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Differences Amplitude Based Improvement of Minimum Variance Distortionless Response Filter

Quan Trong The

Abstract: *In this work, the author introduces a new technique for improving the performance of minimum variance distortionless response filter in condition of coherent noise. The proposal algorithm exploits a priori information of differences amplitude to balance power spectral densities of observed noisy signals. The output signal of MVDR filter is then processed by an additional post-filtering, which based speech presence probability to suppress more noise interference and increase quality speech. In experiments using two noisy signal recordings in anechoic room, the modified MVDR-filter results provides that the suggested algorithm increases speech quality compared to the conventional MVDR filter.*

Keywords: **differences amplitude, microphone array, dual-microphone, speech enhancement, minimum variance distortionless response, post-filtering, speech presence probability.**

I. INTRODUCTION

In recently years, microphone arrays [1] concerned, cause it uses spatial information of geometry microphone array, coherence of condition noise field, coherence of captured noisy signal to remove background noise, and sound interference. The minimum variance distortionless response (MVDR) [2-6] beamforming algorithm is now often popular used in microphone array system. MVDR, which based on a constraint condition of minimization of total power output noise and ensures undistorted target speaker, provides a optimal solution in speech application: speech recognition, speech enhancement, surveillance. In this paper, the author proposed an effective approach to use the amplitude ratio between two noisy signals to improve the performance of MVDR filter. The post-filtering, which is a function depends on speech presence probability (SPP) [7-8], used to reduce the level residual noise, that contained in the output signal. The evaluation results show that modified MVDR increases speech quality in term signal-to-noise ratio. The promising algorithm can be used as fronted in such application hearing aids, biometric voice system.

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II. MVDR FILTER AND BALANCE AMPLITUDE

The scheme of digital processing dual-microphone system shown in Figure 1. Two microphone receives the target speech, which interference by additive noise. The spatial information is very useful for separating sound source and suppress noisy component.

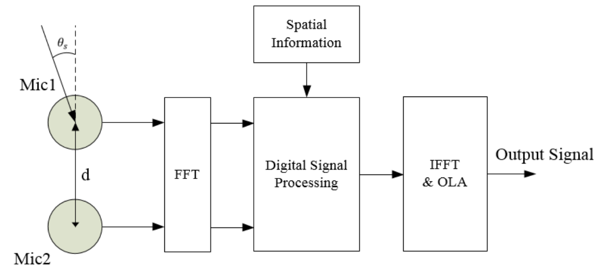

Fig. 1. Digital signal processing of dual-microphone system.

Solving the problem speech enhancement in the frequency domain. With a target speaker, which relatives the axis of MA2 by angle θ_s, the steering vector can be presented as: $\boldsymbol{V}(f,\theta_s) = \boldsymbol{V}_s(f) = [e^{+j\Phi_s} \quad e^{-j\Phi_s}]^T$, c is the sound speed (343(m/s)), $\tau_0 = d/c$ is the sound delay between the microphones, $\Phi_s = \pi f \tau_0 cos(\theta_s)$ is a vector of phase shifts between microphones. In the frequency domain, the model of relation between vector noisy signal $\boldsymbol{X}(f,k) = [X_1(f,k) \quad X_2(f,k)]^T$ and additive noise $N(f,k) = [N_1(f,k) \quad N_2(f,k)]^T$ can be expressed as:

$$X(f,k) = S(f,k)\boldsymbol{V}(f,k + \boldsymbol{N}(f,k) \qquad (1)$$

where f,k are the indexes of frequency and frame number respectively.

A priori information of direction of arrival (DOA) target speaker, a knowledge of coherence noise field or different spatial information used in the most dual-microphone algorithm. The expected results is obtain the speech component while suppressing background noise. The enhanced signal $\hat{S}(f,k)$, which obtained by evaluation can be represented as:

$$\hat{S}(f,k) = \boldsymbol{W}^H(f,k)\boldsymbol{X}(f,k) \qquad (2)$$

Where $\boldsymbol{W}(f,k)$ is the vector of the coefficients, $(\quad)^T$ is the symbol of Hermitian conjugation.

By using the inverse Fourier transform, and the overlap-and-add (OLA), the output signal is the final step of scheme.

The MVDR algorithm, which is the most optimal solution for problem extracting target speaker. Algorithm ensures that speech components, which from determined direction θ_s, remained in the output signal, while removing interference, stationary or non-stationary noise. The requirements is adapted corresponding to matrix spectral power densities [1]. And vector of the coefficients is expressed as:

$$\boldsymbol{W}(f,k) = \frac{\boldsymbol{P}_{NN}^{-1}(f,k)V(f,k)}{V^H(f,k)\boldsymbol{P}_{NN}^{-1}(f,k)V(f,k)} \quad (3)$$

where $P_{NN}(f,k)$ is a cross spectral matrix of noise signals, $P_{NN}(f,k) = E\{\boldsymbol{N}(f,k)\boldsymbol{N}^*(f,k)\}$.

