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An Improvement of Minimum Variance Distortionless Response Filter

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Abstract— In this paper, the author introduces an improvement of Minimum Variance Distortionless Response's performance, which uses a priori information of speech presence probability to estimate the necessary matrix of noise. The proposal algorithm computes the smoothing parameter, that adapts with the presence or absence of speech components. The significant amount of noise reduction has provided the effectiveness and ability of increasing the signal-to-noise ratio of this algorithm's speech enhancement. Post-filtering is an additional technique to enhance the quality of the output signal. The evaluation is presented in promising results of amplitude, spectrogram of original and processed signals.

Keywords—microphone array, dual-microphone, minimum variance distortion less response, post-filtering, speech enhancement, speech presence probability

I. Introduction

Speech enhancement - related methodology are pervasively default available on acoustic devices such as smartphones, hearing aids, voice controlled equipment. Although single channel have been studied for many decades and there are many important achievements, but reverberant, incoherent, complex interference in noisy environment, which influences on useful signal and degrade speech quality, always a difficult problem for scientists. For overcoming these challenges, many current devices nowadays use microphone array [1-3] for purpose of exploiting the spatial information to improve speech quality and reduce speech distortion, that is inherent weaknesses in technique signal processing single-channel. Multi-microphone processing give us various variant of signal processing: pre-filtering, post-filtering, utilizing of characteristic microphone array to obtain suppressing a considerable amount of noise.

Among microphone array signal processing algorithm, Minimum Variance Distortionless Response (MVDR) [4-7] is one of the most widely used, due to the effectiveness and ability of maintaining the target speaker, that arrives at determined direction, while suppressing background noise. Speech presence probability (SPP) [8-9] is an essential priori information, which almost speech application need to accuracy compute required parameters.

In this paper, the author introduces an improved MVDR Filter by estimating matrix power spectral densities (PSD) of noise, instead of matrix PSD of observed signals in dual microphone system. A Minimum Statistics (MS) [10] is applied for calculating noise power. The evaluation is performed in coherent condition and compared achieved results between the proposal algorithm and traditional MVDR filter. The combination of MVDR filter and SPP is a preliminary study, furthermore the expand of research will be evaluated in various environments.

II. Overview of MVDR Filter

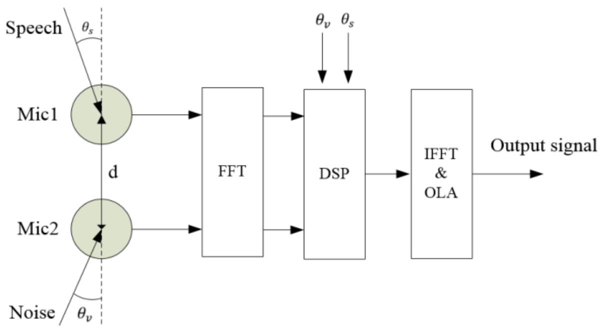

Figure 1. The scheme of dual-microphone system

A scenario of dual-microphone system as demonstrated in Figure 1; Minimum Variance Distortionless Response is an optimal solution, which derived from the problem of maintaining undistorted desired signal, which known direction of arrival related to dual-microphone, while reduce significant total poser of noise. The coefficients of MVDR filter is expressed as [1]:

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

where matrix $\boldsymbol{P}_{xx}(f,k)$ is determined as follows:

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

and $\boldsymbol{D}_s(f)$ is defined as known steering vector, which corresponding to direction of arrival (DOA) of the target speaker. $P_{X_iX_i}(f,k), P_{X_iX_j}(f,k)$ are the smoothed cross-spectra of noisy signals and recursively estimated as:

$$P_{X_iX_j}(f,k) = P_{X_iX_j}(f,k-1) + (1-\beta)X_i^*(f,k)X_j(f,k) \quad (3)$$

Where β is the smoothing parameter in the range $\{0,1\}$. The output signal is obtained by:

$$X_{out}(f,k) = \boldsymbol{W}^H(f,k)\boldsymbol{X}(f,k) \quad (4)$$

Where $(W)^H$ indicates conjugate operator, and $X(f,k) = [X_1(f,k) \quad X_2(f,k)]$ represents vector of input signals. In many others researches, MVDR filter will be extremely effective if use only matrix PSD of noise. This direction leads to approach of estimating matrix PSD of noise.

