An Improved Implementation of GSC Filter

Quan Trong The
Information Technologies and Programming Faculty
University ITMO
St.Petersburg, Russia Federation
quantrongthe1984@gmail.com

***Abstract*— Algorithms, which are used in the microphone array, are currently widely popular, especially the GSC algorithm. Compared with the single-channel algorithm model, GSC uses information of direction of arrival (DOA) of the original speech signal to increase the efficiency of noise suppression. However, in practical applications, there are many restrictions, such as the error of the actual direction, the error of sensitivity between microphones, which affects the output signal, causes speech distortion and performance degradation. In this paper, the author proposed an efficiency method to improve the effectiveness of Generalized Sidelobe Canceller (GSC) for extracting the desired signal. The method, which uses a priori SNR based gain function and additional equalizer, has proven in experiments in reducing speech distortion from 10.5÷19.5 (dB). The use of the proposed adaptive beamforming GSC can effectively reduce the impact of strong interference and directional interference on useful signals when the surrounding working environment deteriorates, greatly improve the output signal-to-noise ratio, and accurately detect target speaker.**

Keywords*— *Generalized Sidelobe Canceller; noise reduction; a priori SNR; equalizer; target speaker

I. Introduction

In a complex acoustic environment, various interference noises will seriously affect the acquisition of the desired voice signal, so that the obtained speech signal is no longer a pure voice signal. If the interference noise is too large, the signal collected through the microphone is often blurred and affects the intelligibility of speech. It is also difficult to obtain satisfactory enhancement effects with a single microphone speech enhancement system. Therefore, in the 1980s, microphone array speech enhancement technology [14] has developed. Compared with single-isolated microphone enhancement methods, microphone array uses the spatial information of the target signal, noise and interference, and enhances the signal in the desired direction and suppresses the signal in other directions, so it can provide better enhancement effects.

Causes unwanted factors, speech signals captured by microphones are seriously usually degraded by reverberation, interference and background noise. Therefore, it is very essential to eliminate the influence of noise and improve the performance of speech processing system. At present, microphone array speech enhancement methods can be roughly divided into three categories: fixed beamforming algorithms, adaptive beamforming algorithms and fixed beamforming algorithms with post filters. The current mainstream is still adaptive beamforming with generalized sidelobe cancellers (GSC) [5-11]. The algorithm is based on suppressing noise, but it has the problems of speech cancellation and weak suppression of incoherent noise. In 1982, Grifft L.G proposed algorithm GSC [5], which nowadays widely used in the most of applications: audio surveillance, speech recognition, speech enhancement, cock-tail party.

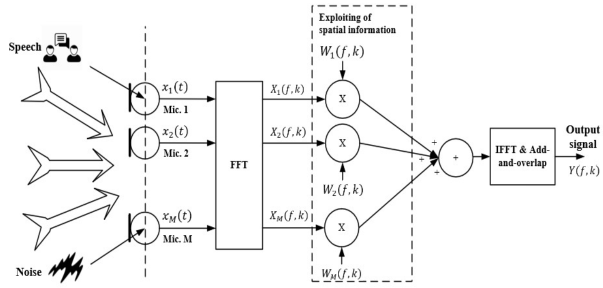


Fig. 1: The scheme of microphone array.

Due to high performance and and low computational complexity, GSC become a major research problem in almost versatile approach to enhance the useful target speaker and suppress noise in microphone array processing technology. Many scholars has carried on improvement of GSC structure for enhancing performance in complex environments. Some optimized technologies has published. Coefficent-Constrained Adaptive Filter-Norm-Constrained Adaptive Filter (CCAFNCAF) [6] is one the most perspective algorithm, which eliminates error of direction of desired signal and improve GSC structure by adding BM module and MC module. In [11], wavelet domain base post-filtering proposed to reduce adaptively interference. Some scholar suggested using linear constraint minimum variance beamformer for obtaining reference signal; but in this approach, the correlation between the final signal GSC beamformer and the output BM module still available; and BM module still has effect on GSC structure.

This paper proposes a microphone array speech enhancement method based on the combination of spectral gain function, which based on a priori SNR, and an equalizer. The basic idea is to first perform processing on the main and reference signal to obtain the a priori SNR, calculate the value of spectral gain to eliminate the residual background. For deriving the original useful captured signal on microphone, an equalizer is used. The principle and structure of the presented method are explained in the rest of this paper. The effectiveness and the quality of processed signal are shown in experiments.

