The Spatial Information - Based Post Filtering for GSC Beamformer in Complex Situation

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Abstract

Speech enhancement plays an important role in numerous speech applications with purpose of preserving the clean speech data while suppressing the background noise, interference and third-party talker. Single-channel approach, which bases on the spectral subtraction, owns the ability of noise reduction in stationary noise environments. Unfortunately, due to the rapidly changing surrounding noise, the existence of non-directional noise, non-stationary noise degrades the performance of signal processing system and causes the speech distortion. Therefore, microphone array (MA) technology has been implemented into various types of acoustic equipment for extracting the target desired speech component while alleviating background noise. MA exploits the spatial priori information of designed configuration of MA, the characteristics of environment, the properties of processed captured MA signals to obtain a high directional beampattern towards at the specified direction. Generalized sidelobe canceller (GSC) beamformer is a very effective beamforming technique for both speech enhancement and noise reduction simultaneously. However, because of the difference microphone sensitivities, the error of sampling rate, the microphone mismatches, the displacement of MA distribution often corrupt the speech quality of GSC beamformer's output signal. In this paper, the author proposed an effective post-Filtering for mitigating noise components and increasing the signal-to-noise ratio. The demonstrated experiment has confirmed the capability of the suggested method in realistic recording scenarios to enhance the signal-to-noise ratio from 12.7 to 15.0 dB. The author's proposed post-Filtering can be integrated into multi-channel system for addressing other complicated problems, such as, automatic speech recognition, reverberation.

Keywords

generalized sidelobe canceller, microphone array, post-filtering, speech enhancement, wiener filter.

1. Introduction

Because of the influence of third-party talker, the complex and annoying environment, the existence of various types of noise, the observed speech signals often corrupted and always polluted by noise, which leads to the speech quality, speech intelligibility deteriorated, and perceptual metric listener are significantly affected, as in Figure 1. Therefore, the requirement of reducing noise and improving signal processing system's performance is an essential core, which attracted numerous scholars and engineering to study and illustrate experiment.

Single - channel approach is an efficient technique to extract the original speech component in stationary noise. However, this direction often based on spectral subtraction, which does not correctly outperform in non - stationary and complex noise field, cause speech distortion and degrades the satisfactory of listener. In recent years, MA technology has been popular installed in numerous speech applications, such as, surveillance device, teleconference system, smart - home, voice - controlled equipment, cochlear implant, hearing aids.

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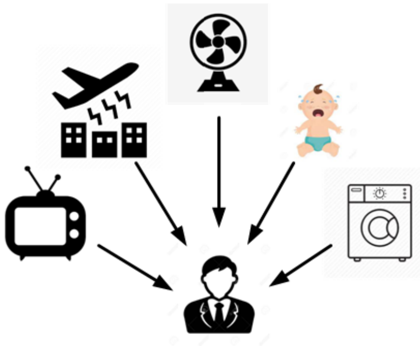

Figure 1: Illustration of multiple acoustic noise source in everyday environmnets.

MA beamforming uses the prior spatial information, the designed configuration of microphone distribution to obtain the steerable beampattern towards the sound source while suppressing the background noise, interference or third - party speaker from other directions. MA beamforming includes different techniques: Delay - and - sum (DAS) [1], Differential microphone array (DIF) [2-4], Minimum variance distortionless response (MVDR) [5-7], Linearly constraint minimum variance (LCMV) [8-10] and GSC beamformer [10-18]. These beamforming methods exploit the preferred steering vector or optimum constrained criteria of minimizing the output noise power while saving the clean speech data, as in Figure 2.

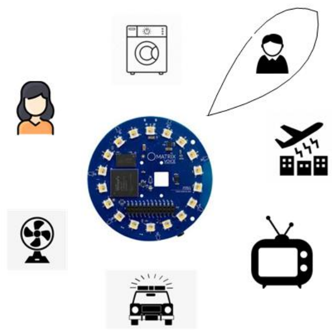

Figure 2: Microphone array beamforming towards the sound source

GSC beamformer concerns the beampattern toward the specified direction and uses an adaptive noise canceller (ANC) for saving the speech component. GSC owns high directivity index, high performance in various types of noise fields. However, in realistic recording scenario, due to the complex and adverse environment, GSC beamformer's evaluation often corrupted. There are many efforts are studied to address this problem. The principal working of MA beamforming can be given in Figure 3.