III. THE PROPOSAL ALGORITHM

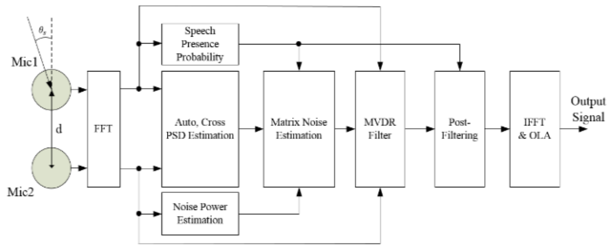

Fig. 2. The proposed algorithm

The current cross power densities is related to the previously frame and present cross power densities of two noisy signals. This effective combination ensures the rapidly adaptation to presence or absence of speech components. The author proposed an adaptive updating of a smoothing parameter, which is needed for calculation of auto and cross power spectral densities of noise.

The smoothing parameter:

$$\alpha = \frac{1}{1+\left(1-SPP(f,k)\right)^2 \times \left(\frac{P_{NN}(f,k-1)}{\sigma_n^2(f,k)}-1\right)^2} \quad (5)$$

Where $P_{NN}(f,k-1)$ is the previously power spectral densities, and $\sigma_n^2(f,k)$ is estimated noise power by MS algorithm. The above smoothing parameter contains information of the speech presence probability, that leads to better tracking the changes of instantaneously speech components. The auto and cross power spectral densities of noise are determined as recursively formulation as follows:

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

A post-filtering, which is defined as follow, is used to enhance the quality of processed speech. And post-filtering also a depended function on speech presence probability.

$$G(f,k) = G_0^{(1-SPP(f,k))} \quad (7)$$

with $G_0 = -25\ (dB)$.

IV. EXPERIMENTS AND RESULTS

In this section, the suggested algorithm (MVDR-Enh-SPP) was tested in condition of coherent noise. Two noisy signals sampled with sampling rate Fs = 16kHz. A segment of 512 samples used to transform into Short-Time Fourier Transform domain, 50% overlap of frames. NIST STNR is the objective measure [11] used to compare and evaluate the effectiveness, capability noise reduction of proposal algorithm with conventional MVDR algorithm (MVDR-CONV).

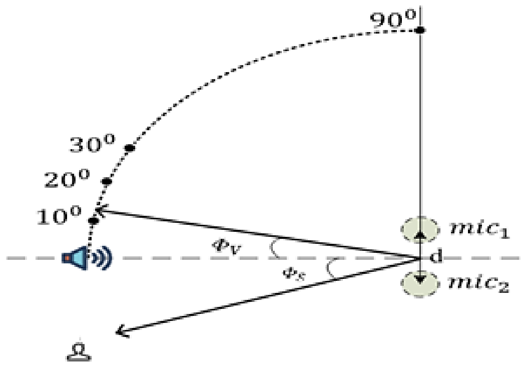

Fig. 3. The scheme of experiments

A speaker, who stood at 5(m) distance from dual-microphone system.

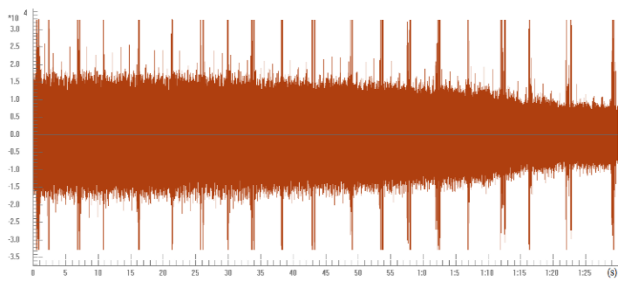

Fig. 4. Amplitude of original signal.