2021 IEEE Conference of Russian Young Researchers in Electrical and Electronic Engineering (ElConRus) | 978-1-6654-0476-1/20/$31.00 ©2021 IEEE | DOI: 10.1109/ElConRus51938.2021.9396698

978-1-6654-0476-1/21/$31.00 ©2021 IEEE

II. GENERALIZED SIDELOBE CANCELLER

The generalized sidelobe canceller is a classic adaptive beamforming algorithm. It is mainly composed of a fixed beamformer, a blocking matrix and an adaptive noise canceller. The fixed beamformer is generally realized by a delay and summation beamformer. The main purpose is to enhance the target signal in the main lobe direction while suppressing others. The purpose of the directional noise signal, the blocking matrix, is to generate a signal that does not contain the target speech. It provides a reference signal for suppressing the noise in the fixed beamformer in the adaptive interference canceller. The adaptive interference canceller generally uses normalized minimum uniformity. GSC uses the reference noise signal to estimate the noise signal in the fixed beam output signal, and then subtract the estimated signal from the fixed beamforming output signal to achieve the enhancement of the target signal. The structure diagram is shown in the diagram

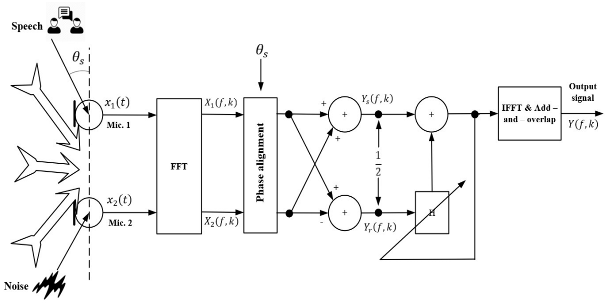


Fig. 2: The GSC Structure.

In the above figure, the signals received by each microphone are estimated and compensated for time delay. Fixed beamformer (FBF), which usually is a delay and sum beamformer, focus on the target direction of speaker. Blocking matrix (BM) ensures only noise and interference signal can pass without speech leakage. Algorithm Wiener filter used for extracting useful signal from FBF through using information of noise via BM.

III. THE PROPOSED METHOD

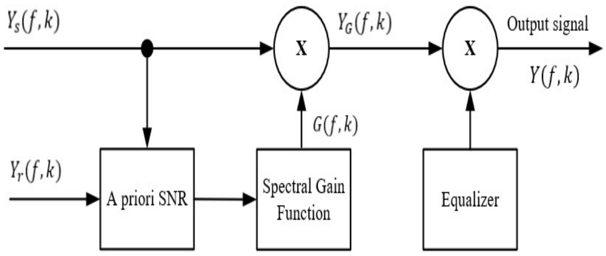


Fig. 3: The proposed method.

Assuming the proposed method is considered in frequency - domain with current frame k, frequency f, two noisy signals $X_1(f,k), X_2(f,k)$ were recorded with desired signal $S(f,k)$ and additive noise. In the vector form, the short-time Fourier transform can be expressed as follows:

$$X_1(f,k) = S(f,k)e^{j\Phi_s} + V(f,k) \tag{1}$$

$$X_2(f,k) = S(f,k)e^{-j\Phi_s} + V(f,k) \tag{2}$$

The direction of arrival of useful signal is θ_s, distance between microphones is d, the sound speed is $c = 340\ (m/s)$, $\tau_0 = d/c$ is the sound delay, $\Phi_s = \pi f \tau_0 cos(\theta_s)$. From GSC struture, the output signal from FBF $Y_s(f,k)$ and BM $Y_r(f,k)$ are given by:

$$\begin{aligned} Y_s(f,k) &= \frac{X_1(f,k)e^{-j\Phi_s} + X_2(f,k)e^{j\Phi_s}}{2} \\ &= S(f,k) + V(f,k)cos(\Phi_s) \end{aligned} \tag{3}$$

$$\begin{aligned} Y_r(f,k) &= \frac{X_1(f,k)e^{-j\Phi_s} - X_2(f,k)e^{j\Phi_s}}{2} \\ &= -jV(f,k)sin(\Phi_s) \end{aligned} \tag{4}$$

When the main signal $Y_s(f,k)$ and the reference signal $Y_r(f,k)$ are available, we easily calculate power spectral densities of two main major signals as:

$$\begin{aligned} E\{Y_s(f,k)Y_s^*(f,k)\} &= E\{S(f,k)S^*(f,k)\} \\ &+ E\{V(f,k)V^*(f,k)\}cos^2(\Phi_s) \end{aligned} \tag{5}$$

$$= E\{V(f,k)V^*(f,k)\}\left(\frac{E\{S(f,k)S^*(f,k)\}}{E\{V(f,k)V^*(f,k)\}} + cos^2(\Phi_s)\right) \tag{6}$$

$$E\{Y_r(f,k)Y_r^*(f,k)\} = E\{V(f,k)V^*(f,k)\}sin^2(\Phi_s)$$

With a priori SNR $\xi(f,k)$ is defined as:

$$\xi(f,k) = \frac{E\{S(f,k)S^*(f,k)\}}{E\{V(f,k)V^*(f,k)\}} \tag{7}$$

From above equations, $\xi(f,k)$ is derived as:

$$\xi(f,k) = \frac{E\{Y_s(f,k)Y_s^*(f,k)\}}{E\{Y_r(f,k)Y_r^*(f,k)\}}sin^2(\Phi_s) - cos^2(\Phi_s) \tag{8}$$

