A Spatial Spectral Mask for Dual - Microphone Systems

Quan Trong The
Post and Telecommunication Institute of Technology (PTIT)
Hanoi, Vietnam
theqt@ptit.edu.vn

Ninh Thi Thu Trang
Post and Telecommunication Institute of Technology (PTIT)
Hanoi, Vietnam
trangntt2@ptit.edu.vn

***Abstract*—With significantly increasing human/machine interaction, the acoustic equipment or any terminal of a multimedia service system is challenged to ensure the speech intelligibility, speech quality or comfortable perception for the listener. Therefore, speech enhancement is very necessary procedure for front-end processing in most speech applications, such as: high quality audio recording, dialog systems, dictation systems, audio - video conference, computer games. Recently, scholars and researchers have paid much attention to studying microphone array (MA) technology for both extracting desired target speech components at a certain direction and suppressing background from other directions. Generalized Sidelobe Canceller (GSC) beamformer is suitable installed for MA - based acoustic devices, which includes hearing aids, mobile phones, teleconferencing system, audio-video controlled device. Because of many reasons, such as the different microphone sensitivities or the error of direction of arrival of interest signal, the GSC beamformer's output often corrupted and speech quality degraded. In this article, the authors suggested a spatial spectral mask to enhance GSC beamformer's evaluation. The illustrated simulation shows the improvement of speech quality in the terms of signal-to-noise ratio (SNR) from 2.5 to 6.8 dB and speech distortion reduced to 11.2 dB. The advantage of the author's method is the capability of incorporating into multi-channel systems.**

I. Introduction

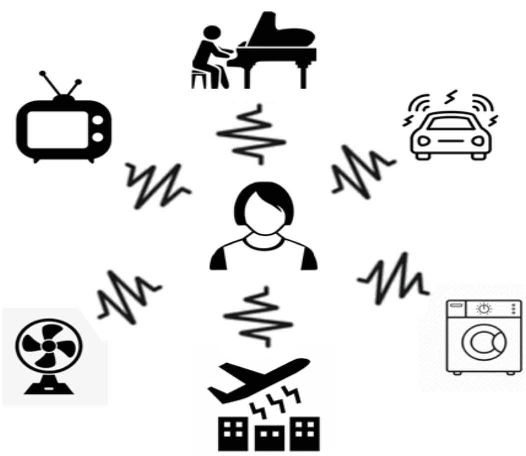

Fig. 1. The complex recording scenario around the human-life.

Stereo - sound systems, audio processing, mobile phones, hearing aids, hand-free communications are widely commonly used in speech communication. Consequently, speech enhancement plays an important role for a satisfactory perceptual listener and ensures the speech quality, understandability. Hence, many algorithms have been devoted to improving the overall performance of extracting clean signals while removing surrounding background noise, which significantly degrades the perceptual metric listener, as in Fig. 1. Generally speaking, these methods are developed based on the type of noise, recording scenarios and interference signal [1].

For single microphone cases, the useful representative algorithms include Wiener filter [2-4], spectral subtraction [5-7] because of its simplicity, low computation. However, the single-channel approach is unable to suppress musical noise or non-stationary noise, causes of speech distortion due to imprecise estimation of noise level in complex noisy environments. Recently, for a few decades, MA has become increasingly important because it both eliminates background noise and retrieves the desired target speaker at the same time, as in Fig. 2.

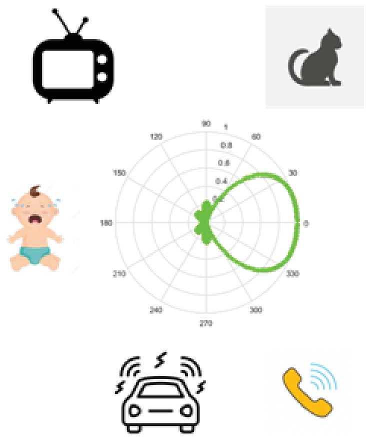

Fig. 2. The directional beampattern of MA toward the target speaker

Multi-channel speech enhancement method ensure to achieve a better evaluation than single - channel approach by utilizing the spatial diversity, such as the direction of arrival of target speaker, the properties of surrounding environment, MA geometry to obtain priori information to reduce an amount of noise level before applying signal processing algorithm, especially under scenario of non-stationary, directional third-party talker or competing speaker. In Fig. 3, as a result, MA and beamforming are now widely common applied into numerous speech application for extracting desired target speaker in complex acoustic environments by using signal processing algorithm to preserve original talker from noisy observations while removing interfering background noise, and many algorithms have been devoted, such as differential beamforming [8-10], superdirective beamforming [11-13], minimum variance distortionless response [14-15], generalized sidelobe canceller [16-18]. Consequently, these beamforming algorithms are installed into embedded man - machine systems, such as: intelligent speakers, cochlear implant device, mobile phone, teleconferencing systems, hearing aid, dialog - voice assistant system.

In beamforming algorithms, the generalized sidelobe canceller is useful for obtaining the clarity of the speech component in a noisy environment. Recently, scholars and researchers are attempting to reduce or remove the speech leakage, which often occurs in realistic GSC beamformer's evaluation.

