VOICE DENOISING USING FIR/IIR FILTERING TECHNIQUES IN MATLAB

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1. INTRODUCTION

Real-world audio recordings are usually contaminated with background noise from various sources such as fans, traffic, or ordinary conversations. In speech-processing applications and communication systems, such noises degrade the quality and intelligibility of voice signals. The goal in this exercise is to capture a noisy voice recording (sound clip of violin with a noise of a fan), examine its spectral content, and implement the suitable digital filter (FIR or IIR) in MATLAB to remove unwanted noise components. Also, the primary goal is to enhance the signal-to-noise ratio (SNR) and preserve speech intelligibility.

2. METHODOLOGY

Audio Acquisition

A 10–15 second voice sample was recorded in a noisy environment with my mobile phone. The sampled file, noisy_recording.wav, had significant background noise, which was primarily of low-frequency sources such as a fan and some environmental hum. The audio was converted to mono and normalized for equal processing.

Noise Analysis

The frequency spectrum and audio waveform were observed using MATLAB. It was identified from the FFT analysis that the dominant noise energy was below 700 Hz. The voice content was identified to be in the range of 800 Hz to 3200 Hz, which overlapped with the normal human speech frequency range.

Filter Design

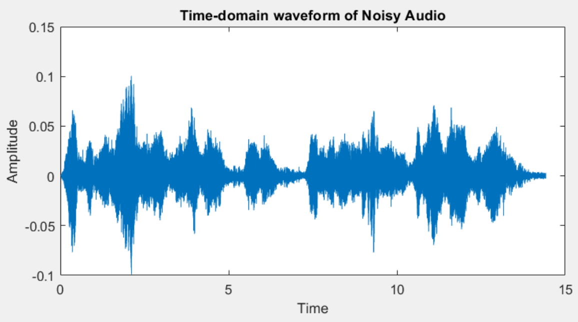
To effectively mask the noise and maintain speech information, a bandpass Butterworth filter was applied using MATLAB's built-in butter() function. The main parameters selected were:

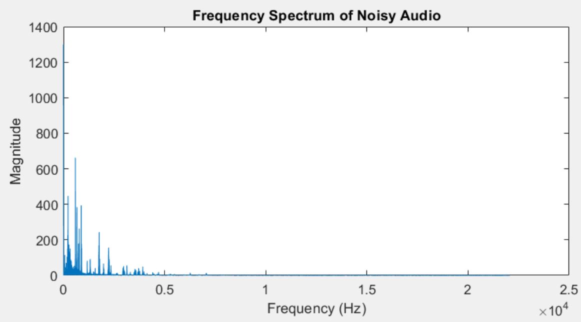
- **Filter Type:** Bandpass
- **Order:** 4
- **Lower Cutoff Frequency:** 800 Hz
- **Upper Cutoff Frequency:** 3200 Hz
- **Sampling Rate:** As in the original audio (Fs)

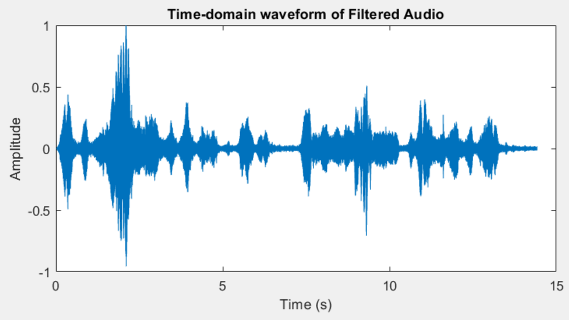
The filter was applied with the filtfilt() function to avoid phase distortion.

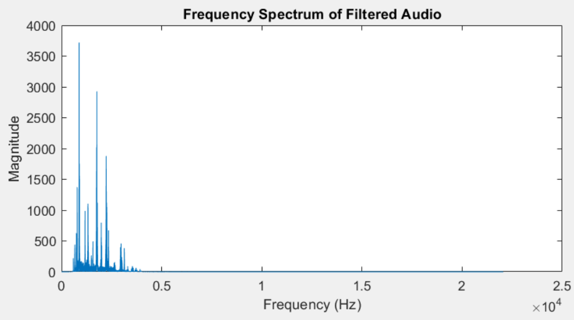
3. RESULTS

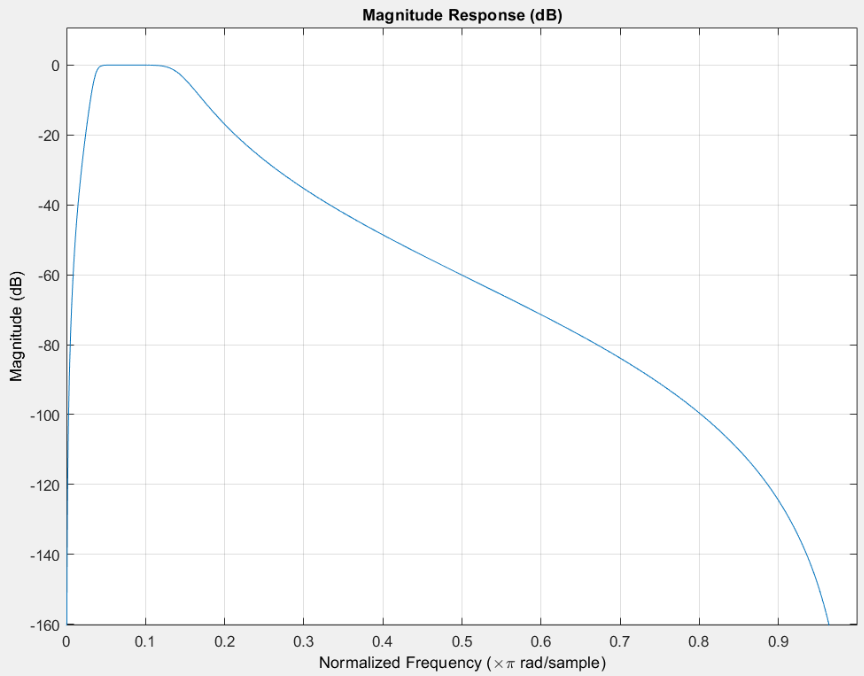
Time and Frequency Domain Plots











SNR Evaluation

SNR values were computed before and after filtering as below.

- **SNR Before Filtering** -13 dB
- **SNR After Filtering** -9.16 dB

Therefore, we can see a better improved SNR value after the filtering process. It means that the noise is reduced by this process.

Subjective Audio Quality

Upon listening to the filtered audio, the voice appeared clearer, with a notable reduction of background noise. Some minor loss of low vocal frequencies was observed due to the elevated lower cutoff, but intelligibility remained high.

4. CONCLUSION

The bandpass filter implemented and used in MATLAB effectively eliminated background noise in the noisy audio recording. SNR improvement from -13dB to -9.16 dB confirms the efficacy of this method. For future improvements, adaptive filtering methods like spectral subtraction may be investigated. Dynamically adjusting filter parameters according to real-time spectral analysis could also improve performance in changing environments.