The spectral gain function, which depends on $\xi(f,k)$:

$$G(f,k) = max\left(G_{min}, 1 - \sqrt{\frac{\mu}{1+\xi(f,k)}}\right) \tag{9}$$

Where $G_{min} = 0.03$; $\mu = 1.25$. After using the spectral gain function, the obtained signal is:

$$Y_G(f,k) = Y_s(f,k) \times G(f,k) \tag{10}$$

The effectiveness of the spectral gain function is eliminating the background noise and save the desired target speaker. However, in some real-life environment (such as surrounding environment with fan noise, wind noise, reverberation room) affect on the output signal. Adaptive beamforming GSC has strong directional noise cancellation, but much more sensitive to steering errors, which leads to degradation and leakage of useful desired signal. For overcoming this disadvantage, the author uses an equalizer H_{eq} [12] to save target speaker.

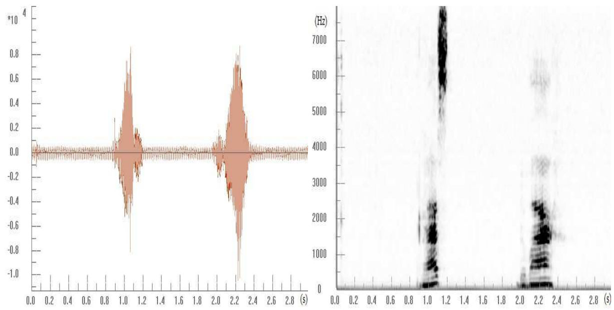
$$H_{eq}(f) = \begin{cases} \dfrac{1}{sin\left(\dfrac{\pi}{2}\dfrac{f}{f_c}\right)} & 0 < f < f_c \\ 1 & otherwise \end{cases} \tag{11}$$

Where $f_c = 1/(4\tau_0)$. The value of $H_{eq}(f)$ is limited with a determined threshold 15(dB). This equalizer ensures deriving desired signal. The output signal is:

$$Y(f,k) = Y_G(f,k) \times H_{eq}(f) \tag{12}$$

IV. EXPERIMENTS

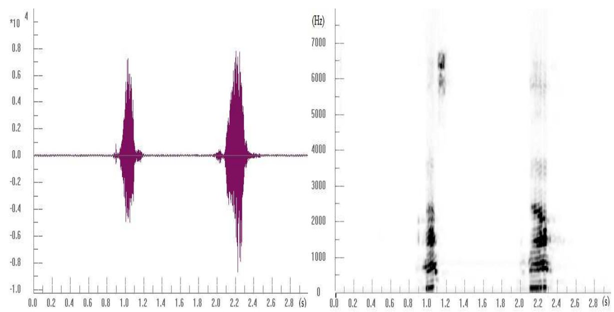
The purpose of experiment was to test the proposed algorithm (GSC-Mod) in saving speech signals and reducing noise in comparison with conventional GSC (GSC-CONV). In this section, the suggested algorithm is performed to deal problem of extracting desired signal and reducing interference noise. The speaker in front of dual - microphone at distance 1(m), distance between two microphones $d = 2.5\ (cm)$, direction of arrival target speaker $\theta_s = 0^0$. The objective measure NIST STNR [13] used to measure the signal-to-noise ratio (SNR) of captured signal on microphone and output signal. Two noisy recorded signals were sampled at sampling rate 16 kHz. For calculating PSD estimation: Hamming window, overlap 50%, 512 point FFT.



(a) (b)

Fig. 4: Amplitude and spectrogram of original signal.

As Figure 6 shows, the proposed method not only allows saving desired target speaker, reducing speech distortion; also attenuating surrounding background noise. The level of noise reduction is 20-22 (dB), speech distortion 10.5÷19.5 (dB). Method NIST STNR measured the speech quality and confirmed the increasing of the ratio signal-to-noise (SNR) from 21.3 to 34.8 (dB).



(a) (b)

Fig. 5: Amplitude and spectrogram of processed signal.

