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The Application of a Finite Impulse Response Low-Pass Filter for Noise Reduction in Voice Signals

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ABSTRACT

This study investigates the effect of noise on human voice signals and noise reduction efforts using digital signal processing techniques. The objective of this research is to analyze and compare the frequency characteristics and clarity of the original voice signal with those of noise-contaminated signal after undergoing a filtering process in MATLAB. The research methodology includes voice recording, superposition of the voice signal with noise, and filtering using a 50th-order Finite Impulse Response (FIR) low-pass filter with a cutoff frequency of 500 Hz implemented in MATLAB. The analysis is conducted in the frequency domain using the Fast Fourier Transform (FFT) and in the time domain through waveform observation. The results indicate that noise introduces high-frequency components and irregular amplitude fluctuations. After filtering, the high-frequency components are effectively attenuated, resulting in a smoother and more stable signal while preserving the primary characteristics of the human voice. These findings demonstrate that the FIR low-pass filter is effective in improving the quality of human voice signals.

1. INTRODUCTION

The rapid advancement of science is closely aligned with the continuous development of technology. In the era of digitalization, almost all human activities are associated with digital technology; therefore, it is essential to study and understand digital engineering principles (Cindy et al., 2025). One of the fundamental disciplines in digital engineering is digital signal processing (DSP). DSP is a method used to analyze, modify, and synthesize signals that have been converted into digital form. These signals may include audio, image, video, or other data originating from the real world. The signal conversion process is performed by transforming analog signals into digital signals through sampling and quantization, enabling computers to process the data effectively (Hikmah, 2025). One important application of digital signal processing is in audio signal processing, where voice signals can be digitally modified using audio filtering techniques.

Sound is a wave that propagates through solid objects as well as through air. In solids, sound travels faster because the constituent particles are more closely packed than in fluids. As sound propagates, it produces periodic variations in air pressure, and these pressure changes cause vibrations of the eardrum (Syahputri & Sari, 2025). Voice signals originate from human speech, with a frequency range of approximately 0 Hz to 5 kHz. These signals tend to vary slowly over short time durations of about 20-40 ms, during which the voice signal can be considered stationary. A voice signal is a continuous acoustic wave with a fundamental frequency typically ranging from 100 to 400 Hz, within which harmonic variations of the fundamental frequency may occur (Hartono, Suyuti, et al., 2023). Voice signals are highly susceptible to noise, which can be defined as unwanted disturbance perceived by the audio system (Sugiantoro, 2022). Noise often degrades audio quality; therefore, a system capable of reducing or eliminating noise in voice signals is required (Septiawan et al., 2024).

Various filtering methods have been extensively investigated, including Infinite Impulse Response (IIR) Butterworth filters (Subandrio et al., 2023) (Sujiwa & Maniani, 2025), Fast Fourier Transform (FFT) based approaches (Septiawan et al., 2024), and Least Mean Square (LMS) adaptive filters (Untsa et al., 2023); however, their effectiveness depends on the type of noise, signal characteristics, and filter design parameters. One software tool commonly used to support computational processing is MATLAB (Hartono, Husain, et al., 2023) (Sujiwa et al., 2025). This study aims to analyze and compare the frequency characteristics and clarity of the original voice signal with those of the signal after undergoing a filtering process in MATLAB.

2. METHOD

The research techniques employed in this study are explained in this section. From data collection to signal processing and analysis, the approach is described in methodical steps. The study utilizes a MATLAB-implemented low-pass filter to reduce and add noise to human voice signals. Every step is made to make it possible to see and assess how filtering affects loud voice signals.

2.1. Research Method

The recording and signal processing phases of this study are supported by both hardware and software. The tools used are tailored to the requirements of digital signal processing and voice signal recording. The primary instrument for signal processing and simulation in this study is MATLAB R2022a. Documentation and report preparation are done with Microsoft Office 2007, and all software activities are supported by Windows 11. A laptop to run MATLAB and a smartphone with speech recording capabilities make up the hardware. After that, the voice data recording is moved to the laptop for additional processing.