In real application of speech enhancement, the estimation $P_{NN}(f,k)$ is not usually available; so in equation (3), the cross spectral matrix of observed signals is frequency used. $P_{XX}(f,k) = E\{\boldsymbol{X}(f,k)\boldsymbol{X}^*(f,k)\}$. So (3) becomes:

$$\boldsymbol{W}(f,k) = \frac{\boldsymbol{P}_{XX}^{-1}(f,k)V(f,k)}{V^H(f,k)\boldsymbol{P}_{XX}^{-1}(f,k)V(f,k)} \quad (4)$$

where matrix $P_{XX}(f,k)$ is computed as follows:

$$\boldsymbol{P}_{XX}(f,k) = \begin{bmatrix} P_{X_1X_1}(f,k) * 1.001 & P_{X_1X_2}(f,k) \\ P_{X_2X_1}(f,k) & P_{X_2X_2}(f,k) * 1.001 \end{bmatrix} \quad (5)$$

where $P_{X_iX_i}(f,k), P_{X_iX_j}(f,k), i,j \in \{1,2\}$ are the smoothed cross-spectra:

$$P_{X_iX_j}(f,k) = \gamma P_{X_iX_j}(f,k-1) + (1-\gamma)X_i(f,k)X_j^*(f,k) \quad (6)$$

γ is the smoothing parameter in the range $[0 \ldots 1]$.

III. THE PROPOSAL ALGORITHM

The author proposes a new algorithm, which uses the difference of amplitude two noisy signals for improving performance MVDR filter.

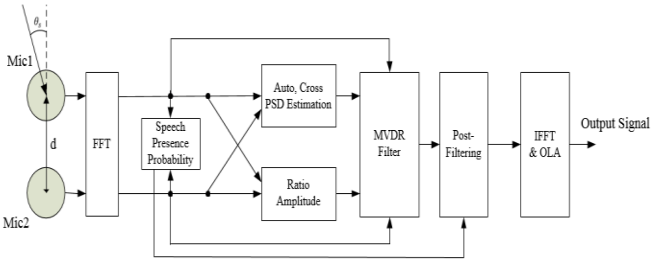

Fig. 2. The scheme of the system.

Parameter $r(f,k)$ which denotes the difference of amplitude, is:

$$r(f,k) = \frac{|X_1(f,k)|}{|X_2(f,k)|} \quad (7)$$

Matrix of observed signals $\boldsymbol{P}_{XX}(f,k)$ is modified with corresponding to difference ampltidue as:

$$\boldsymbol{P}_{XX}(f,k) = \begin{bmatrix} P_{X_1X_1}(f,k) * 1.001 & P_{X_1X_2}(f,k) * r(f,k) \\ P_{X_2X_1}(f,k) * r(f,k) & P_{X_2X_2}(f,k) * 1.001 * r(f,k)^2 \end{bmatrix} \quad (8)$$

Finally, the coefficients of modified MVDR filter are calculated by equation (4).

IV. POST-FILTERING

The post-filtering uses an estimation of speech presence probability to enhance the performance of MVDR filter. Due to microphone mismatch, DOA mismatch, the different sensitivity of microphones; so implementation of additional post-filtering is a necessary step in speech application. The post-filtering can be expressed as:

$$H(f,k) = G_{H0}^{1-SPP(f,k)} \quad (9)$$

where G_{H0} is a constant noise floor $25(dB)$.

V. EXPERIMENTS AND RESULTS

In this section, the author apply the suggested method to the speech enhancement problem and evaluate its performance in a anechoic chamber. The purpose of experiments was to compare the proposed algorithm (MVDR-Am) on real signals to the conventional MVDR (MVDR-CONV) by using the objective measure NIST STNR [9] to estimate the quality of original and processed signal. 512 point FFT, a Hamming window, and an overlap of 256 samples, $\gamma = 0.5$ are the parameters used for PSD, Cross PSD estimation. The scheme of the experiment is shown in Fig. 3. The target direction was set in the direction of the speaker ($\Phi_s = -30^0$), the distance between the microphones d = 5 cm.

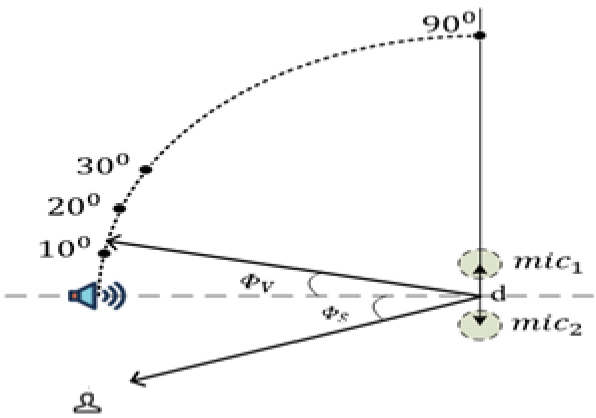

Fig. 3. The scheme of experiments.

