Bonus Filter Design Project Report

Course: CSE 3313 - 001 Spring 2025

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1. Introduction

In this project, a Butterworth lowpass filter was designed and implemented to remove high-frequency noise from a noisy audio sample. The goal was to preserve speech frequencies (approximately 100 Hz to 2 kHz) while attenuating noise between 2.5 kHz and 5.5 kHz. MATLAB was utilized for signal analysis, filter design, audio processing, and evaluation.

2. Methodology

Step 1: Frequency Analysis

- The noisy audio file 'noisyaudio-1.wav' was read into MATLAB.

- DFT of the audio was calculated and plotted.

Step 2: Filter Design

- Passband: 2000 Hz

- Stopband: 2500 Hz

- Passband Ripple: 1 dB

- Stopband Attenuation: 30 dB

- Filter Order: 14

- Normalized Cutoff Frequency: 0.3779

Step 3: Filter Implementation

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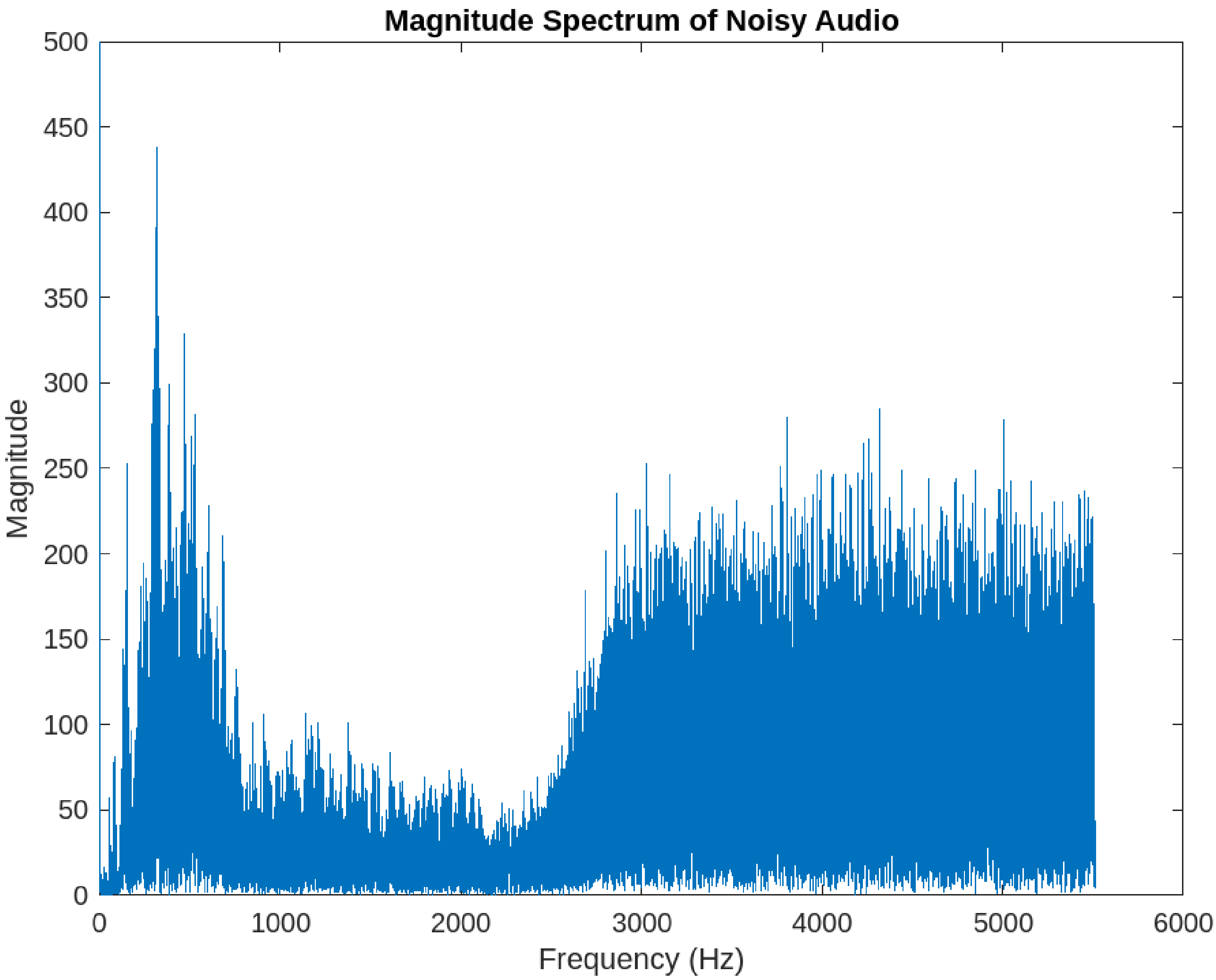
- The filter was applied to the noisy audio.