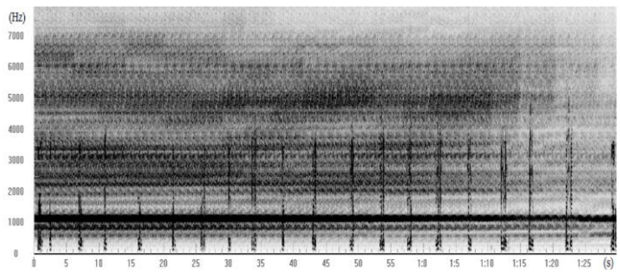

Fig. 5. Spectrogram of original signal.

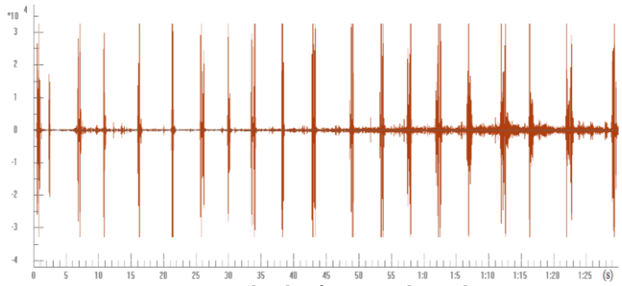

Fig. 6. Amplitude of processed signal.

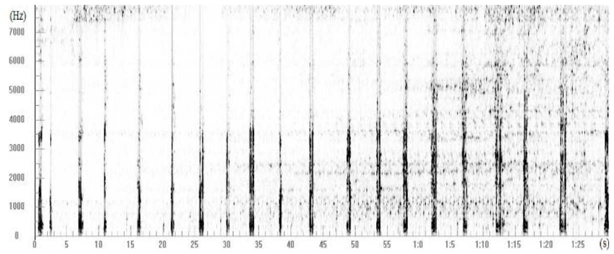

Fig. 7. Spectrogram of processed signal.

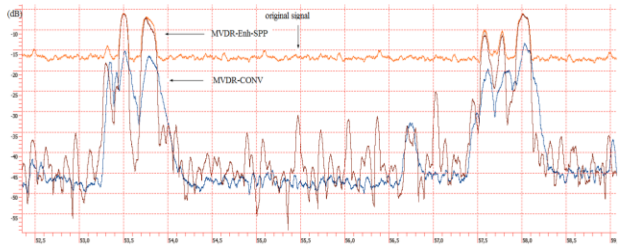

Fig. 8. Energy of original signal and processed signal by MVDR-CONV, MVDR-Enh-SPP.

The amplitude and spectrogram of original and processed signal by MVDR-Enh-SPP shown in Figure 4, 5, 6, 7. The significant noise reduction is about 23 ÷ 28 (dB). And the signal-to-noise ratio (SNR) increased from 23.5 to 39.3 (dB) as in Table I. From Figure 8, the main advantage of proposal algorithm is reduce speech distortion about 7 ÷ 8 (dB) when compared to the conventional MVDR.

Table I. The signal-to-noise ratio (dB)

Method Estimation	Original signal	MVDR-CONV	MVDR-Enh-SPP
NIST STNR	4.0	27.5	39.3

V. CONCLUSION

Noise suppression is a real challenge in speech enhancement, especially in real noisy environment. So, the tracking of power spectral densities is an obvious demand in almost speech enhancement method. In this paper, the author presented an useful method, that utilizes a priori information of speech presence probability to accurate estimate necessary matrix in MVDR filter. The evaluation is proven the ability suppress background noise and improve quality of processed signal in term signal-to-noise ratio. The promising major advantage is reducing speech distortion when compared to conventional MVDR.

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Authors Profile

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