Li S [11] proposed using the power ratio of GSC beamformer's output signal and blocking matrix to control the step size for adjusting the weight coefficients of adaptive noise suppression. The numerical simulation has confirmed the effectiveness of the suggested technique in diffuse noise field with improving the speech quality.

Wang J [12] described the adaptation control of ANC, which follows a time - varying Gaussian distribution. The advantage of this approach is calculating the target speech variance and coefficient according to the maximum likelihood criterion. The conducted experiments were demonstrated under various conditions to verify the robustness and effectiveness of the proposed method.

In [13], the external microphones were implemented to GSC structure to minimum the power distortionless response beamformer and suppress interferer. The numerical simulations shown the better speech enhancement in adverse recording scenario.

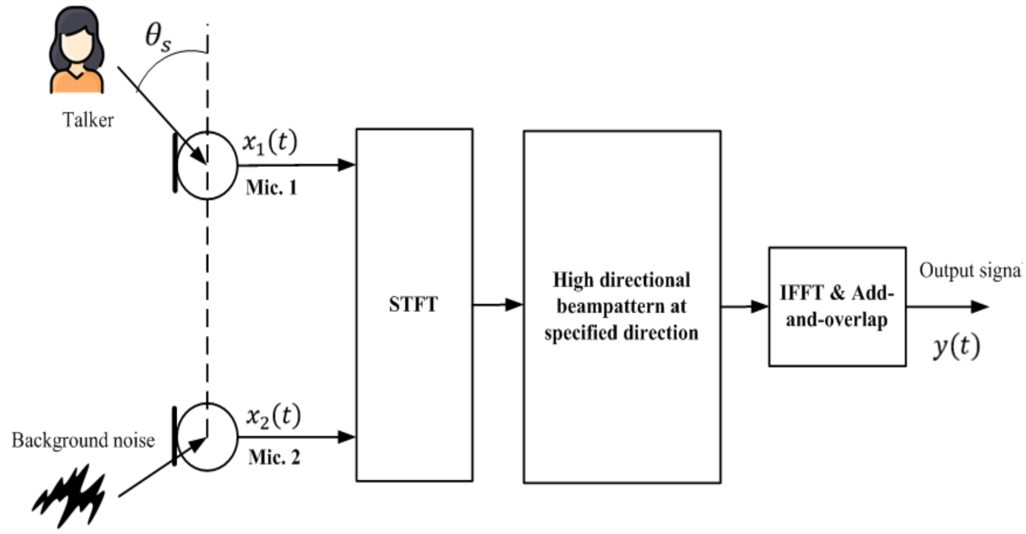

Figure 3: The pricinpal working of MA beamformer in the frequency - domain

Dai S [14] proposed an efficient algorithm for generating blocking matrix of GSC beamforming. The approach applies simplified zero placement algorithm to obtain independent full space vectors. In comparison with conventional singular value decomposition, the suggested method owns more constraints in numerous situations.

Park J [15] proposed method with two stages in GSC beamformer, which contains determinant Wiener filter and subsequence noise canceller to alleviate musical noise, residual noise. The author's technique employs signal activity detector to increase the overall performance. The demonstrated experiment has confirmed the effectiveness through spectrogram, mean opinion score and objective measures.

Li B [16] suggested new optimization algorithm for removing leakage of speech caused by inaccurate estimation of direction of arrival (DoA) and the correlation between the output signal of beamformer and blocking matrix module is used for adjusting adaptively ANC's coefficient. The simulation results show that the proposed technique has better noise suppression and speech enhancement.

In [17], the feature of interaction between adaptive beamforming and multi-channel post-filtering applied for achieving more robust adaptive beamforming in adverse environment. The experiment show that suggested direction obtains considered speech enhancement, improvement in the terms of noise suppression.

Li J [18] incorporates frequency domain independent component analysis and GSC beamformer to perform a steep null in desired target talker. The experimental and numerical simulation show the advantage of author's approach in better noise reference canceller.

These above efforts attempt to increase the GSC beamformer's performance in complex and adverse environment with modifying the internal structure, the control of blocking matrix or the ANC rule. However, because of the microphone mismatches, the different microphone sensitivities, the error of sampling frequency, the displacement of MA, the moving head of speaker, which distort speech component. The above research cannot properly outperform in complex and annoying conditions. In this paper, the author proposed a modified Wiener filter, which works well in realistic recording situation for suppressing the musical noise, residual noise, considerable noise level and improving the signal-to-noise ratio of the GSC beamformer's output. The author's method can be applied into real-time impact acoustic equipment, which does not require larger database or multi-channel microphones. The author's approach is introducing an easy executed technique, which can be integrated into compact acoustic equipment for improving GSC beamformer's performance in adverse or complex environment. This method does not require large database for training or testing.