Wang J et al. [19] introduced a robust adaptation control for adaptive interference canceller (AIC) to reduce the speech distortion, background noise and increase the perceptual metric listener and speech quality in the term of signal-to-noise ratio. This approach is based on the time-varying Gaussian distribution to adjust the controlled parameter under various complex environmental conditions.

Kim S.M et al. [20] improved GSC beamformer's evaluation by applying a phase error filter for enhancing the fixed beamformer. The author's proposed method provides better speech enhancement and noise reduction in numerous noise conditions.

In [21], external microphones were used for improving noise and interferer suppression. The research direction accurately estimates the relative transfer function vector of target talkers to improve GSC beamformer's evaluation in adverse conditions.

Based on hearing aid, Kim S. M et al. [22] proposed a new method for improving GSC beamformer's speech enhancement. The author used phase - difference to improve the fixed beamformer and adaptive noise canceller algorithm. The obtained experiment shows better intelligibility scores than other traditional methods under multiple surrounding noise.

Qayyum A et al. [23] presented a novel method for calculating the direction of sound source from the mixed speech with various different levels of background noise. This method has confirmed the effectiveness of noise reduction and extracting the desired target speaker.

However, the above research has the drawback of implementation, due to the microphone mismatches, the inaccurate estimation of preferred steering vector, the moving head of target speaker, the different microphone sensitivities, the error of sampling rate, GSC beamformer's performance often corrupted.

Beamforming technique and an additive spectral mask are two popular classes of multi-channel methods to obtain the original clean speech. The coherence - based technique was studied to perform speech enhancement in presence of speech and noise and eliminate the directional noise in situations with a dual - microphone system. As a result, the premise of the coherent noise field, the real and imaginary parts of coherence characteristics between two microphone signals, derived the optimum spectral gain.

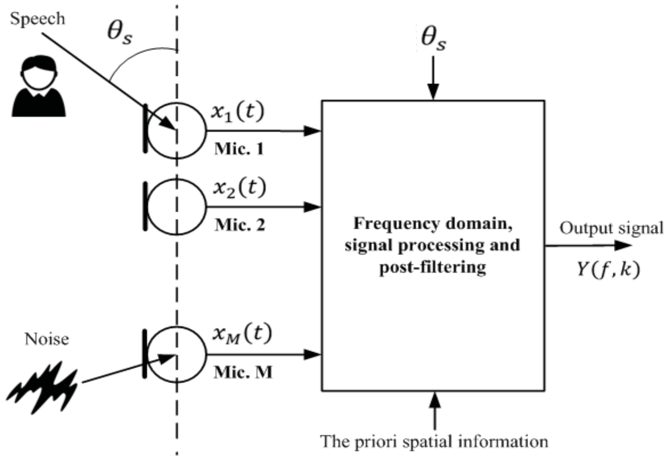

Fig. 3. The principal working of MA in the frequency domain.

This research proposed an additive spectral mask, which based on the properties of observed phase and coherence between MA signals. The main idea of the proposed scheme is forming an additive spectral mask, which allows removing the speech leakage in the reference signal of GSC beamformer. These above properties exploit the available information of microphone signals for creating an effective spectral mask.

Numerical results show that the suggested method successfully eliminates speech component level, which still exists in the reference signal, and allows achieving a better performance of GSC beamformer. Therefore, the obtained speech intelligibility and speech quality was improved from 2.5 to 6.8 (dB), speech distortion reduction to 11.2 (dB) in the realistic recording situation.

This article is organized in the following way. Section I describes the principle working of speech enhancement and MA beamforming. Section II formulates the scheme of GSC beamfomer and explains the reason for degraded evaluation. The promising phase and coherence - based spectral mask is modeled in Section III. Section IV shows the conducted experiment as figures of waveform, energy and conclusion is presented in Section V.

II. Generalized Sidelobe Canceller Beamformer

The principal working of GSC beamformer for extracting desired target speaker and suppressing background noise is described in Fig. 4. The performance in the frequency - domain.

The observed MA signals can methematical expressed as the following equations:

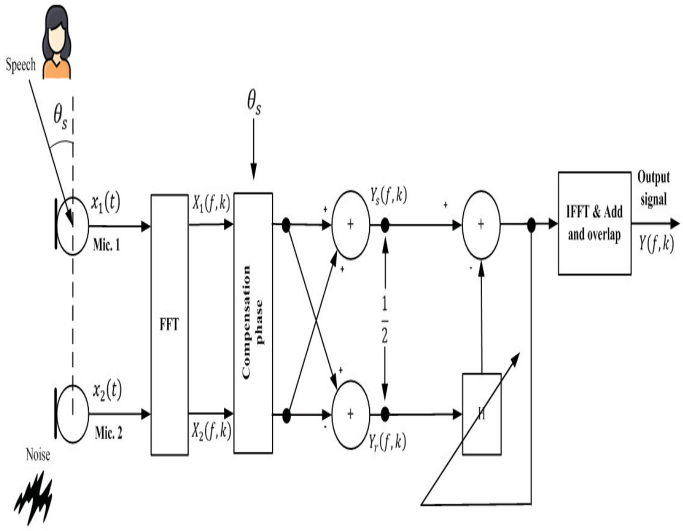

Fig. 4. The scheme of Generalized Sidelobe Canceller Beamformer.