To simulate realistic noise conditions, additive white Gaussian noise (AWGN) was applied to the original voice signal using MATLAB. The noise signal was generated with zero mean and unit variance, then scaled to 10% of the maximum amplitude of the original voice signal. The noisy signal was obtained by linearly combining 90% of the original voice signal and 10% of the noise signal, as expressed in Equation (1).

$$\boldsymbol{mix = 0.9_{y1} + 0.1_{y2}} \quad \textbf{(1)}$$

To prevent signal clipping and ensure numerical stability during processing, the mixed signal was normalized by dividing it by its maximum absolute amplitude. All signals (original, noisy, and filtered) were processed at a sampling frequency of 44,100 Hz to maintain consistency and reproducibility.

2.2. Research Procedure

The research process consists of a number of primary steps that are organized methodically. These actions are intended to accomplish the research's goals. Using a recording device with a predetermined sample frequency, human voice sounds are recorded using a smartphone at a standardized sampling frequency (f_s) of 44,100 Hz. The original voice signal is represented by the recorded signal, which is saved digitally for processing. Adding noise to the voice signal that has been recorded is the second stage. MATLAB is used to create noise in order to replicate disruptions that frequently arise during actual recording situations. The noisy speech signal generated by this method will be fed into the filtering stage. Using a low-pass filter to filter the noisy voice signal is the last stage. The filter is used to preserve the primary features of the human speech signal while lowering high-frequency noise components.

2.3. Data Analysis

The original voice signal, the noisy voice signal, and the filtered voice signal are compared to analyze the data. Time-domain graphs that are automatically created when the MATLAB program is run are used to visually compare the data. The goal of this analysis is to determine how well the low-pass filter reduces noise and enhances the recorded voice signal's quality.

3. RESULTS AND DISCUSSION

This study begins with the process of combining the original audio signal with a noise signal. This stage aims to generate a signal that represents realistic disturbance conditions, allowing the noise characteristics and the effectiveness of the filtering method in reducing noise to be analyzed. Prior to mixing, both the original signal and the noise are processed to have the same sampling rate and data length. This standardization is essential to prevent waveform distortion due to data mismatch and to ensure that the two signals can be synchronously combined.