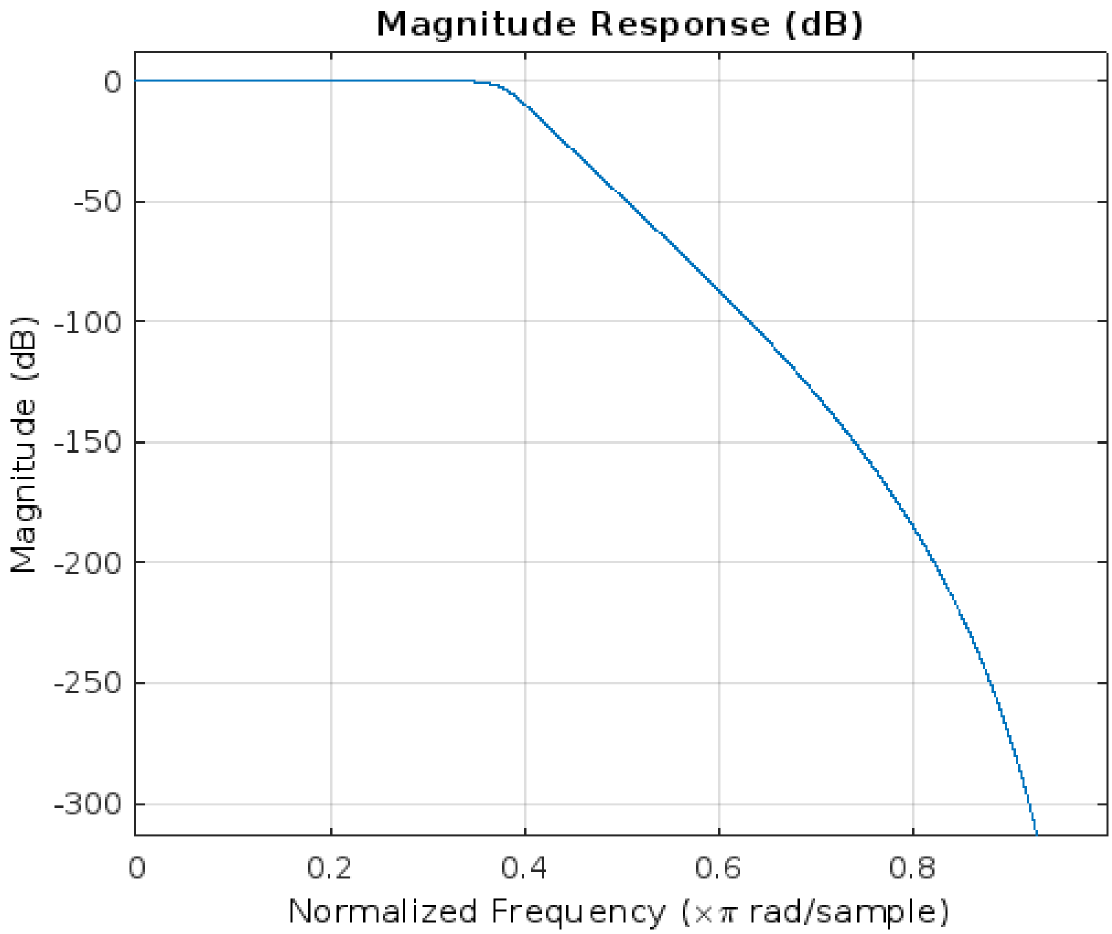
- Filtered output saved as 'filteredaudio.wav'.

- DFT was recalculated and compared.

DFT of Noisy Audio



Magnitude Response of Designed Butterworth Filter



3. Results

The Butterworth lowpass filter significantly attenuated the high-frequency noise while preserving important speech frequencies. Listening tests confirmed a noticeable improvement in audio clarity.

