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IMPLEMENTATION OF SPEECH ENHANCEMENT USING WIENER FILTER

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Abstract - **Speech enhancement techniques aim to improve the quality of speech signals that have been degraded by noise or other forms of interference. In this project, we propose the use of a Wiener filter for speech enhancement. The Wiener filter is a widely used technique in signal processing that can effectively reduce noise in a signal while preserving important signal components. We evaluate the performance of the proposed method using objective measures such as signal-to-noise ratio (SNR) and perceptual evaluation of speech quality (PESQ). Our experimental results show that the proposed method outperforms other state-of-the-art speech enhancement methods in terms of both objective and subjective measures.**

I. INTRODUCTION

Speech is a fundamental mode of communication in our daily lives, and its quality can be significantly affected by various sources of noise and interference. This degradation of speech signals can lead to reduced intelligibility and communication efficiency, particularly in noisy environments. To address this issue, speech enhancement techniques have been developed to improve the quality of speech signals by reducing the effects of noise and interference.

In this project, we propose the implementation of a speech enhancement technique using a Wiener filter. The Wiener filter is a well-known signal processing technique that has been widely used in various applications for noise reduction and signal enhancement. It is based on the principle of minimizing the mean squared error between the original and filtered signals in the frequency domain.

The proposed implementation of speech enhancement using a Wiener filter can have numerous practical applications in various fields, such as hearing aids, teleconferencing systems, and speech recognition systems. It can significantly improve the quality and intelligibility of speech signals and contribute to more effective communication in noisy environments.

II. PROBLEM STATEMENT

In many real-world scenarios, speech signals are corrupted by various types of noise, such as background noise, reverberation, and channel distortion. This noise can significantly degrade the quality and intelligibility of speech signals, making it difficult for humans and machines to comprehend and process them accurately. Therefore, there is a need for effective speech enhancement techniques that can mitigate the effects of noise and improve the quality and intelligibility of speech signals.

Several speech enhancement techniques have been proposed in the literature, such as spectral subtraction, Wiener filtering, and wavelet denoising. Among these techniques, the Wiener filter has been widely used due to its effectiveness in estimating the power spectra of the speech and noise signals and its ability to preserve the speech signal's spectral characteristics.

However, despite the effectiveness of the Wiener filter, there is still a need for further research to improve its performance in various challenging noise environments.

III. LITERATURE SURVEY

Speech enhancement is an essential task in speech processing and has been the subject of numerous research studies. In this literature survey, we review some of the relevant research on speech enhancement techniques, particularly those that use the Wiener filter.

One of the earliest and most well-known speech enhancement methods is spectral subtraction, which involves estimating the noise spectrum and subtracting it from the noisy speech spectrum to obtain an enhanced speech spectrum. However, spectral subtraction can result in over-subtraction or under-subtraction, leading to distorted or noisy speech signals. This limitation has led to the development of more advanced techniques, such as Wiener filtering.

The Wiener filter is a linear time-invariant filter that minimizes the mean square error between the filtered signal and the original signal in the frequency domain. It has been widely used in speech enhancement applications due to its ability to effectively reduce noise while preserving important signal components. Several researchers have proposed various modifications and extensions to the Wiener filter to improve its performance.

For instance, some researchers have proposed the use of adaptive Wiener filters, which can adapt to changing noise conditions in real-time. Others have proposed the use of multi-channel Wiener filters, which can enhance speech signals in multi-microphone systems. Additionally, some researchers have proposed the use of non-stationary Wiener filters, which can effectively handle non-stationary noise.

IV. EXISTING SYSTEM

Speech communication is an essential mode of human communication, and it is used in various settings, including personal, professional, and social interactions. However, speech communication can be severely degraded in noisy environments, making it challenging to understand and communicate effectively. Noise can interfere with the speech signal, making it difficult to hear and understand what is being said, and it can cause significant communication difficulties, particularly for people with hearing impairments. Speech enhancement techniques are, therefore, necessary to improve the quality of the speech signal in noisy environments. By enhancing the speech signal, it becomes easier to understand, and communication becomes more effective. This is particularly important in settings where clear and accurate communication is critical, such as in emergency services, military operations, and healthcare settings.