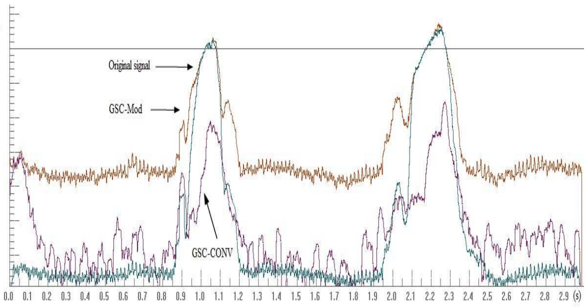


Fig. 6: Energy of captured and processed signal.

TABLE I: THE SIGNAL-TO-NOISE RATIO SNR (DB)

Method Estimation	Original signal	GSC-Mod
NIST STNR	21.3	34.8

V. CONCLUSION

The microphone array has many obvious advantage to extract the desired signal and suppress the influence of interference signals and noise. The performance of microphone array system has high noise reduction, a better signal gain, a useful spatial resolution. Due to many objective reasons, such as: the different sensitivities of microphones, the mismatch of sampling recorded signals, the error of direction of arrival, that leads to affect processing of speech enhancement microphone array. In this paper, the author proposed a microphone array speech enhancement method based on generalized sidelobe canceller, a gain spectral function and an equalizer. The target speech is initially enhanced by the modified generalized sidelobe canceller. The method filters out coherent noise, some weak coherence and background noise, residual noise to improve the signal-to-noise ratio and intelligibility of the speech. In this article, the simulation experiment was carried out to prove the effectiveness and capabilities of the proposed method in increasing the ratio SNR and noise reduction. The above technology can be incorporated into system multimicrophone.

REFERENCES

[1] Microphone Arrays / ed. by M. Brandstein, D. Ward. Heidelberg, Germany: Springer-Verlag, 2001. XVIII, 398 p. doi: 10.1007/978-3-66204619-7.

[2] Benesty J., Chen J. Huang Y. Microphone Array Signal Processing. Berlin, Germany: Springer-Verlag, 2008.240 p. doi: 10.1007/978-3-54078612-2

[3] Benesty J., Chen J., Pan C. Fundaments of Differential Beamforming. Springer, 2016.122 p. doi: 10.1007/978-981-10-1046-0

[4] Benesty J., Cohen I., Chen J. Fundamentals of Signal Enhancement and Array Signal Processing. Wiley, IEEE Press, 2017.440 p.

[5] Griffiths L., Jim C. An alternative approach to linearly constrained adaptive beamforming // IEEE Transactions on Antennas and Propagation, vol. 30, no. 1, pp. 27-34, January 1982, doi: 10.1109/TAP.1982.1142739.

[6] Hoshuyama O., Sugiyama A., Hirano A. A robust adaptive beamformer for microphone arrays with a blocking matrix using constrained adaptive filters // IEEE Transactions on Signal Processing, vol. 47, no. 10, pp. 2677-2684, Oct. 1999, doi: 10.1109/78.790650.

[7] Ali R., Waterschoot T., Moonen M. Generalised Sidelobe Canceller for Noise Reduction in Hearing Devices Using an External Microphone //

[8] Proc. 2018 IEEE International Conference on Acoustics, Speech and Signal Processing (ICASSP), Calgary, AB, 2018, pp. 521-525, doi: 10.1109/ICASSP.2018.8461366.

[9] Arote S., Deshpande M. Multichannel Speech Dereverberation using Generalized Sidelobe Canceller and Post Filter // Proc. 2019 5th International Conference On Computing, Communication, Control And Automation (ICCUBEA), Pune, India, 2019, pp. 1-4, doi: 10.1109/ICCUBEA47591.2019.9129323.

[10] Wu K., Su T. Design of generalised sidelobe canceller-based adaptive wideband beamformer without pre-steering delays // Electronics Letters, vol. 52, no. 3, pp. 177-179, 4 2 2016,

[11] doi: 10.1049/el.2015.3488.

[12] Kim S. M. Hearing Aid Speech Enhancement Using Phase Difference Controlled Dual Microphone Generalized Sidelobe Canceller // IEEE Access, vol. 7, pp. 130663-130671, 2019,

[13] doi: 10.1109/ACCESS.2019.2940047.

[14] Mahmoudi D., Drygajlo A. Combined Wiener and coherence filtering in wavelet domain for microphone array speech enhancement // Proc. 1998 IEEE International Conference on Acoustics, Speech and Signal Processing, ICASSP 98 (Cat. No.98CH36181), Seattle, WA, USA, 1998, pp. 385-388 vol.1, doi: 10.1109/ICASSP.1998.674448.

[15] Buck M., Roßler M. First order differential microphone arrays for¨ automotive applications // Proc. 7th International Workshop on Acoustic Echo and Noise Control, IWAENC. 2001. P. 19-22.

[16] https://labrosa.ee.columbia.edu/projects/snreval/.