DFT of Filtered Audio

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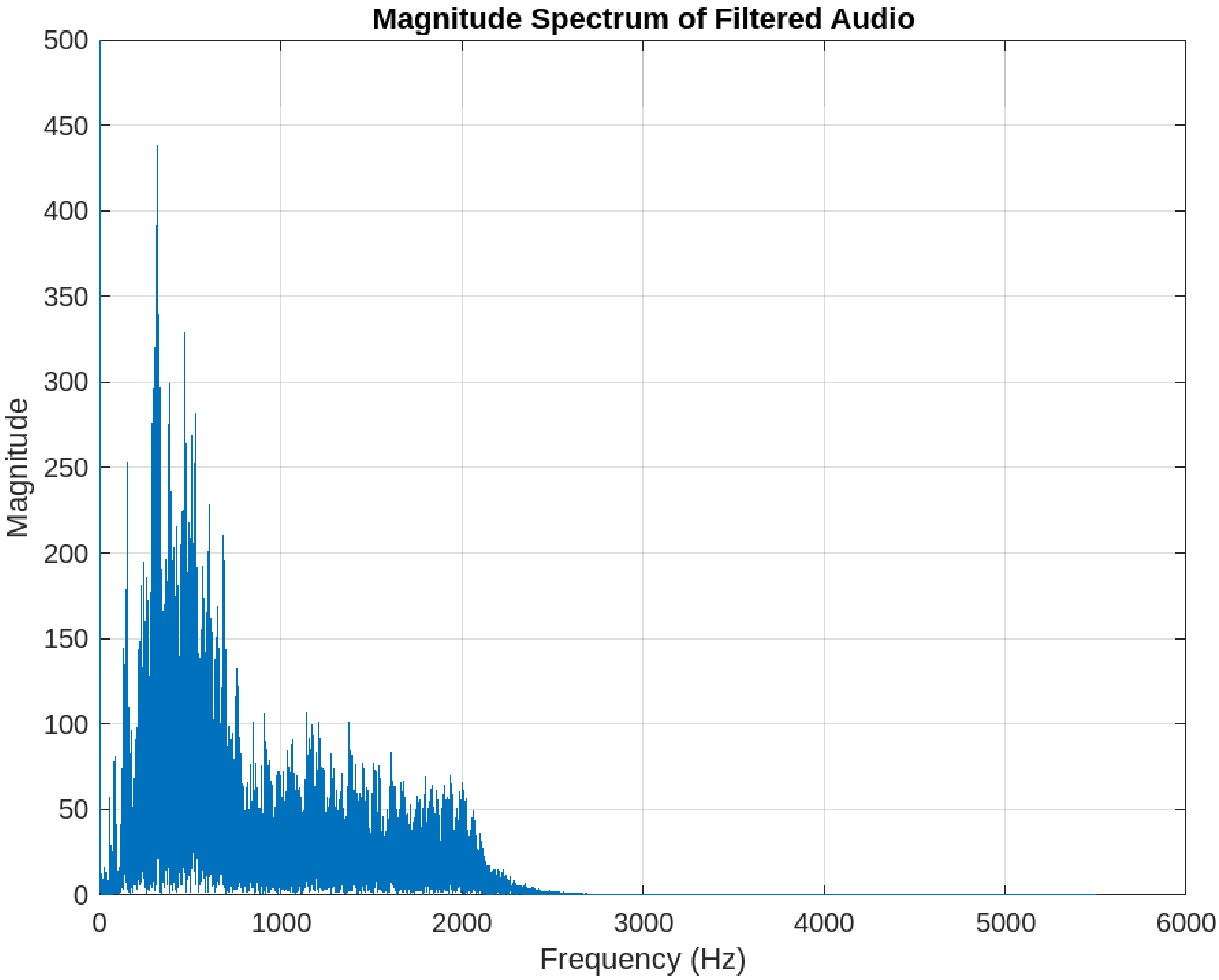

Comparison: Original vs Filtered Spectrum

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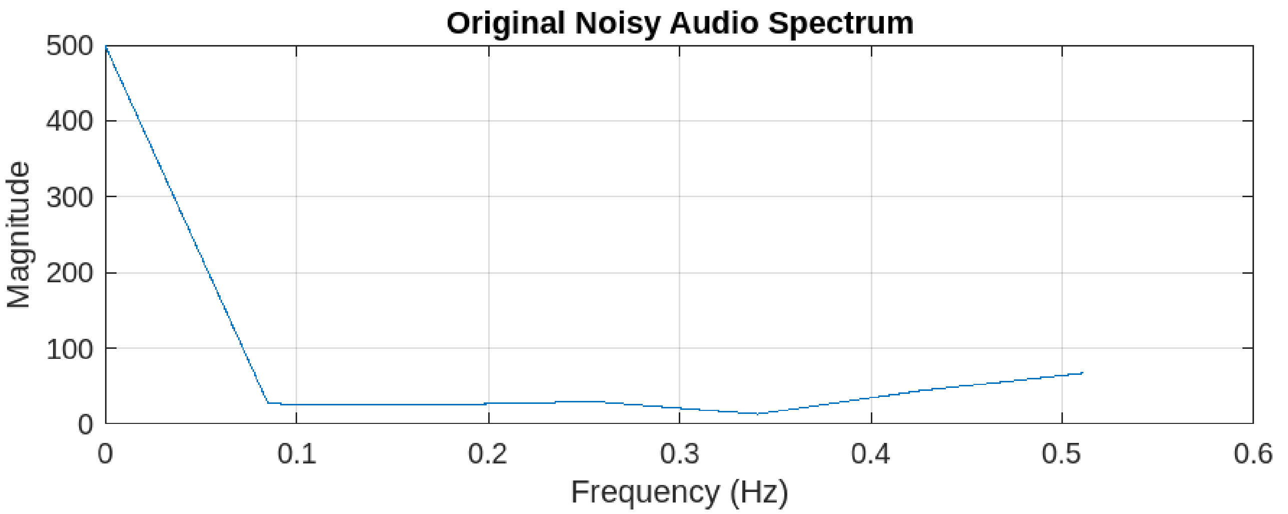