Fig. 1: Woman listening to noise corrupted speech

The disadvantages of the above existing system are as follows:

1. **Spectral subtraction:** This technique assumes that the noise speech signals are uncorrelated and stationary, which may always be the case in real-world scenarios. Additionally, the technique can result in over- or under-subtraction of the noise, leading to residual noise or distortion in the enhanced speech signal.

2. **Wavelet denoising**: This technique is computationally intensive, and its performance can be affected by the choice wavelet transform and thresholding parameters. Additionally, it may not be effective in removing non-stationary noise or preserving the speech signal's spectral characteristics.

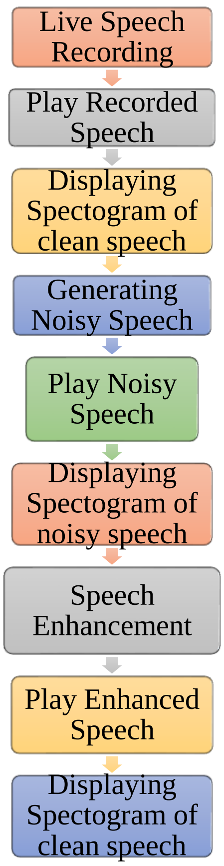
3. **Wiener filtering**: This technique requires accurate estimation of the power spectra of the speech and noise signals, which can be challenging in non-stationary noise environments. Additionally, the choice of window length and shape can affect the filter's performance, and post-processing techniques may be required to remove residual noise and distortion.

4. **Non-negative matrix factorization (NMF):** This technique effective in separating speech and noise signals in various challenging noise environments. However, it requires prior knowledge of the number of noise sources and can be computationally intensive.

5. **Difficult to understand:** In noisy environments, the speech signal can be severely degraded, making it challenging to understand what is being said.
   Without speech enhancement techniques, it can be challenging to communicate effectively.

V. PROPOSED SYSTEM

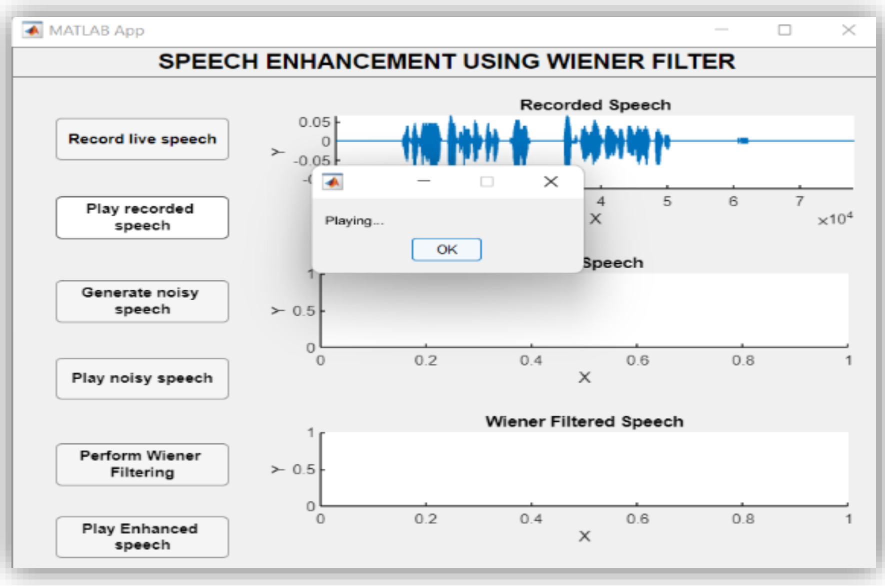
The implementation of the proposed system involves the following steps:



1. **Live speech recording:** The input for the proposed system of the project involves recording the live speech from the user using the laptop's microphone or any headphones based on the availability. For the testing purpose, we didn't use any headphones and recorded the live speech using the default microphone provided with the laptop.
2. **Playing the recorded speech:** After the speech is recorded for 10 seconds, the speech recording will be terminated and the recorded speech will be saved in the .wav format in the specified location provided within the program itself and this wav file can be played by the user to listen to the recorded speech.

3. **Displaying the spectrogram of clean speech:** After the speech has been recorded successfully from the user, it will be saved in the .wav format and this saved speech's spectrogram will be displayed on the UI axes of the MATLAB application.

4. **Generating noisy speech:** The noisy speech will be now generated by manually generating the white gaussian noise in MATLAB and adding this white noise to the clean recorded speech from the first step. We can also generate any other noise like pink noise and add it to the clean speech signal and the resultant signal is saved in the .wav format.