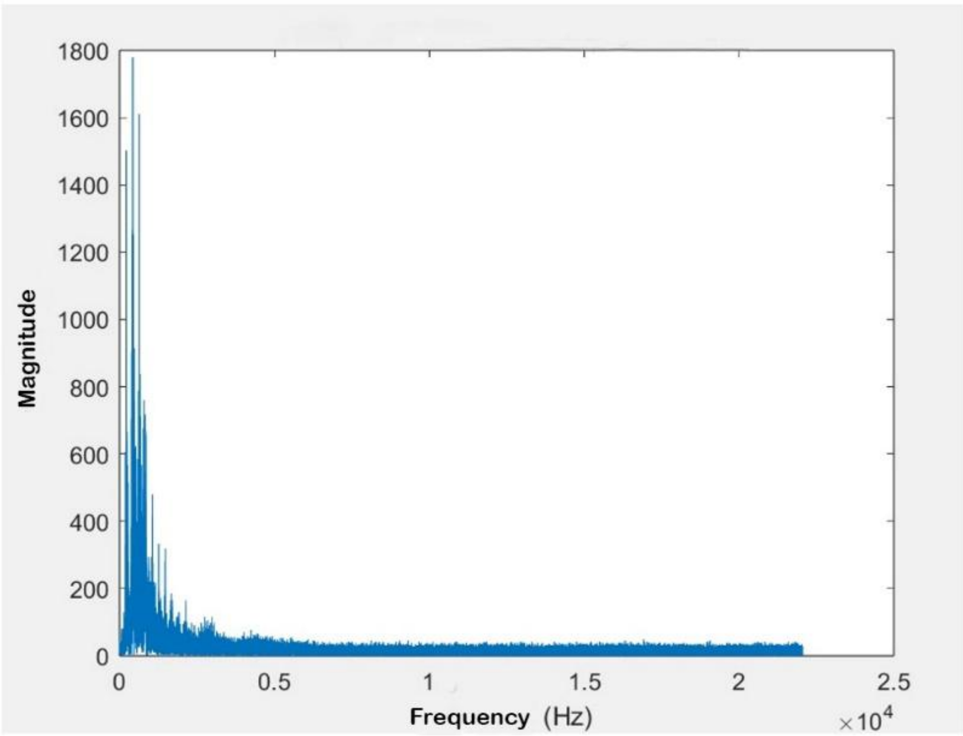

Figure 1. Frequency spectrum of the combined signal

In Equation (1) indicates that the final signal is composed of 90% original voice and 10% noise, such that the noise has sufficient magnitude to affect the signal without completely obscuring the structure of the human voice. The mixed signal is then normalized and saved as a combined audio file. The influence of noise on the frequency structure of the signal is analyzed using the FFT, as shown in Figure 1, which presents the frequency spectrum of the combined signal.

The frequency spectrum in Figure 1 illustrates the distribution of signal energy after the original voice signal is mixed with noise. It can be observed that most of the energy remains concentrated at low frequencies, particularly in the 0-500 Hz range, which corresponds to the primary region of human vocal components (Dewi et al., 2021). The high energy magnitude in this band indicates that the original voice signal still constitutes the dominant components of the overall mixed signal. However, additional energy with lower magnitude and irregular patterns appears at frequencies above 500 Hz up to approximately 2000 Hz. This random pattern is a characteristic feature of noise, which lacks the harmonic structure inherent to human speech (Siahaan & Darianto, 2020). Although the energy gradually decreases with increasing frequency, the presence of small fluctuations spread across the high-frequency region confirms that noise affects this portion of the spectrum. This finding indicates that the addition of noise alters the signal characteristics, particularly at mid and high frequency ranges. Therefore, the next step is to examine the signal behavior in the time domain to illustrate the temporal impact of noise and the extent of disturbance introduced into the original audio signal. The application of a low-pass filter is required to suppress these interference components while preserving the essential information contained in the low-frequency range.

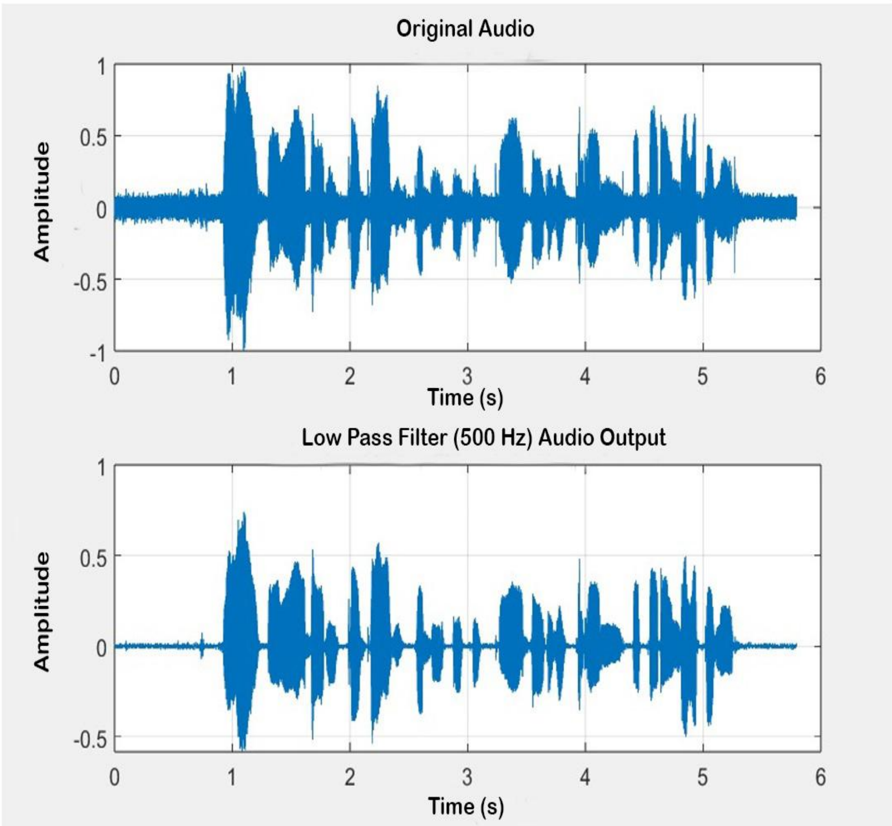

Figure 2. Comparison of the original audio signal and the signal after applying a 500 Hz Low-Pass Filter