The rest of this paper is organized as: Section II presents the GSC beamformer structure and the principal working to obtain the helpful signal, Section III illustrates the author's proposed post–Filtering, Section IV illustrates the experiment in realistic experiment, Section V concludes and the author's future direction.

2. Generalized sidelobe canceller beamformer

As in Figure 4, in this section, the author describes the representation of GSC beamformer by using dual-microphone system (DMA2). In the short-time Fourier transform, the formulation of received array signals $X_1(f,k), X_2(f,k)$ can be presented at current frequency f, current frame k, as:

$$X_1(f,k) = S(f,k)e^{j\Phi_s} + N_1(f,k) \tag{1}$$

$$X_2(f,k) = S(f,k)e^{-j\Phi_s} + N_2(f,k) \tag{2}$$

where $S(f,k)$ is the original clean speech data, $N_1(f,k), N_2(f,k)$ is additive noise at two microphones, $\Phi_s = \pi f \tau_0 cos(\theta_s)$, θ_s is the preferred incident angle of desired target speaker to the axis of DMA2, $\tau_0 = d/c$, d is the range between microphones, $c = 343(m/s)$ is sound speed propagation in the air.

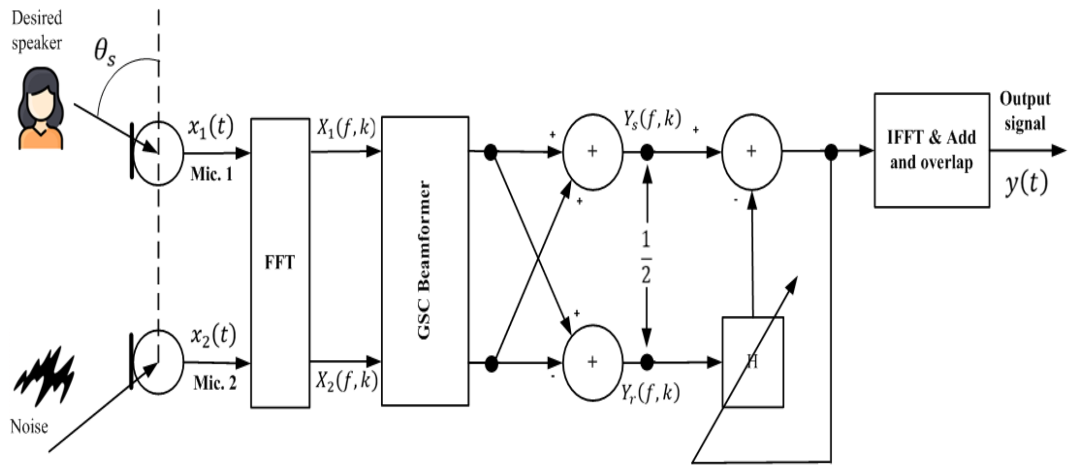

Figure 4: The scheme of GSC beamformer to recover the original speech component.

GSC structure contains three parts: the fixed beamformer, and blocking matrix and ANC. The fixed beamformer steer the beampattern at specified direction of target speech source, while blocking matrix attenuates the speech component for obtaining the only noise, and ANC extracts the desired clean speech data from fixed beamformer's output. The fixed beamformer often uses DAS beamformer, due to its simplicity computation. DIF is applied for blocking matrix to achieve the only noise component and Wiener filter is implemented as adaptive signal processing algorithm.

The output of fixed beamformer and blocking matrix is $Y_s(f,k)$, $Y_r(f,k)$ respectively. The formulation of $Y_s(f,k), Y_r(f,k)$ can be expressed as:

$$Y_s(f,k) = \frac{X_1(f,k)e^{-j\Phi_s} + X_2(f,k)e^{j\Phi_s}}{2} \tag{3}$$

and

$$Y_r(f,k) = \frac{X_1(f,k)e^{-j\Phi_s} - X_2(f,k)e^{j\Phi_s}}{2} \quad (4)$$

The Wiener filter $H_{Wiener}(f,k)$ is applied for extracting the desired target speaker as the formulation:

$$H_{Wiener}(f,k) = \frac{E\{Y_s(f,k)Y_r^*(f,k)\}}{E\{Y_r(f,k)Y_r^*(f,k)\}} \quad (5)$$

where $*$ is the asterisk operator.