$$X_1(f,k) = S(f,k)e^{j\Phi_s} + N_1(f,k) \quad (1)$$
$$X_2(f,k) = S(f,k)e^{-j\Phi_s} + N_2(f,k) \quad (2)$$

Where $\Phi_s = \pi f \tau_0 cos(\theta)$, $\tau_0 = d/c$, d is the distance between two mounted microphones, c is the sound speed in the air $c = 343(m/s)$, θ_s is the direction of arrival of interest talker, $S(f,k)$ denotes the original speech components, $N_1(f,k), N_2(f,k)$ are additive noise, which derived at microphone, (f,k) is current considered frequency and frame.

The upper branch of GSC beamformer focuses on the directional speech source to achieve the main signal, while the lower branch gets the information about surrounding noise or reference signal by using a designed block matrix, which depends on the microphone array distribution. The important block is the adaptive Wiener filter, that efficiently removes background noise while saving speech components. The theoretical evaluation of GSC beamformer is forming beam patterns to sound source and noise signal.

The output of upper branch $Y_s(f,k)$, lower branch $Y_r(f,k)$ can be derived as:

$$Y_s(f,k) = \frac{X_1(f,k)e^{-j\Phi_s} + X_2(f,k)e^{j\Phi_s}}{2} \quad (3)$$
$$Y_r(f,k) = \frac{X_1(f,k)e^{-j\Phi_s} - X_2(f,k)e^{j\Phi_s}}{2} \quad (4)$$

The auto power spectral densities (PSD) of $Y_s(f,k)$ and cross-psd between $Y_s(f,k), Y_r(f,k)$ computed as:

$$P_{YsYr}(f,k) = (1-\alpha)P_{YsYr}(f,k-1) + \alpha Y_s(f,k)Y_r^*(f,k) \quad (5)$$
$$P_{YrYr}(f,k) = (1-\alpha)P_{YrYr}(f,k-1) + \alpha Y_r(f,k)Y_r^*(f,k) \quad (6)$$

where α is a smoothing parameter, which in range of $\{0..1\}$.

The Wiener filter coefficients are calculated as:

$$H(f,k) = \frac{P_{YsYr}(f,k)}{P_{YrYr}(f,k)} \quad (7)$$

The GSC beamformer's output signal can be derived by:

$$Y(f,k) = Y_s(f,k) - Y_r(f,k) * H(f,k) \quad (8)$$

Due to the microphone mismatches, the inaccurate direction of arrival of useful speaker, the imprecise MA geometry, the remained speech leakage in the reference signal, the GSC beamforming often degraded, the speech quality and speech intelligibility decreased, and the resulting signal usually corrupted. In the ideal case, the reference signal contains only noise signals, and the adaptive filter is capable of saving clean speech while alleviating noisy components from the main signal. However, GSC beamformers are sensitive to speech leakage. The existing remaining speech in reference signal often decreases the satisfactory perceptual for the listener, the MA beamforming's evaluation and the effectiveness of the designed digital signal processing system. In the next section, the authors proposed a spectral mask, which based on the observed properties of MA to mitigate speech leakage for saving useful signals.

III. The proposed spectral mask based on priori phase - difference and the temporal $S\hat{N}R(f,k)$

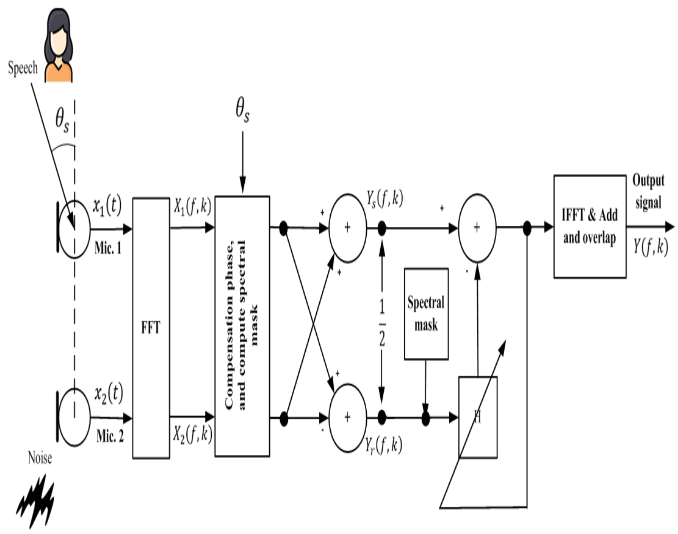

Fig. 5. The proposed method suppress the speech leakage, which often occurs in the reference signal.

