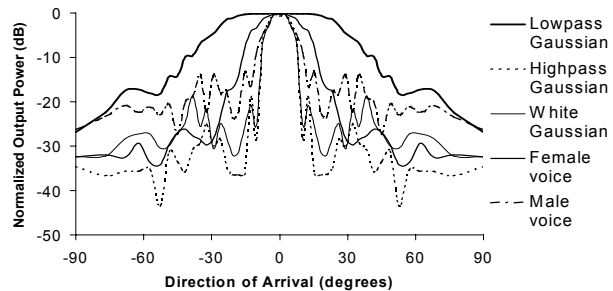


Fig. 6 Directional response of GSC with the conventional ABM in the mistracking simulation

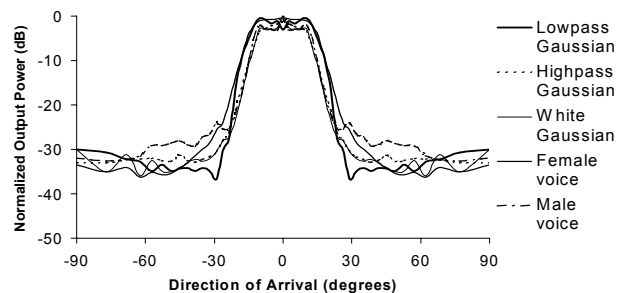


Fig. 7 Directional response of GSC using the proposed ABM with Exact FIR structure in the mistracking simulation

SNR misdetection in a typical office room with reverberation time of about 300ms. The target talker was active during the period of [7.38, 8.54], and there were two interference talkers speaking from  $45^\circ$  and  $-45^\circ$  during the period of [6.5, 7.0], and [9.6, 12.7]. The output power of the two GSC beamformers with the conventional ABM and the proposed ABM were compared to a GSC beamformer with "ideal" AMC using prior knowledge. The parameters of the microphone array and the ABM constraint settings of the two beamformers were the same as those in the simulations. The output powers of (a) single microphone, (b) the "ideal" GSC, (c) the frequency domain GSC with conventional ABM [7], and (d) the GSC with the proposed ABM are drawn in Fig. 8. For the beamformer (b), the adaptation of ABM was active while the adaptation of AIC was frozen during the active period of target talker, and the blocking matrix was fixed to  $0^\circ$  while the adaptation of AIC was active in other periods. For the beamformers (c) and (d), frequency-dependent DFT-bin-wise SNR estimation in [7] was adopted in the AMC.

The target cancellation, the rejection of directional interference, and the rejection of diffuse field noise were measured on the average of corresponding periods of power curve in Fig. 8. They are shown in Table 1. As a result of actual misdetection of AMC in real environment, the interference rejection of the frequency domain GSC with conventional ABM is deteriorated by 6.31dB, but for the GSC enhanced with the proposed ABM, the loss of

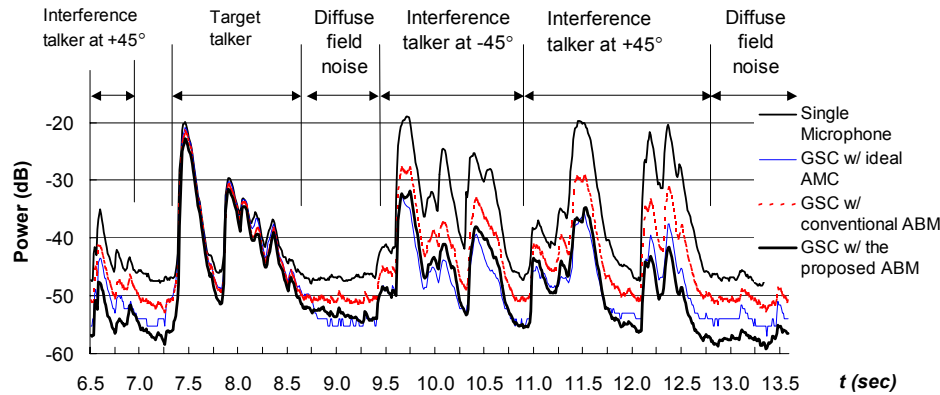


Fig. 8 Output power of single microphone, GSC with ideal AMC, GSC with conventional ABM, and GSC with the proposed ABM ( $M=8$ ,  $d=4\text{cm}$ ,  $f_s=12\text{kHz}$ )

interference rejection caused by mistracking is reduced to 1.06dB. Furthermore, the GSC with the proposed ABM can also help reduce the loss of the rejection of diffuse field noise from 3.92dB to 1.13dB. At the same time, the target cancellation still can be kept at acceptable level.

GSC code available and for fruitful discussions about robust GSC.

## REFERENCES

- [1] L. J. Griffiths and C. W. Jim, "An alternative approach to linear constrained adaptive beamforming," *IEEE Trans. Antennas Propagat.*, vol. AP-30, pp. 27-34, Jan. 1982.
- [2] B.D. Van Veen, K.M. Buckley, "Beamforming: A versatile approach to spatial filtering," *IEEE ASSP Magazine*, pp. 4-24, April 1988.
- [3] J. E. Greenberg, and P. M. Zurek, "Evaluation of an adaptive beamforming method for hearing aids," *J. Acoust. Soc. Amer.*, vol. 91, no. 3, pp. 1662-1676, Mar. 1992.
- [4] M. S. Brandstein, D. B. Ward. *Microphone arrays: Signal processing techniques and applications*, Springer, Berlin, 2001.
- [5] O. Hoshuyama, A. Sugiyama, A. Hirano, "A robust adaptive beamformer for microphone arrays with a blocking matrix using constrained adaptive filters," *IEEE Trans. on Signal Processing*, vol. 47, no. 10, pp. 2677-2684, October 1999
- [6] O. Hoshuyama, A. Sugiyama, "An adaptive microphone array with good sound quality using auxiliary fixed beamformers and its DSP implementation," in *Proc. IEEE Int. Conf. Acoust., Speech, Signal Process.*, Mar 1999, pp. 949-952.
- [7] W. Herboldt, W. Kellermann. "Frequency-domain integration of acoustic echo cancellation and generalized sidelobe canceller with improved robustness". *European Transactions on Telecommunications*, vol. 13, no. 2, pp. 123-132, March 2002.
- [8] W. H. Neo and B. Farhang-Boroujeny, "Robust microphone arrays using subband adaptive filters," *IEE Proc.-Vis. Image Signal Process.*, Vol. 149, No. 1, February 2002, pp.17-25

Table 1. Loss of performance compared with GSC using ideal adaptation mode control (AMC) (from data in Fig. 8)

Beamformer	GSC w/ Ideal AMC	GSC w/ conventional ABM	GSC w/ proposed ABM
Target Cancellation (dB)	-0.80	-1.40	-2.69
Interference Rejection (dB)	-15.44	-9.13	-14.38
-Loss of performance		-6.31	-1.06
Rejection of diffuse field noise (dB)	-7.74	-3.82	-6.60
-Loss of performance		-3.92	-1.13

## 5. CONCLUSIONS

A new adaptive blocking matrix with exact FIR structure is proposed in this paper to enhance the robustness of the generalized sidelobe canceller against inevitable mistracking in real environment. Both simulations and experiment on real recordings show that the loss of interference rejection caused by mistracking is reduced significantly.

## 6. ACKNOWLEDGEMENT

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