The auto-cross power spectral densities of $Y_r(f,k)$, $Y_r(f,k)$ and $Y_s(f,k)$ can be calculated as the following equations:

$$P_{Y_sY_r}(f,k) = \alpha P_{Y_sY_r}(f,k-1) + (1-\alpha)Y_s(f,k)Y_r^*(f,k) \quad (6)$$

$$P_{Y_rY_r}(f,k) = \alpha P_{Y_rY_r}(f,k-1) + (1-\alpha)Y_r(f,k)Y_r^*(f,k) \quad (7)$$

where α is the smoothing parameter in the range $\{0 \dots 1\}$.

The final output signal can be derived as the following equation:

$$Y_{GSC}(f,k) = Y_s(f,k) - Y_r(f,k)H_{Wiener}(f,k) \quad (8)$$

In realistic recording scenario, due to complex and annoying recording scenario, microphone mismatches, the error of estimation of steering vector, the displacement of microphone geometry, GSC beamformer's performance under real-life situations constantly degraded. Therefore, the following section describes the suggested approach for further suppressing remained noise at GSC beamformer's output.

3. The author's suggested approach

The author's idea is modifying the Wiener filter - based traditional post – Filtering according to the prior information of direction of arrival of interest talker.

The Wiener filter - based post-Filtering has the formulation as:

$$pF(f,k) = \frac{\sigma_s^2(f,k)}{\sigma_s^2(f,k)+\sigma_n^2(f,k)} \quad (9)$$

Where $\sigma_s^2(f,k)$ is variance of speech and $\sigma_n^2(f,k)$ is the variance of noise.

By applying the microphone array beamforming, the variance of speech can be computed as:

$$\sigma_s^2(f,k) = \frac{1}{D_s^H(f,\theta_s)\boldsymbol{\Phi}_{XX}^{-1}(f,k)D_s(f,\theta_s)} \quad (10)$$

where $D_s(f,\theta_s) = [e^{j\Phi_s} \quad e^{-j\Phi_s}]^T$, $\boldsymbol{\Phi}_{XX}(f,k) = E\{\boldsymbol{X}^H(f,k)\boldsymbol{X}(f,k)\}$, $\boldsymbol{\Phi}_{XX}(f,k)$ is the covariance matrix of observed microphone array signals, $\boldsymbol{X}(f,k) = [X_1(f,k) \quad X_2(f,k)]^T$, and $\sigma_n^2(f,k)$ is variance of noise, which can be calculated by minimum statistic [19].

Based on the prior information of the impinging incident of speaker, θ_s, the author uses the spectral gain $G_k^{DSB}(f,k)$ and $G_k^{BM}(f,k)$ [20] which determined as:

$$G_k^{DSB}(f,k) = 0.5 + 0.5e^{j\psi_{12}^{norm}(2\pi f)} \quad (11)$$

$$G_k^{BM}(f,k) = 1 - e^{j\psi_{12}^{norm}(2\pi f)} \quad (12)$$

Where $\psi_{12}^{norm}(2\pi f) = \frac{\vartheta_{12}.c}{f.d}$, ϑ_{12} is the phase difference of two received array signals.

The author's suggested post - Filtering is combining the Wiener filter - based traditional post-Filtering and the spectral gain to form an appropriate post-Filtering as:

$$pF(f,k) = \beta \frac{\sigma_s^2(f,k)}{\sigma_s^2(f,k)+\sigma_n^2(f,k)} + (1-\beta)\frac{G_k^{DSB}(f,k)}{G_k^{DSB}(f,k)+G_k^{BM}(f,k)} \tag{13}$$

where β is a smoothing parameter, in the range of $\{0 \ldots 1\}$.

The final output signal of GSC beamformer is filtered as:

$$\hat{Y}_{GSC}(f,k) = Y_{GSC}(f,k) \times pF(f,k) \tag{14}$$

The advantage of the author's proposed method is utilizing the spatial information direction of arrival of helpful signal, the phase difference to adaptively control the Wiener filter - based post Filtering for suppress residual noise, a certain noise level. The effectiveness of this approach can be depicted in the next section.