In Fig. 5, due to the existence of speech leakage in the reference signal, so the implementation of GSC beamformer in the frequency domain usually corrupted, the speech quality, speech intelligibility and the satisfactory perceptual metric listener significantly affected. The author proposed an combination of the priori information of phase difference $\Theta(f,k)$ between two microphone signals and the formulation of the temporal signal-to-noise $S\hat{N}R(f,k)$ to form an effective spectral mask $auspm(f,k)$. The formulation can be defined as:

$$auspm(f,k) = \frac{1}{1+\beta\sin^2(\Theta(f,k))} \times \frac{1}{1+S\hat{N}R(f,k)} \quad (9)$$

The phase difference between two observerd MA signals yields as:

$$\Theta(f,k) = \arg(X_1(f,k)) - \arg(X_2(f,k)) - \omega\tau_0 cos(\theta_s) \quad (10)$$

with the constrained criteria $-\frac{\pi}{2} \le \Theta(f,k) \le \frac{\pi}{2}$.

The relation between microphone array signals is presented by the coherence $\Gamma_{X_1X_2}(f,k)$, which is defined as [24]:

$$\Gamma_{X_1X_2}(f,k) = \frac{P_{X_1X_2}(f,k)}{\sqrt{P_{X_1X_1}(f,k) \times P_{X_1X_1}(f,k)}} \quad (11)$$

where $P_{X_1X_2}(f,k)$ is cross PSD, $P_{X_1X_1}(f,k), P_{X_2X_2}(f,k)$ is auto PSD of microphone array signals $X_1(f,k), X_2(f,k)$.

$$P_{X_iX_j}(f,k) = \gamma P_{X_iX_j}(f,k-1) + (1-\gamma)X_i(f,k)X_j^*(f,k), i,j = 1,2 \quad (12)$$

with γ is a smoothing parameter, which in the range $0..1$.

In MA beamforming,the coherence of dual - microphone relatives to the coherence of speech component, noise and the temporal $S\hat{N}R(f,k)$ as the following way:

$$\Gamma_{X_1X_2}(f,k) = \frac{S\hat{N}R(f,k)}{1+S\hat{N}R(f,k)}\Gamma_s(f,\theta_s) + \frac{1}{1+S\hat{N}R(f,k)}\Gamma_n \quad (13)$$

where $\Gamma_s(f,\theta_s) = e^{j\Phi_s}$, $\Gamma_n = 1$ in the coherent noise field, and $\Gamma_n = \frac{\sin(\omega\tau_0)}{\omega\tau_0}$ in diffuse noise field.

If we denote $\frac{1}{1+S\hat{N}R(f,k)} = \rho(f,k)$, the equation (13) can be rewritten as:

$$\Gamma_{X_1X_2}(f,k) = (1-\rho(f,k))\Gamma_s(f,\theta_s) + \rho(f,k)\Gamma_n \quad (14)$$

And

$$\rho(f,k) = \frac{1-\Gamma_{X_1X_2}(f,k)}{\Gamma_s(f,\theta_s) - \Gamma_n} \quad (15)$$

Consequently, the author's suggested spectral mask $auspm(f,k)$ own the formulation:

$$auspm(f,k) = \frac{1}{1+\beta\sin^2(\Theta(f,k))} \times \frac{1-\Gamma_{X_1X_2}(f,k)}{\Gamma_s(f,\theta_s) - \Gamma_n} \quad (16)$$

As we known that, at the frame with the existence of speech component, the phase different often occurs. In the ideally recording scenario, the $\Theta(f,k) = 0$; but in practical situations, this parameter often $\le -\frac{\pi}{2}$ or $\ge \frac{\pi}{2}$. Therefore, in the frame with speech, with an appropriate value of β, the ratio $\frac{1}{1+\beta\sin^2(\Theta(f,k))} < 1$. With a combination with $\frac{1}{1+S\hat{N}R(f,k)}$, the spectral mask $auspm(f,k) < 1$ will block the remained speech component at the reference signal.

The modified of $Y_r(f,k)$ is derived as:

$$\hat{Y}_r(f,k) = auspm(f,k) * Y_r(f,k) \quad (17)$$

In the next section, the authors will perform a promising experiment to confirm the effectiveness of the above proposed method in a realistic recording environment.

IV. Experiments

In this section, the authors illustrated an experiment to verify the effectiveness of the proposed method (sp_GSC) in comparison with the conventional GSC beamformer. Dual - microphone systems were used, due to the compact size, easily integrated signal processing algorithm in most acoustic equipment and low computation. The scheme of demonstrated experiment was shown in Fig. 6.