Based on Figure 2, a comparison between the original audio signal and the signal after the application of a low-pass filter is presented. The filter employed in this study is a 50th-order FIR filter with a cutoff frequency of 500 Hz. The selection of this cutoff frequency is based on the characteristics of human speech, whose fundamental components lie within the low-frequency range (Dewi et al., 2021). With this cutoff, the filter is designed to pass signals below 500 Hz while attenuating higher-frequency components that are typically associated with noise. The 50th-order FIR filter is chosen because it provides a sufficiently sharp frequency response while maintaining system stability and avoiding phase distortion. This combination of parameters allows the filter to operate effectively in reducing noise while preserving the quality of the vocal signal.

The time-domain signal plot shows changes in the audio waveform structure, where the original signal exhibits a more complex waveform characterized by rapid and sharp amplitude fluctuations. This pattern reflects the presence of high-frequency components previously identified in the frequency spectrum analysis. These high-frequency components are characteristic of random noise, resulting in a rougher and less stable signal appearance in the time domain. After the application of the low-pass filter, the signal waveform becomes significantly smoother and more stable. The previously sharp amplitude spikes are substantially reduced, indicating that the filter effectively suppresses high-frequency components. As a result, the signal more closely resembles the characteristics of human speech, whose energy is predominantly concentrated at frequencies below 500 Hz. Despite the attenuation of noise components, the main structure of the vocal signal, such as speech rhythm and amplitude contour, is preserved. This demonstrates that the filtering process does not degrade the essential information contained in the original signal.

Based on the analysis of the frequency spectrum, time-domain waveforms, and evaluation of the filter parameters, it can be concluded that the application of a 50th-order FIR low-pass filter with a cutoff frequency of 500 Hz successfully improves audio signal quality. The filter effectively suppresses high-frequency components that serve as sources of noise while preserving the main structure of the vocal signal located in the low-frequency range. This is clearly evidenced by the transformation of the waveform from one with sharp fluctuations to a smoother and more stable from after the filtering process. Consequently, noise reduction is not only effective in the frequency domain but also results in improved signal characteristics in the time domain. These findings indicate that the applied filter design is well-suited to the characteristics of human voice signals and can be effectively implemented for audio processing applications.

Although various noise reduction techniques such as IIR filters, adaptive LMS filters, and FFT-based methods have been widely studied, many of these approaches require complex parameter tuning or higher computational costs. The novelty of this study lies in demonstrating that a properly designed FIR low-pass filter with simple and well-justified parameters can effectively reduce noise in voice signals while maintaining signal integrity.

This work provides a clear and systematic comparison of signal behavior in both time and frequency domains, emphasizing simplicity, reproducibility, and educational value. The proposed approach is particularly suitable for low-complexity audio processing applications and digital signal processing learning environments.

4. CONCLUSION

This study demonstrates that a 50th-order FIR low-pass filter with a cutoff frequency of 500 Hz is effective for reducing noise in human voice signals. Both qualitative and quantitative analyses confirm that high-frequency noise components are significantly attenuated while essential low-frequency vocal characteristics are preserved. The improvement in SNR and the low MSE values indicate that the filtering process enhances signal quality without introducing substantial distortion.

From a practical perspective, the proposed method is suitable for implementation in voice communication systems, educational digital signal processing laboratories, and low-power embedded audio applications. Additionally, it can serve as a preprocessing stage for speech analysis and recognition systems. Although this study focuses on a single filter configuration, future work may explore adaptive filtering techniques, parameter optimization, and real-time implementation to further enhance performance.

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