4. Experiments

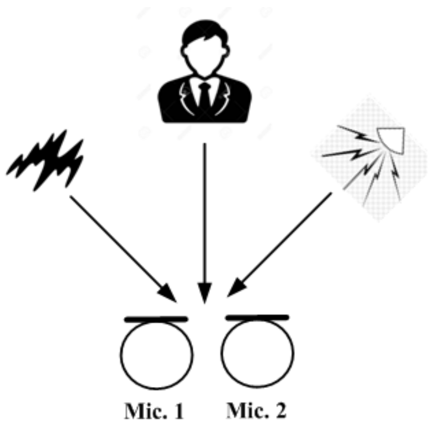

Figure 5: The demonstrated experiment by using dual - microphone system.

The purpose of this section is illustrating the experiment in real - life recording scenario to demonstrate the efficient post-Filtering in reducing residual noise, surrounding noise and enhancing the overall speech enhancement of GSC beamformer. The author uses an objective measurement [20] for calculating the SNR between the observed MA signals, the processed signals by traditional GSC beamformer (tGb) and the author's proposed post-Filtering (appF). The scheme of experiment is given in Figure 5, in living room $5x4.5x3.5$ (m) with existence of third - party talker, washing machine, non-directional noise.

A speaker stands at distance $L = 5(m)$ to dual - microphone system (DMA2), the impinging incident angle of helpful signal is $\theta_s = 90$ (deg) relatives to the axis of DMA2, the range between two mounted microphones is $d = 5(cm)$. For capturing the original speech data, sampling frequency $Fs = 16kHz$, overlap 50%. The received array signals can be shown in Figure 6 and Figure 7.

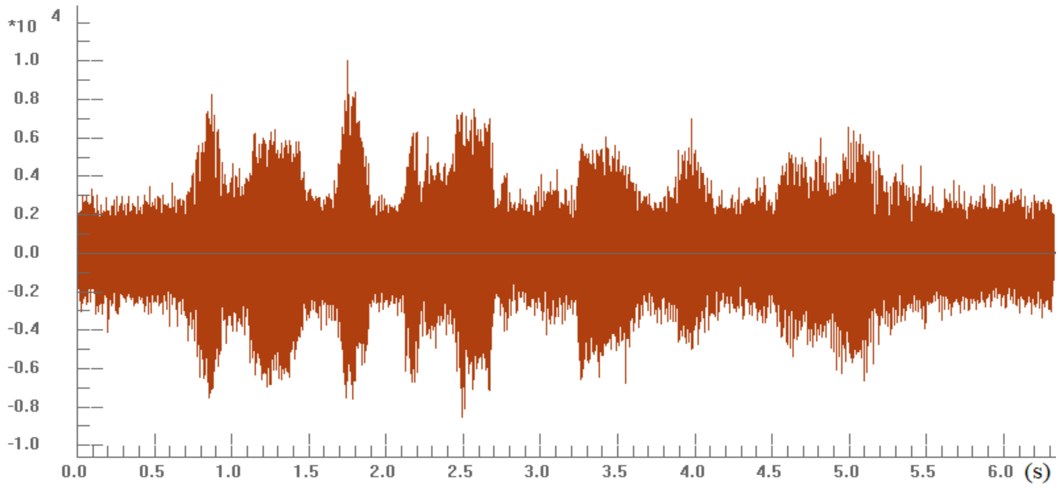

Figure 6: The waveform of observed microphone array signals.