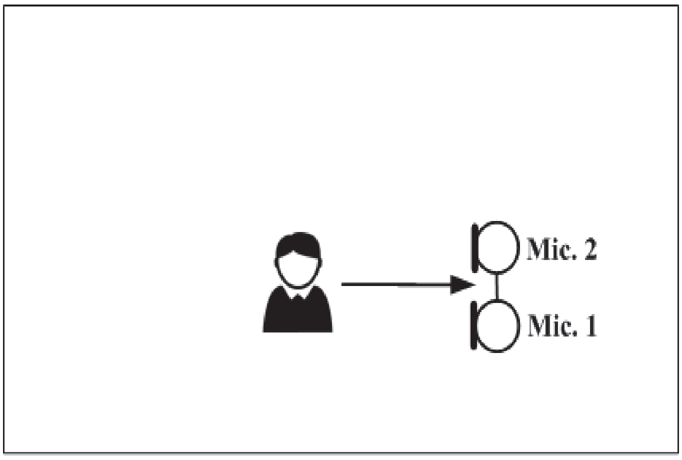

Fig. 6. The scheme of performed experiments.

The recording scenario in an anechoic room, diffuse noise field, a stand speaker at the distance $L = 2(m)$ relative to the axis of the dual - microphone system. The range between two microphones is $d = 5(cm)$, the direction of arrival of interest of a useful signal is $\theta_s = 90(deg)$. The surrounding environment is white noise. An objective measurement [25] is used for calculating the speech quality in the term of the signal-to-noise ratio (SNR).

For further signal processing, these necessary following parameters were set. The sampling rate is $F_s = 16kHz$, $NFFT = 512$, overlap 50%, $\alpha = 0.1$, $\gamma = 0.1$, $\beta = 5$ for computing the auto - cross power spectral densities. The

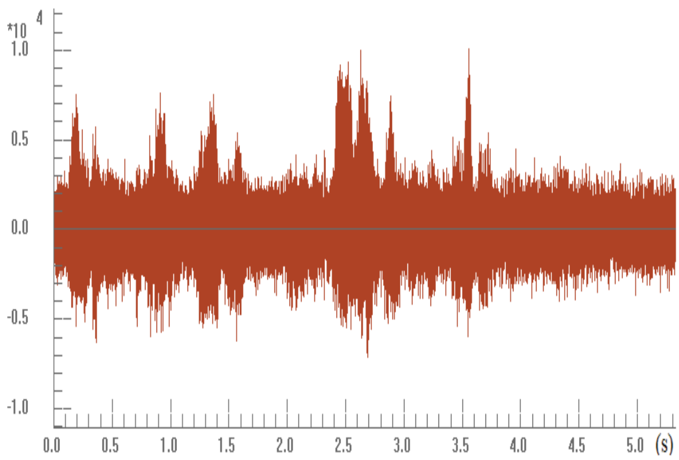

Fig. 7. The waveform of original microphone array signals.

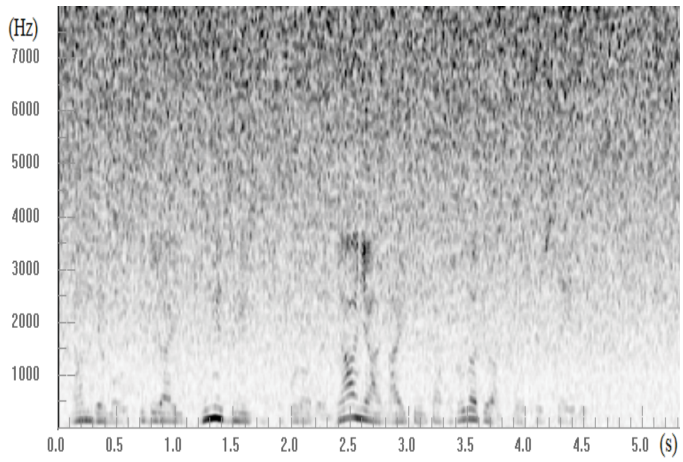

Fig. 8. The spectrum of observed microphone signals.

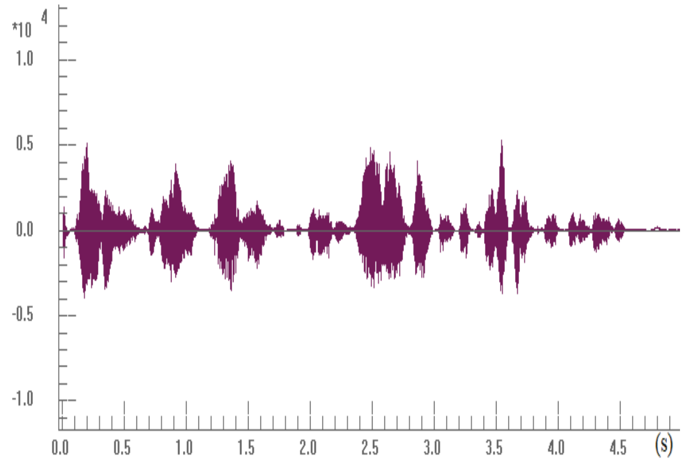

Fig. 9. The waveform of processed signal by GSC beamformer.

authors compare the SNR between the original microphone array signals, the processed by traditional GSC beamformer and the suggested method.

The original microphone array signal is shown in Fig. 7. By using the traditional GSC beamformer, the obtained signal was derived in Fig. 8.

Fig. 9 describes the resulting signal by applying the suggested technique.