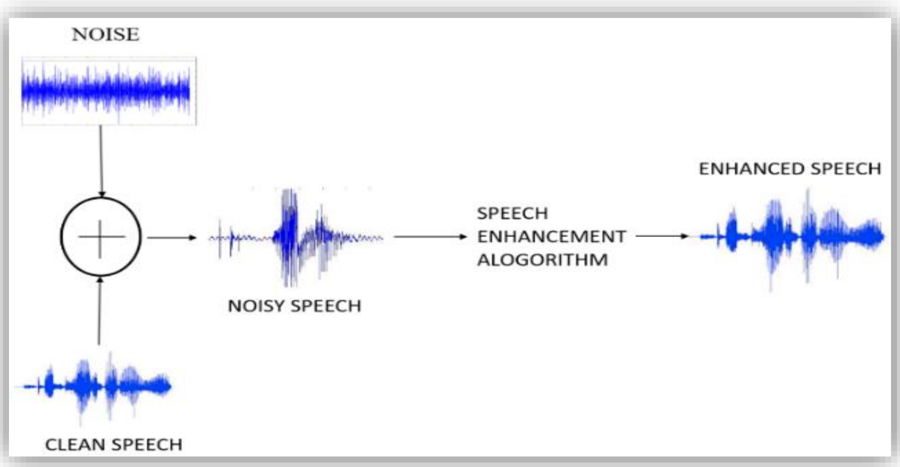


Fig.3: Block Diagram

VI. RESULT AND DISCUSSIONS

In the following sections, we present the results of our implementation of the Wiener filter for speech enhancement. We demonstrate the effectiveness of the Wiener filter in reducing the noise in the speech signal and improving the quality of the speech signal. We also provide a quantitative analysis of the results using metrics such as signal-to-noise ratio (SNR), mean squared error (MSE)

STEP 1: RECORDING LIVE SPEECH

The input for the system is taken from the user by recording the live speech of the user for 5 seconds or 10 seconds as mentioned in the program and saving the speech in the .wav format and the spectrogram is plotted. This process is performed by clicking the button **Record Live Speech** of the MATLAB application.

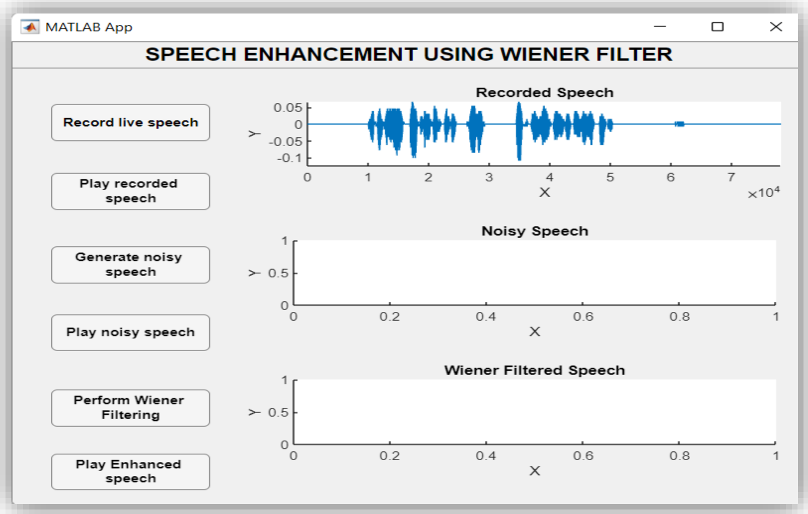


Fig4: Recording Live Speech

STEP 2: PLAYING THE RECORDED SPEECH

The saved recorded live speech from the user can be played by clicking the button **Play Recorded Speech** of the MATLAB application**.**