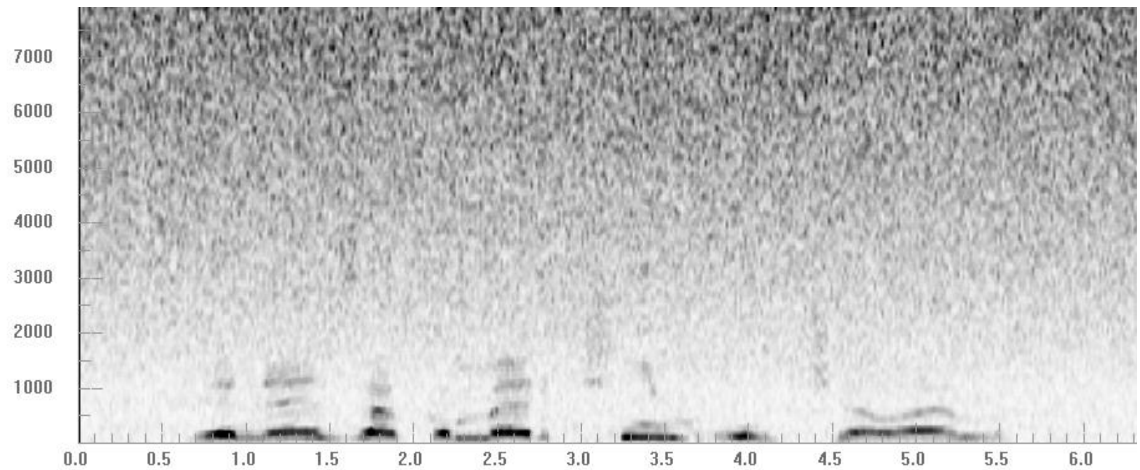

Figure 7: The spectrogram of received array signals.

With smoothing parameter $\alpha = 0.1$, the output signal of tGB is derived as:

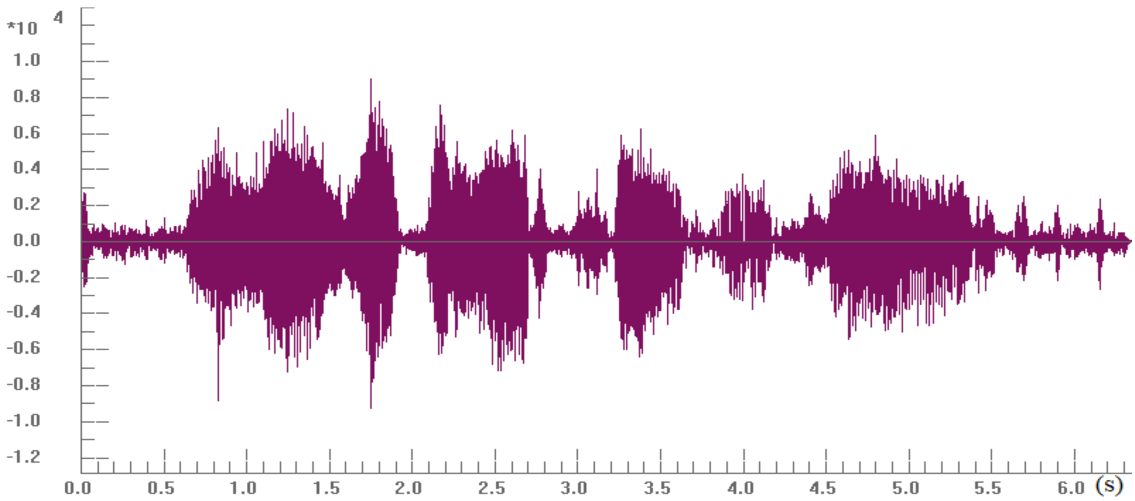

Figure 8: The processed signal by applying tGb.

Due to the complex and adverse environment, the undetermined acoustical factors, the different microphone sensitivities, the error of sampling frequency, the moving head of talker, microphone mismatches, the GSC beamformer's performance often degraded. Hence, remained musical noise, residual noise, speech distortion is challenging task in almost speech applications.

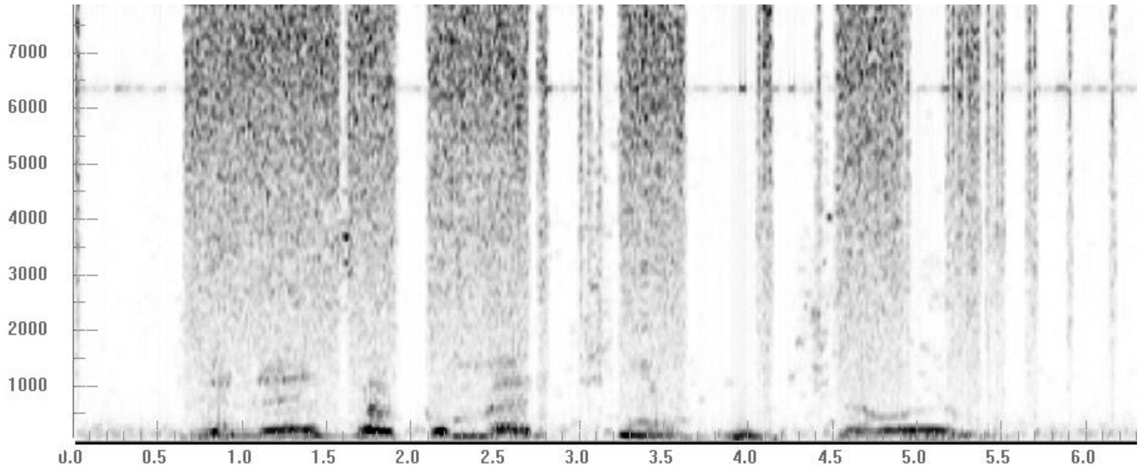

Figure 9: The spectrogram of tGb's output signal.