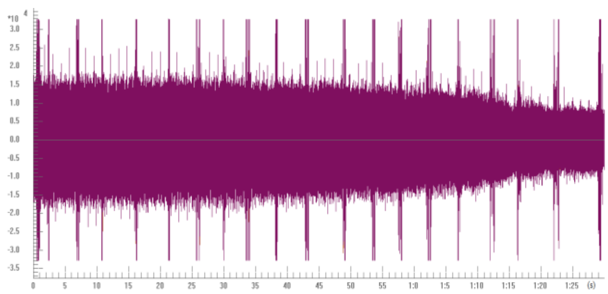

Fig. 4. Amplitude of original signal.

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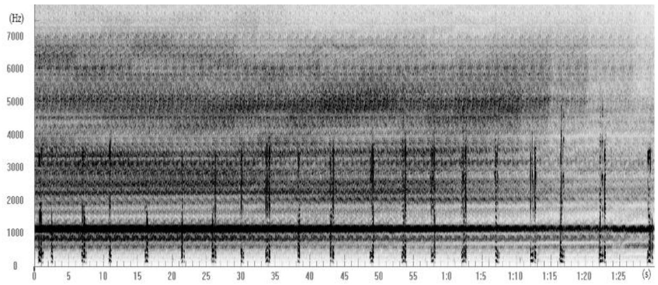

Fig. 5. Spectrogram of original signal.

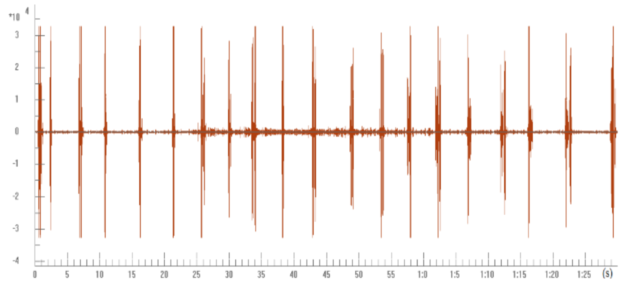

Fig. 6. Amplitude of processed signal.

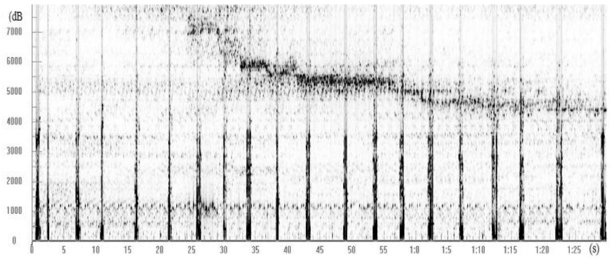

Fig. 7. Spectrogram of processed signal.

From Figure 4, 5, 6, 7; the modified MVDR filter proved that, the mismatch or sensitivity of microphones can effect on algorithm's performance. With balance amplitude, MVDR-Am remain obviously speech components, and suppress non-stationary background noise. The quality of speech enhancement is shown Table 1, signal-to-noise ratio increases to 44 (dB).

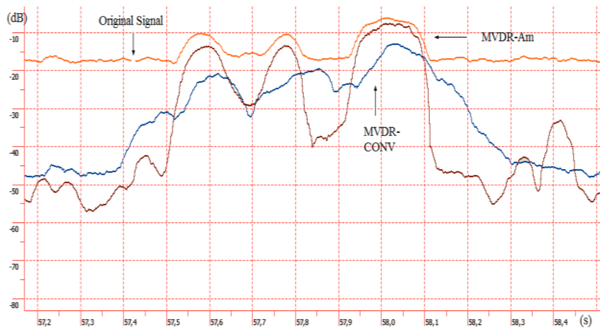

Fig. 8. Energy of original and processed signal by MVDR-CONV, MVDR-Am.

Figure 8 demonstrates that the suggested algorithm overcomes the disadvantage of MVDR-CONV. At the output, speech distortion is reduced about 8 ÷ 10 (dB).

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

Estimation Method	Signal		
	Original	MVDR-CONV	MVDR-Am
NIST STNR	4.0	27.0	44.0

VI. CONCLUSION

Dual-microphone exploits the spatial distribution of the target speaker, direction of speaker, or unwanted interfering signals and therefore reduces speech distortion than single channel algorithms. In this work, the author have introduced a favorable algorithm, which capable with problem speech enhancement in diffuse noise field. These simulations were sufficient to demonstrate the increasing of signal-to-noise ratio, and ensures the quality of processed signal.

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