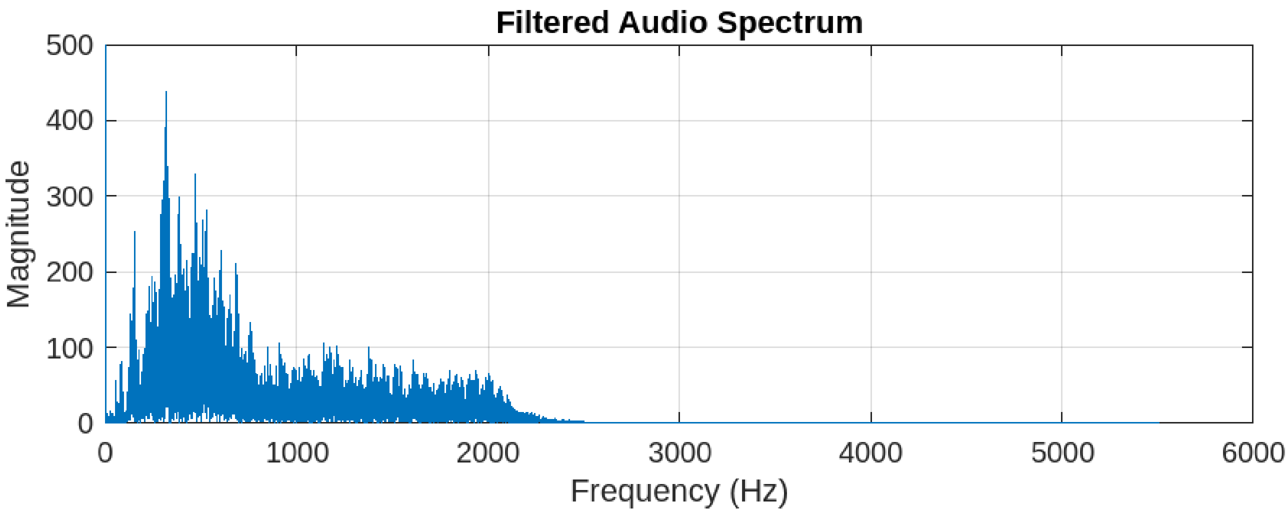

4. Conclusion

The project successfully demonstrated the use of Butterworth lowpass filtering to clean noisy audio. MATLAB's capabilities allowed efficient design, application, and evaluation of the filter.

5. Appendix: MATLAB Code

```
y, Fs = audioread('noisyaudio-1.wav');

N = length(y);

Y = abs(fft(y));

f = (0:N-1)*(Fs/N);
```

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```
wp = 2000; ws = 2500; Ap = 1; As = 30;

wp_norm = wp / (Fs/2);

ws_norm = ws / (Fs/2);

[N, Wn] = buttord(wp_norm, ws_norm, Ap, As);

[b, a] = butter(N, Wn);


y_filtered = filter(b, a, y);

audiowrite('filteredaudio.wav', y_filtered, Fs);

N_filtered = length(y_filtered);

Y_filtered = abs(fft(y_filtered));

f_filtered = (0:N_filtered-1)*(Fs/N_filtered);
```

6. Bonus: Manual Difference Equation Implementation

To verify the Butterworth filter manually, a custom difference equation was implemented using the b and a coefficients. The recursive equation was solved with the initial condition $y[n] = 0$ for $n < 0$. The magnitude spectrum of the manually filtered audio matched the spectrum obtained using MATLAB's filter() function, confirming the correctness of the manual implementation.

Magnitude Spectrum of Manually Filtered Audio

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