Therefore, efficient post-Filtering allows removing this drawback and enhancing the speech quality. The promising result by implementing appF with $\beta = 0.98$ is given in Figure 10 and Figure 11.

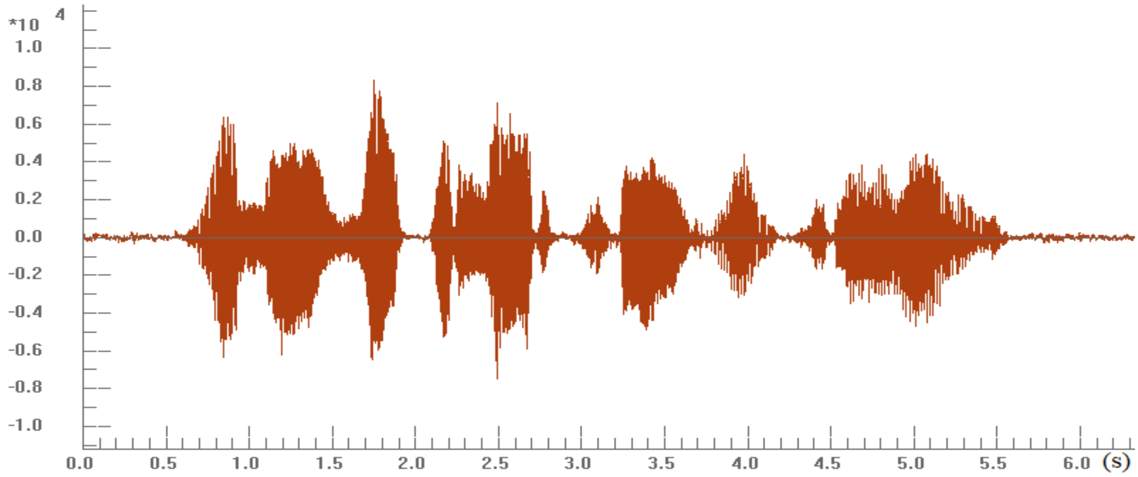

Figure 10: The promising result by using appF.

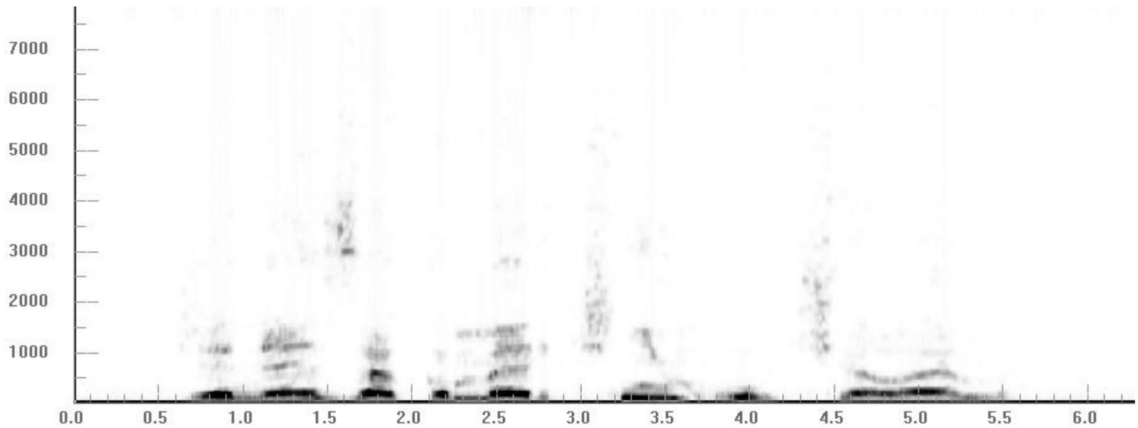

Figure 11: The spectrogram of appF's output signal.