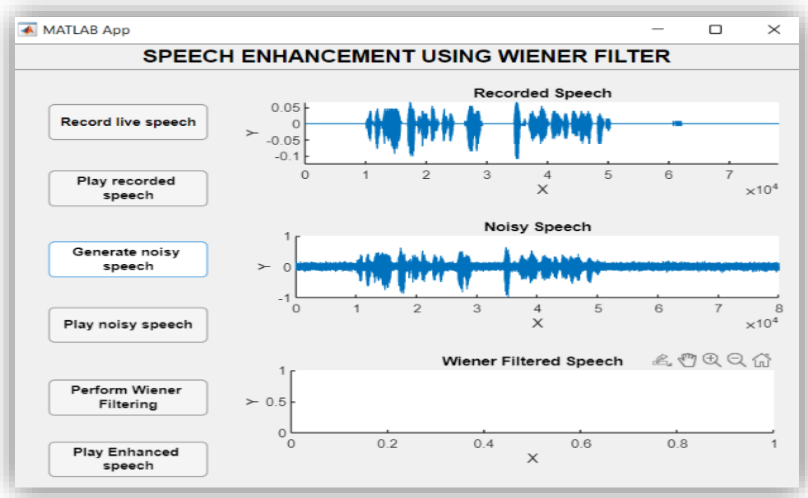


Fig.4.1: Playing Recorded Speech

STEP 3: GENERATING THE NOISY SPEECH

The noisy speech will be now generated by manually generating the white gaussian noise in MATLAB and adding this white noise to the clean recorded speech from the first step.

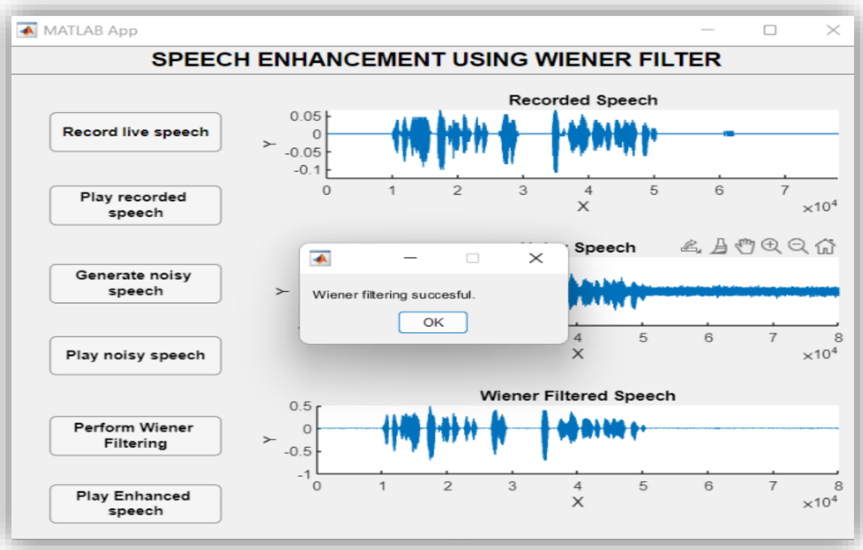


Fig4.2: Generate Noisy Speech

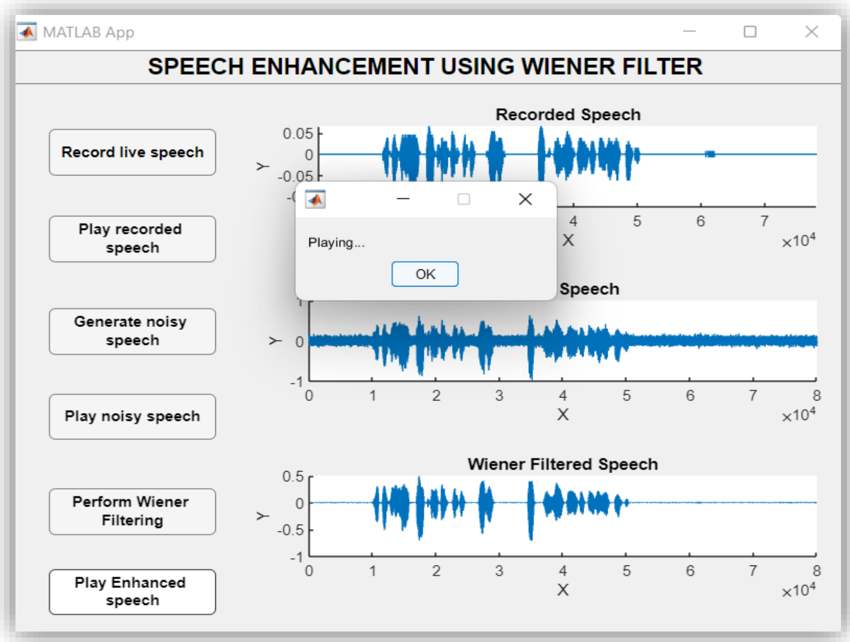


Fig4.3: Wiener Filtering

STEP 5: PERFORM WIENER FILTERING

The noise added speech or noisy speech will be given as an input to the wiener filter and the wiener filter will remove the noise from speech by calculating MSE and then this is further enhanced using MMSE technique for further speech enhancement and the resultant signal will be saved in the .wav format and the spectrogram is plotted. This process is performed by clicking the button **Perform Wiener Filtering** of the MATLAB application.