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

Method Estimation	Microphone array signal	GSC beamformer	sp_GSC
NIST STNR	11.2	17.2	24.0
WADA SNR	8.3	17.9	20.4

From these above figures, we can see that s p_GSC removes the speech leakage, which often occurs in the reference signal and ensures achieving better performance. From Table I, the obtained SNR has been improved and increased from 2.5 to 6.8 (dB) and the speech distortion was reduced to 11.2 (dB), which was shown in Figure 10. The performance of GSC beamformer is often corrupted due to many reasons,

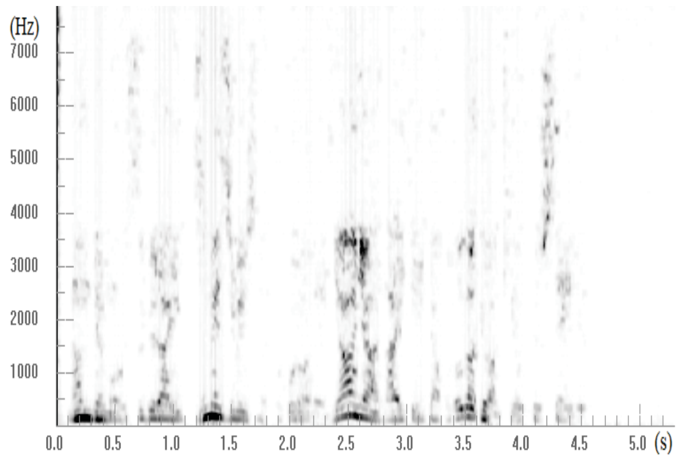

Fig. 10. The obtained spectrum of GSC beamformer's output signal.

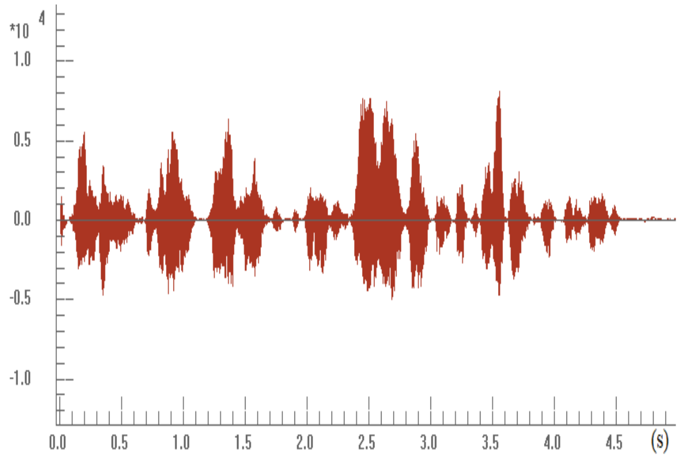

Fig. 11. The waveform of processed signal by sp_GSC.

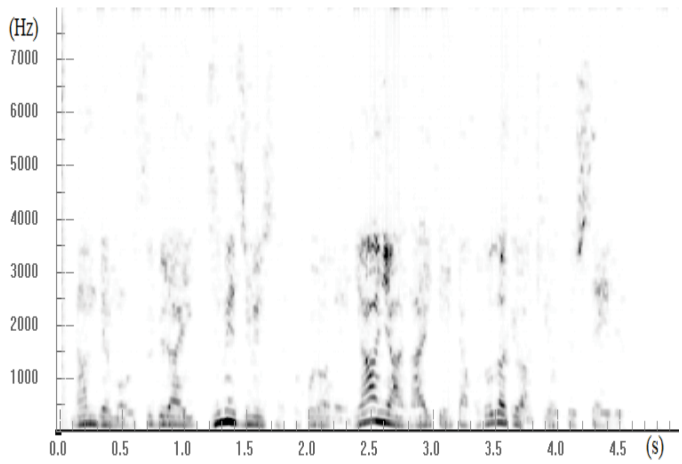

Fig. 12. The spectrum of processed signal by applying sp_GSC.

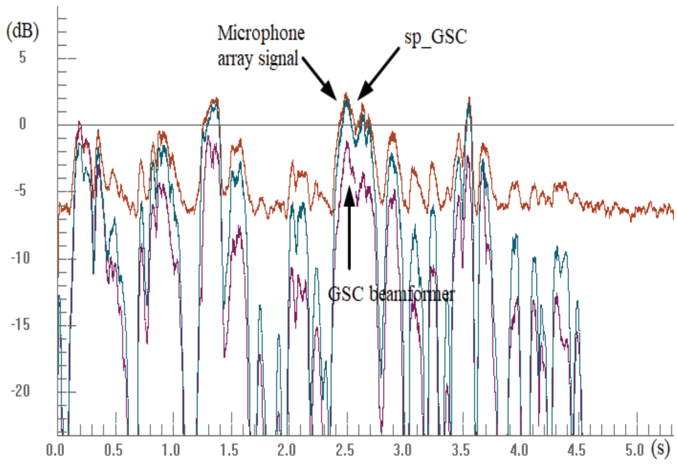

Fig. 13. The comparison of energy between the microphone array signals, processed by GSC beamformer and the proposed method.

such as: the different microphone sensitivities, the error of direction of arrival of useful signal, imprecise MA distribution, microphone mismatch and inaccurate frequency sampling. Consequently, the remaining speech component, which still exists in the reference signal, causes the degraded performance of GSC beamforming technique as the result of speech distortion and decreased the SNR at the output signal. The spectral mask, which uses the a priori information of phase and coherence, has the ability of removing remaining speech components and enhancing speech intelligibility, speech quality in comparison with the traditional GSC beamformer. In the case of using multiple microphones, the proposed spectral mask can be determined by calculating the averaged coherence and phase difference between pairs of microphones.