Figure 12 describes the comparison of energy between MA signals, the processed signals by tGb and appF.

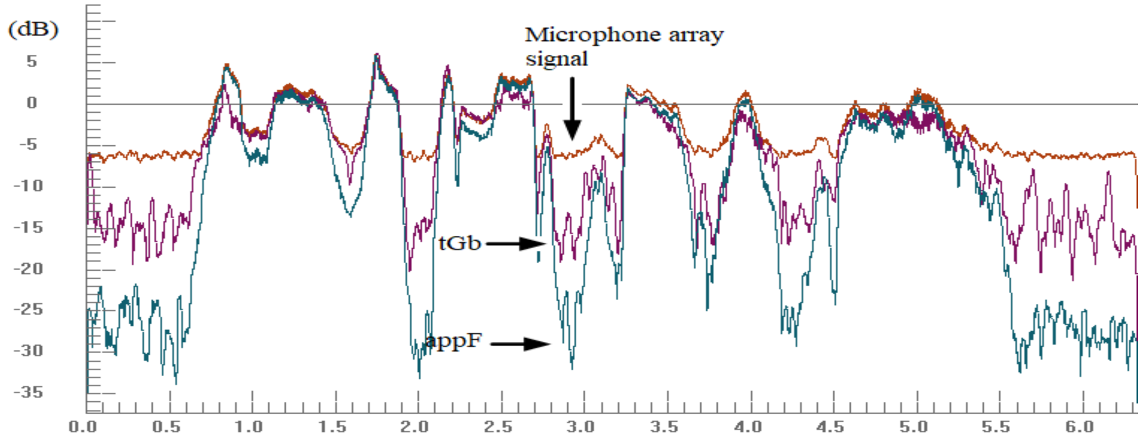

Figure 12: The comparison energy between microphone array signal and processed signals by tGb and appF.

From the numerical simulation, we can see that appF not only suppressed residual and musical noise, but also removed noise level to 12.5 dB and improved the overall speech intelligibility in the term of SNR from 12.7 to 15.0 dB as in Table 1.

The advantage of the author's proposed method is utilizing the spectral gain of MA beamforming to modified the post-Filtering, which based on Wiener filter. Because of the complex condition, the presence and absence speech component cause the difficulty of computing Wiener filter, this approach compensates the distortion of Wiener filter. The promising result of the above method is improving the accuracy of Wiener filter according to the presence/absence of speech component. The numerical simulation shows noise reduction and preserving clean speech data. The described technique can be integrated to dual-microphone system for immediately addressing speech enhancement problem in real-life recording environment.

Table 1
The signal-to-noise ratio (dB)

Method Estimation	Microphone array	tGb	appF
NIST STNR	7.5	16.5	29.2
WADA SNR	3.8	10.7	25.7

WADA (Waveform Amplitude Distribution Analysis) SNR [21] based on the constrained criteria of the speech component distributed according to Gamma model and additive Gaussian noise. Based on sequential Gaussian mixture estimation, NIST (National Institute of Standards and Technology) [22] calculates the STNR (Signal-To-Noise Ratio).

5. Conclusion

Microphone technology has been commonly used in numerous acoustic equipment, such as teleconference system, hearing aids, voice - controlled device, smart-phone, cochlear implant, surveillance device. MA beamforming uses the priori information of spatial distribution, designed configuration to obtain the advantage of noise reduction, speech enhancement simultaneously with acceptable perceptual metric listener, speech quality. GSC beamformer owns the high directivity index at the specified direction and exploits an adaptive noise canceller to extract the desired target speaker while suppressing background noise. In this paper, the author proposed using an efficient post-Filtering to remove the residual noise, surrounding noise at GSC beamformer's output signal to achieve the clean speech data with satisfied speech intelligibility. The approach uses the prior information of preferred steering vector, the spectral gain to modify essential signal-to-noise ratio to form an accurate post-Filtering. The numerical simulations have confirmed the effectiveness of the author's proposed technique in reducing the noise level to 12.5 dB and increasing the speech quality from 12.7 to 15.0 dB. The promising result has verified the capability of the suggested post-Filtering under realistic recording condition and this approach can be integrated into multi-channel signal processing system for dealing other complicated problems, such as speech recognition, reverberation. In the future, the author will incorporate the characteristic of environment to enhance post-Filtering.

Declaration on Generative AI

The author(s) have not employed any Generative AI tools.

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