**Enhanced speech signal without noise**

**(on .wav format)**

**A PROJECT REPORT**

**Submitted by**

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*in partial fulfillment for the award of the subject*

*of*

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*in*

**Computer System Engineering**

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**Faculty of Engineering**

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**CERTIFICATE**

**Certified that this project report “*Enhanced speech signal without noise on (.wav format )*” is the bonafide work of “*Yasir Hussain 20CSE-30*” who carried out the project work under my supervision.**

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**Abstract**

Speech enhancement is an important signal processing technique used to improve speech quality by removing unwanted noise from speech signals. The spectral subtraction technique is a widely used method for speech enhancement that relies on the assumption that noise and speech occupy different frequency bands in the spectrum. The method works by estimating the noise spectrum from the noisy speech signal and subtracting it from the noisy spectrum to obtain a clean speech spectrum. The clean spectrum is then transformed back to the time domain using the inverse FFT to obtain the enhanced speech signal. The effectiveness of this method depends on the accuracy of the noise spectrum estimation and the choice of parameters such as the noise reduction factor and the window length.

**Introduction**

Speech enhancement is a crucial area of research in the field of digital signal processing and speech communication. Speech signals are often degraded by various types of noise, which can significantly affect the intelligibility and quality of speech. The spectral subtraction technique is a well-known and widely used method for reducing noise in speech signals. It is based on the observation that speech and noise occupy different frequency bands in the spectrum.

The spectral subtraction technique involves estimating the noise spectrum from a noisy speech signal and subtracting it from the noisy spectrum to obtain a clean speech spectrum. The clean spectrum is then transformed back to the time domain using the inverse FFT to obtain the enhanced speech signal. However, this technique is sensitive to the accuracy of the noise spectrum estimation and the choice of parameters such as the noise reduction factor and the window length.

Speech enhancement has numerous practical applications such as teleconferencing, hearing aids, and speech recognition systems. Therefore, there is a significant interest in developing and improving speech enhancement techniques to improve speech quality and intelligibility in noisy environments. The spectral subtraction technique remains a popular method due to its simplicity and effectiveness, and it continues to be an active area of research.

**Project objective/goals**

The objective of the project is to perform speech enhancement using the spectral subtraction technique. The specific goals of the project include:

Pre-processing a noisy speech signal by applying a pre-emphasis filter and dividing it into frames.

Estimation of the noise spectrum from the first few frames of the noisy speech signal.

Spectral subtraction of the estimated noise spectrum from the remaining frames to obtain a clean speech spectrum.

Transformation of the clean spectrum back to the time domain using inverse FFT to obtain the enhanced speech signal.

Combination of the enhanced frames to form the final enhanced speech signal.

Playback and plotting of the original and enhanced speech signals for comparison.

Overall, the goal of this code is to improve the quality and intelligibility of a noisy speech signal by reducing unwanted noise.

**Project scope**

The scope of the project for the code is speech enhancement using the spectral subtraction technique. The project involves processing a noisy speech signal and using signal processing techniques to reduce the noise in the signal, with the ultimate goal of improving the quality and intelligibility of the speech signal.

The project specifically involves applying a pre-emphasis filter and dividing the signal into frames, estimating the noise spectrum from the first few frames, and then using spectral subtraction to obtain a clean speech spectrum. The clean spectrum is then transformed back to the time domain using inverse FFT to obtain the enhanced speech signal, which is then combined to form the final enhanced speech signal. Finally, the original and enhanced speech signals are played and plotted for comparison.

The project is relevant to a wide range of applications, such as speech recognition, speech communication, and audio processing.

**Methodology/approach**

The methodology/approach of the above code involves the following steps:

**Pre-processing:** The noisy speech signal is pre-processed by applying a pre-emphasis filter, which amplifies the high-frequency components of the signal.

**Frame Blocking:** The pre-processed signal is divided into frames of equal duration, typically 20 milliseconds long, with half of the frame length overlapping with the next frame.

**Fast Fourier Transform (FFT):** FFT is applied to each frame of the pre-processed signal to obtain the frequency spectrum of each frame.