V. Conclusion

Speech enhancement, which plays an important role in various speech applications, is vital for enhancing perceptual listeners. Consequently, MA are promising assets for this purpose due to their superposition of spatial information, high directivity beampattern towards speech source while suppressing interference from other directions. A novel spectral mask was introduced for using spatial geometry, which includes phase and coherence, for forming an appropriate spectral mask to remove speech leakage of the reference signal in GSC beamformer. The numerical results show the advantage of the proposed method for enhancing speech intelligibility, speech quality in the term of signal-to-noise ratio (SNR) from 2.5 to 6.8 (dB) and speech distortion reduced to 11.2 (dB). The promising direction of the above approach is combination of priori information phase and coherence to configure an additive spectral mask, which applied to GSC beamformer's performance. The proposed method can be integrated into a system with multichannel microphone signals. In the future, the authors continue using other MA characteristics for improving the effectiveness of the described algorithm.

References

[1] J. Benesty, M. M.Sondhi, and Y. A. Huang, "Introduction to Speech Processing," Springer Handbook of Speech Processing. Springer, Berlin, Heidelberg, 2008: 1-4.

[2] M. Neumayer and T.Bretterklieber, "Bandwidth Based Design Methodology for Wiener Filters for Online Signal Denoising," 2020 IEEE International Instrumentation and Measurement Technology Conference (I2MTC), Dubrovnik, Croatia, 2020, pp. 1-6, doi: 10.1109/I2MTC43012.2020.9128772.

[3] S.Ruiz, T.Dietzen, T. Van Waterschoot and M.Moonen, "A comparison between overlap-save and weighted overlap-add filter banks for multi-channel Wiener filter based noise reduction," 2021 29th European Signal Processing Conference (EUSIPCO), Dublin, Ireland, 2021, pp. 336-340, doi: 10.23919/EUSIPCO54536.2021.9616352.

[4] Z.Chen, "Noise Reduction of Bird Calls Based on a Combination of Spectral Subtraction, Wiener Filtering, and Kalman Filtering," 2022 IEEE 5th International Conference on Automation, Electronics and Electrical Engineering (AUTEEE), Shenyang, China, 2022, pp. 512-517, doi: 10.1109/AUTEEE56487.2022.9994282.

[5] A.H.Rakshitha, S.Chikkamath and S. D.Desai, "Speech Enhancement Algorithms for Wind Noise Reduction," 2022 3rd International Conference for Emerging Technology (INCET), Belgaum, India, 2022, pp. 1-6, doi: 10.1109/INCET54531.2022.9824706.

[6] J.Liang and Q.Zeng, "Improved Spectral Subtraction Based on Second-Order Differential Array and Phase Spectrum Compensation," 2023 3rd International Symposium on Computer Technology and Information Science (ISCTIS), Chengdu, China, 2023, pp. 658-661, doi: 10.1109/ISCTIS58954.2023.10212993.

[7] J.Wu, G.Chong, W.Pang and L.Wang, "Speech Endpoint Detection Based on EMD and Improved Spectral Subtraction," 2023 5th International Conference on Natural Language Processing (ICNLP), Guangzhou, China, 2023, pp. 126-130, doi: 10.1109/ICNLP58431.2023.00029.

[8] G.Huang, I.Cohen, J.Benesty and J.Chen, "Kronecker Product Beamforming with Multiple Differential Microphone Arrays," 2020 IEEE 11th Sensor Array and Multichannel Signal Processing Workshop (SAM), Hangzhou, China, 2020, pp. 1-5, doi: 10.1109/SAM48682.2020.9104333.

[9] G. Huang, J. Benesty, and I. Cohen, "Robust and steerable kronecker product differential beamforming With rectangular microphone arrays," ICASSP 2020 - 2020 IEEE International Conference on Acoustics, Speech and Signal Processing (ICASSP), Barcelona, Spain, 2020, pp. 211-215, doi: 10.1109/ICASSP40776.2020.9052988.

[10] X.Zhao, X.Luo, G.Huang, J.Chen and J.Benesty, "Differential Beamforming with Null Constraints for Spherical Microphone Arrays," ICASSP 2024 - 2024 IEEE International Conference on Acoustics, Speech and Signal Processing (ICASSP), Seoul, Korea, Republic of, 2024, pp. 776-780, doi: 10.1109/ICASSP48485.2024.10446768.

