# ASSIGNMENT 01 - VOICE DENOISING USING FIR/IIR FILTERING TECHNIQUES IN MATLAB


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# INTRODUCTION

This report contains the process for denoising a voice signal by applying a digital bandpass filter. In day today life, voice signals are usually corrupted by different noises during transmission or acquisition. That noise can reduce the quality of the voice and also can affect for the intelligibility. The objective of this assignment is to design and implement a suitable filter to reduce unwanted noise while preserving the essential characteristics of the voice signal.

# METHODOLOGY

**Signal Acquisition and Noise Analysis**
I have loaded the noisy voice recording to MALAB. We can use time domain and frequency domain plots to get idea about the characteristics of voice signal and noise. The frequency spectrum is used to show the presence of significant energy outside the typical human speech frequency range, indicating noise.

**Filter Design & Parameters**
I have tried several filters like lowpass and highpass filters, but after testing, I chose Butterworth bandpass filter (IIR filtering). It gives good passband response and smooth roll off.

- Filter type: Bandpass
- Order: $8^{th}$ (tried using several order values, when I was decreasing the order, it was filtered less effectively, so for good filtering after testing I chose the 8).
- Passband frequencies: 1100Hz to 3000Hz (Typically for human voice the range was 300-3400Hz, but when I tried that range and got the noise of fan, then I tested some ranges and analyzed the frequency graphs and finally selected this range.)
- Sampling frequency: The filter design was normalized to the Nyquist frequency (Fs/2) of the input signal to ensure correct frequency scaling.

Using the 'filtfilt' function, I could applied the designed filter to the noisy recording. This function can implement zero phase digital filtering by processing the input data in forward and reversed directions. After the filtering I got the time domain and frequency domain waveforms and spectrum of the filtered signal. To quantify the improvement of the filtered voice audio, SNR (Signal-to-Noise-Ratio) was calculated. The higher SNR values are better.

# RESULTS

## Plots

The time domain plots that the filtered voice audio has reduced amplitude of some fluctuations. It has appeared smoother compared to the noisy audio. The high frequency erratic variations in the noise were attenuated in the filtered output that indicates a good noise suppression.

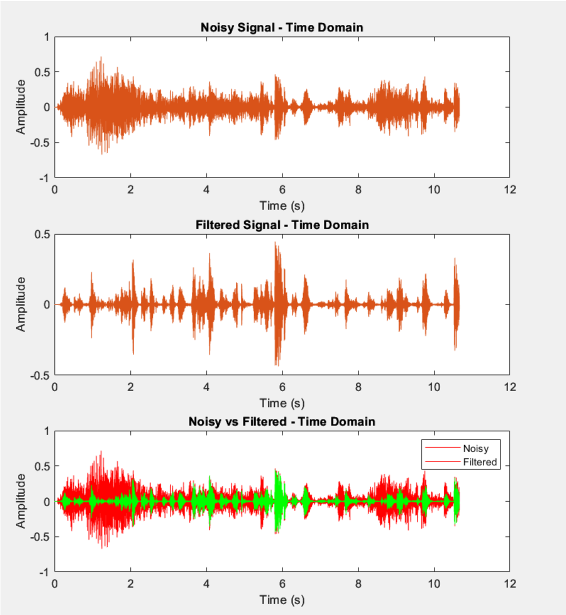


Figure 01: Time domain plots

In frequency spectrum plot, we can see the effectiveness of the bandpass filter. The noisy audio's spectrum shows significant energy across through wide range of frequencies (including outside the 1100-3000Hz). After filtering, we can see the energy outside of this range was attenuated. (Sharp drop after 3000 Hz and reduction of noise around 45000 Hz if present in the original signal). The main speech components within the 1100-3000 Hz range are preserved, leading to a much cleaner spectrum.

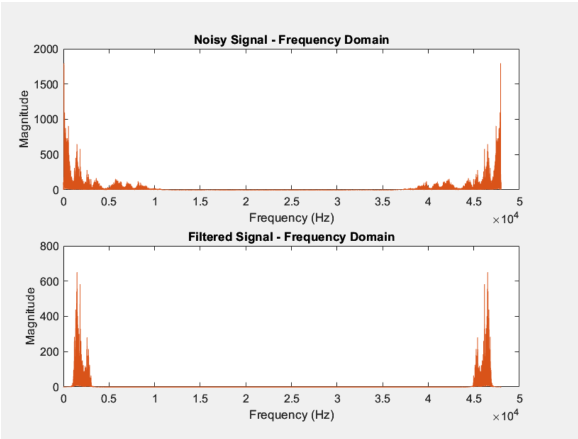


Figure 02: Frequency domain plots

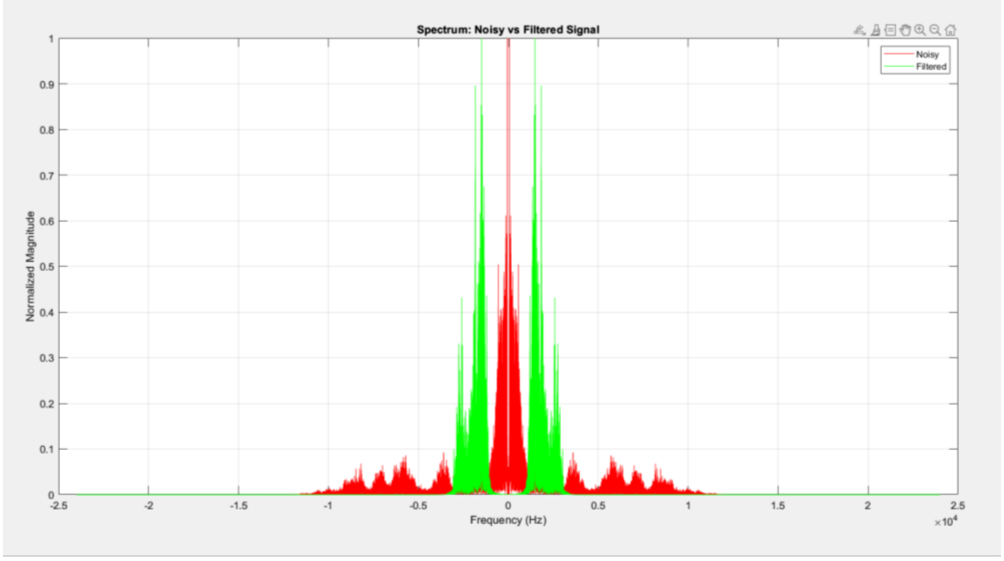


Figure 03: Spectrum plot

**SNR values**
The SNR calculation provides a quantitative measure of the improvement.

· SNR Before Filtering: -21.97dB
· SNR After Filtering: 1.01dB

The increase in SNR after filtering confirms that the noise power has been significantly reduced relative to the signal power, resulting in a cleaner audio output.

## CONCLUSION

**Filtering effectiveness**
The implemented 8th-order Butterworth bandpass filter successfully denoised the voice signal. Both visual inspection of the time and frequency domain plots and the quantitative SNR improvement confirm the effectiveness of the filtering process. The filtered signal retains the essential speech components while significantly reducing unwanted noise, thereby improving audio quality and potentially intelligibility.

**Future improvements**
As future improvements, we can use adaptive filtering techniques like LMS or RLS. They can dynamically adjust to changing noise characteristics. It will offer better performance. We can use More Sophisticated Noise Reduction Algorithms beyond traditional filters like spectral subtraction, wiener filtering or machine leaning based approaches. We can integrate psychoacoustic models into the filtering process for removing noise components, potentially leading to a more natural-sounding output at higher noise reduction levels.