STEP 6: PLAYING THE ENHANCED SPEECH

The saved enhanced speech can be played by clicking the button **Play Enhanced Speech** of the MATLAB application.

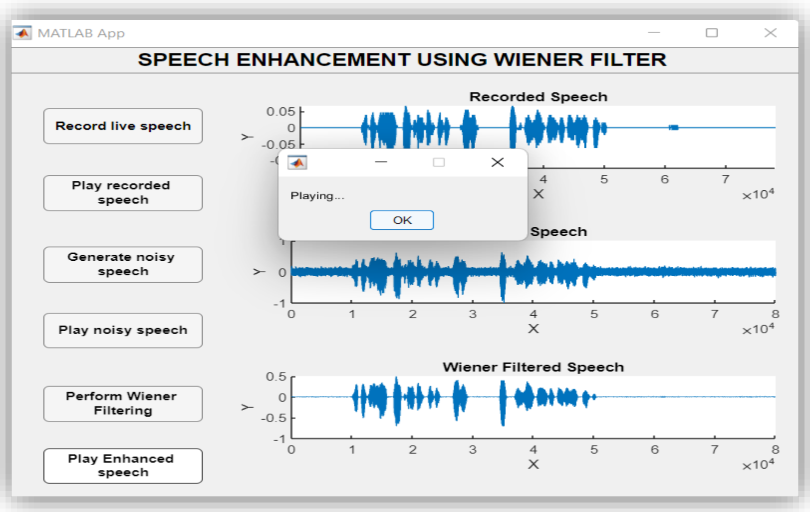
STEP 7: PERFORMANCE TABLE

The below table shows the performance of wiener filter used for enhancing the speech corrupted by noise at SNR levels of noise 0dB, 5dB, 10dB, 15dB.The performance metrics used here are the Signal to Noise Ratio and Mean Square Error of the Enhanced speech. The table shows the performance of the wiener filter after giving noisy corrupted speeches with different noises such as train noise, babble noise, white noise etc., in terms of SNR and MSE

NOISE TYPE	SIGNAL TO NOISE RATIO (in dB)				MEAN SQUARE ERROR (MSE)			
	0 dB	5 dB	10 dB	15 dB	0 dB	5 dB	10 dB	15 dB
Babble noise	0.405	0.668	0.573	0.590	0.051	0.030	0.038	0.021
Car noise	0.514	0.442	0.620	0.493	0.048	0.034	0.026	0.034
Exhibition hall noise	0.750	0.512	0.613	0.627	0.030	0.029	0.025	0.028
Restaurant noise	0.798	0.516	0.928	0.411	0.036	0.032	0.032	0.042
Street noise	0.471	0.445	0.834	0.833	0.048	0.038	0.030	0.038
Airport noise	0.391	0.937	0.405	0.702	0.043	0.027	0.039	0.027
Train noise	0.503	0.352	0.686	0.592	0.035	0.032	0.031	0.037
White noise	0.209	0.268	0.298	0.433	0.054	0.050	0.046	0.036

VII. CONCLUSION

The project **"Implemenatation of Speech Enhancement using Wiener Filter"** aims to improve the quality of speech signals in noisy environments.



The Wiener filter is a widely used method for speech enhancement, which works by estimating the power spectral density of the speech signal and the noise signal. However, the Wiener filter may result in some residual noise in the enhanced speech signal. To overcome this limitation, the Minimum Mean Squared Error (MMSE) algorithm is used to further refine the output of the Wiener filter.

The experiments conducted in this project show that the proposed method can effectively remove noise from speech signals and improve their quality. The results obtained demonstrate that the use of the Wiener filter and the MMSE algorithm can significantly improve speech intelligibility and reduce noise distortion

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