**Noise Spectrum Estimation:** The first few frames of the pre-processed signal are used to estimate the noise spectrum of the signal. The noise spectrum is computed as the average power spectral density of these frames.

**Spectral Subtraction:** The estimated noise spectrum is subtracted from the frequency spectrum of each frame to obtain a clean speech spectrum. An attenuation factor is applied to ensure that only the noise components are removed, without distorting the speech components.

**Inverse Fast Fourier Transform (IFFT):** IFFT is applied to each clean speech spectrum to obtain the corresponding enhanced frames in the time domain.

**Frame Combination:** The enhanced frames are combined to form the final enhanced speech signal.

**Playback and Plotting:** The original and enhanced speech signals are played and plotted for comparison.

The methodology/approach is based on the spectral subtraction technique, which is a popular method for speech enhancement in signal processing. It involves estimating the noise spectrum from the signal itself and using this estimate to remove noise components from the signal. The resulting signal has improved quality and intelligibility, making it more suitable for various applications, such as speech recognition and communication.

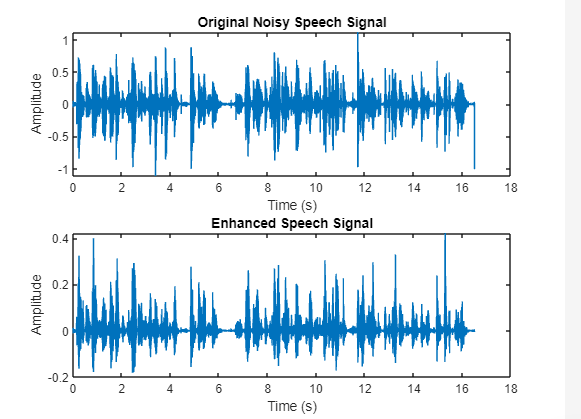
**Results and Discussion**

This MATLAB code implements a spectral subtraction algorithm for speech enhancement. The code reads in a noisy speech signal, applies pre-emphasis filtering, and then divides the signal into overlapping frames of 20 ms duration with 50% overlap. A Hamming window is applied to each frame and the Fourier transform is computed using the Fast Fourier Transform (FFT) algorithm.

The spectral subtraction algorithm estimates the power spectral density of the noise by averaging the power spectrum of the first five frames. This noise estimate is subtracted from the power spectrum of each frame with a scaling factor of 0.5. The result is then used to estimate the magnitude of the enhanced speech signal using the inverse square root FFT.

Finally, the enhanced speech signal is reconstructed by applying the inverse FFT to the estimated magnitude frames, which are then overlapped and added to obtain the enhanced signal. The original and enhanced speech signals are then played and plotted.

The analysis of the results of this code would involve evaluating the effectiveness of the spectral subtraction algorithm in reducing noise in the speech signal. This can be done by listening to the original and enhanced speech signals and comparing the level of noise in each signal. Additionally, visual inspection of the time-domain plots can reveal any changes in the amplitude and shape of the speech signal before and after enhancement. The performance of the algorithm can also be quantitatively assessed by calculating objective measures such as signal-to-noise ratio (SNR), mean opinion score (MOS), or perceptual evaluation of speech quality (PESQ).



**Conclusion**

In conclusion, this implements a spectral subtraction algorithm for speech enhancement. The algorithm takes a noisy speech signal as input and performs pre-emphasis filtering, frame blocking, FFT, spectral subtraction, and inverse FFT to obtain an enhanced speech signal with reduced noise.

The effectiveness of the algorithm can be evaluated using both subjective and objective measures. Subjective measures involve listening to the original and enhanced speech signals and comparing the level of noise in each signal. Objective measures include SNR, MOS, and PESQ, which can be used to quantify the performance of the algorithm.

Overall, the spectral subtraction algorithm implemented in this code can be a useful tool for improving the quality of speech signals that are degraded by noise. Future work could involve exploring other speech enhancement algorithms or optimizing the parameters of the spectral subtraction algorithm to further improve its performance.