[11] X.Luo, J.Jin, G.Huang, J.Chen and J.Benesty, "Design of Steerable Linear Differential Microphone Arrays With Omnidirectional and Bidirectional Sensors," IEEE Signal Processing Letters, vol. 30, pp. 463-467, 2023, doi: 10.1109/LSP.2023.3267969.

[12] X.Wang, G.Huang, I.Cohen, J.Benesty and J.Chen, "Robust Steerable Differential Beamformers with Null Constraints for Concentric Circular Microphone Arrays," ICASSP 2021 - 2021 IEEE International Conference on Acoustics, Speech and Signal Processing (ICASSP), Toronto, ON, Canada, 2021, pp. 4465-4469, doi: 10.1109/ICASSP39728.2021.9414119.

[13] G.Yu, Y.Qiu, N.Wang, "A Robust Wavenumber-Domain Superdirective Beamforming for Endfire Arrays," IEEE Transactions on Signal Processing, vol. 69, pp. 4890-4905, 2021, doi: 10.1109/TSP.2021.3105754.

[14] N.Shankar, G. S.Bhat, I. M. S.Panahi, "Real-time dual-channel speech enhancement by VAD assisted MVDR beamformer for hearing aid applications using smartphone," Proc 2020 42nd Annual International Conference of the IEEE Engineering in Medicine & Biology Society (EMBC), Montreal, QC, Canada, 2020, pp. 952-955, doi: 10.1109/EMBC44109.2020.9175212.

[15] D.Wang, C.Bao, "Multi-channel Speech Enhancement Based on the MVDR Beamformer and Postfilter," 2020 IEEE International Conference on Signal Processing, Communications and Computing (ICSPCC), Macau, China, 2020, pp. 1-5, doi: 10.1109/ICSPCC50002.2020.9259489.

[16] S.Li, Q.Liu, W.Wang, ”Generalized Sidelobe Canceller with Variable Step-Size Least Mean Square Algorithm Controlled by Signal-to-Noise Ratio,” 2022 5th International Conference on Data Science and Information Technology (DSIT), Shanghai, China, 2022, pp. 1-6, doi: 10.1109/DSIT55514.2022.9943855.

[17] J.Wang, F.Yang, J.Guo, J.Yang, ”Robust Adaptation Control for Generalized Sidelobe Canceller with Time-Varying Gaussian Source Model,” 2023 31st European Signal Processing Conference (EUSIPCO), Helsinki, Finland, 2023, pp. 16-20, doi: 10.23919/EUSIPCO58844.2023.10289801.

[18] F. M.Ribas, ”Minimum Variance Generalized Sidelobe Canceller and Eigenspace-based Generalized Sidelobe Canceller Beamformers Combined with Frost Postfilter for Medical Ultrasound Imaging,” 2021 IEEE UFFC Latin America Ultrasonics Symposium (LAUS), Gainesville, FL, USA, 2021, pp. 1-4, doi: 10.1109/LAUS53676.2021.9639164.

[19] J. Wang, F. Yang, J. Guo and J. Yang, ”Robust Adaptation Control for Generalized Sidelobe Canceller with Time-Varying Gaussian Source Model,” 2023 31st European Signal Processing Conference (EUSIPCO), Helsinki, Finland, 2023, pp. 16-20, doi: 10.23919/EUSIPCO58844.2023.10289801.

[20] S. M. Kim and H. K. Kim, ”Multi-Microphone Target Signal Enhancement Using Generalized Sidelobe Canceller Controlled by Phase Error Filter,” in IEEE Sensors Journal, vol. 16, no. 21, pp. 7566-7567, Nov.1, 2016, doi: 10.1109/JSEN.2016.2602375.

[21] W. Middelberg and S. Doclo, ”Comparison of Generalized Sidelobe Canceller Structures Incorporating External Microphones for Joint Noise and Interferer Reduction,” Speech Communication; 14th ITG Conference, online, 2021, pp. 1-5.

[22] S. M. Kim, ”Hearing Aid Speech Enhancement Using Phase Difference-Controlled Dual-Microphone Generalized Sidelobe Canceller,” in IEEE Access, vol. 7, pp. 130663-130671, 2019, doi: 10.1109/ACCESS.2019.2940047.

[23] A. B. A. Abdul Qayyum et al., ”Direction of Arrival Estimation through Noise Supression: A Novel Approach using GSC Beamforming and Room Acoustic Simulation,” 2019 IEEE International Conference on Signal Processing, Information, Communication & Systems (SPICSCON), Dhaka, Bangladesh, 2019, pp. 104-108, doi: 10.1109/SPICSCON48833.2019.9065151.

[24] N. Yousefian and P. C. Loizou, ”A Dual-Microphone Speech Enhancement Algorithm Based on the Coherence Function,” in IEEE Transactions on Audio, Speech, and Language Processing, vol. 20, no. 2, pp. 599-609, Feb. 2012, doi: 10.1109/TASL.2011.2162406.

[25] SNRVAD. [Online]. Available: https://labrosa.ee.columbia.edu/projects/snreval/