Department of Electrical & Electronic Engineering

Independent University, Bangladesh

EEE 321L Section 2

Open Ended Lab Report

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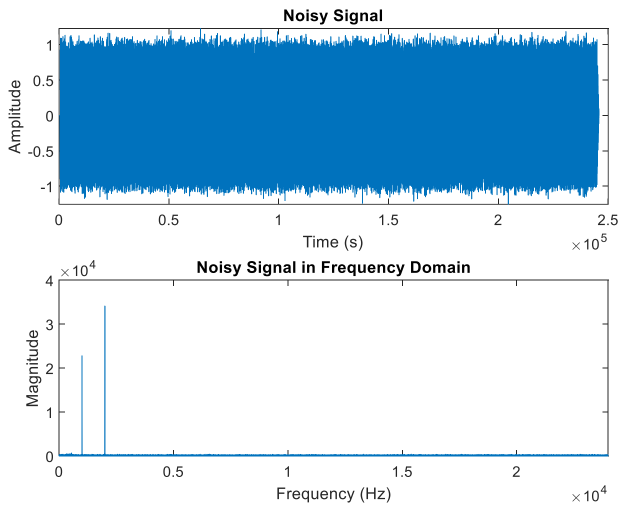
Introduction:

In the realm of audio signal processing, the mitigation of noise from corrupted audio signals is a critical endeavor aimed at enhancing the quality and intelligibility of audio content. Various techniques and methodologies have been developed to address this challenge, with digital filtering standing out as a prominent solution. This report delves into the application of Infinite Impulse Response (IIR) Butterworth bandstop filters and Finite Impulse Response (FIR) bandpass filters as a combined approach to denoise corrupted audio signals.

Objective:

To filter out noise from a corrupted audio file using the MATLAB software and investigate the performance of the filter.

The "corrupted.m4a" file was imported onto MATLAB and analyzed in both the time and spectral domain.



Code:

```
%Hemal Sharma
%ID: 2221855

% Import audio file
[y, Fs] = audioread("corrupted.m4a");     % 'y' contains the audio samples, 'Fs' is the sample rate

% Plot the waveform
subplot(2, 1, 1);
plot(y);
xlabel('Time (s)');
ylabel('Amplitude');
title('Noisy Signal');

% Compute and plot the frequency spectrum
N = length(y);                      % Length of the signal
Y = fft(y);                         % Compute the FFT
f = (0:N-1)*(Fs/N);                 % Frequency vector
subplot(2, 1, 2);
plot(f, abs(Y));
xlabel('Frequency (Hz)');
ylabel('Magnitude');
title('Noisy Signal in Frequency Domain');
xlim([0 Fs/2]);                     % Display only positive frequencies
```

<u>**Methodology:**</u>

Removing the high-intensity pitch:

There are two sharp monotone peaks at 1000 Hz and 2000 Hz and a white broadband noise that overlaps the information signal (the speaker's audio). To remove any sharp peaks, band-stop filters are recommended. Two filters in series, i.e., one filter after the other are implemented in cascaded manner. IIR (Infinite Impulse Response) filters were used as they closely resemble an ideal response. Compared to its FIR (Finite Impulse Response) filter counterpart, the IIR filter approximates a response closest to the ideal with a small transition BW (bandwidth), for only a small order of the filter. Increasing the order of the filter adds more complexities to the overall network and increases the number of delay elements – generally, we want to reduce our filter order to create an efficient design.

Design specifications/ features:

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• Maximum filter order, $N \ll (1+8+5+5 = 19) \therefore N = 16$

• $As\ dB \gg (2+2+2+1+8+5+5 = 25\ dB)$

• $\Delta\omega \ll (2+2+2+8 = 14) \therefore \Delta\omega = 8$

The design process:

Following design specifications, a 16-th Order IIR Butterworth Band Stop filter with center frequencies of 1kHz and 2 kHz has been designed with the aid of the filterDesigner app found in MATLAB.

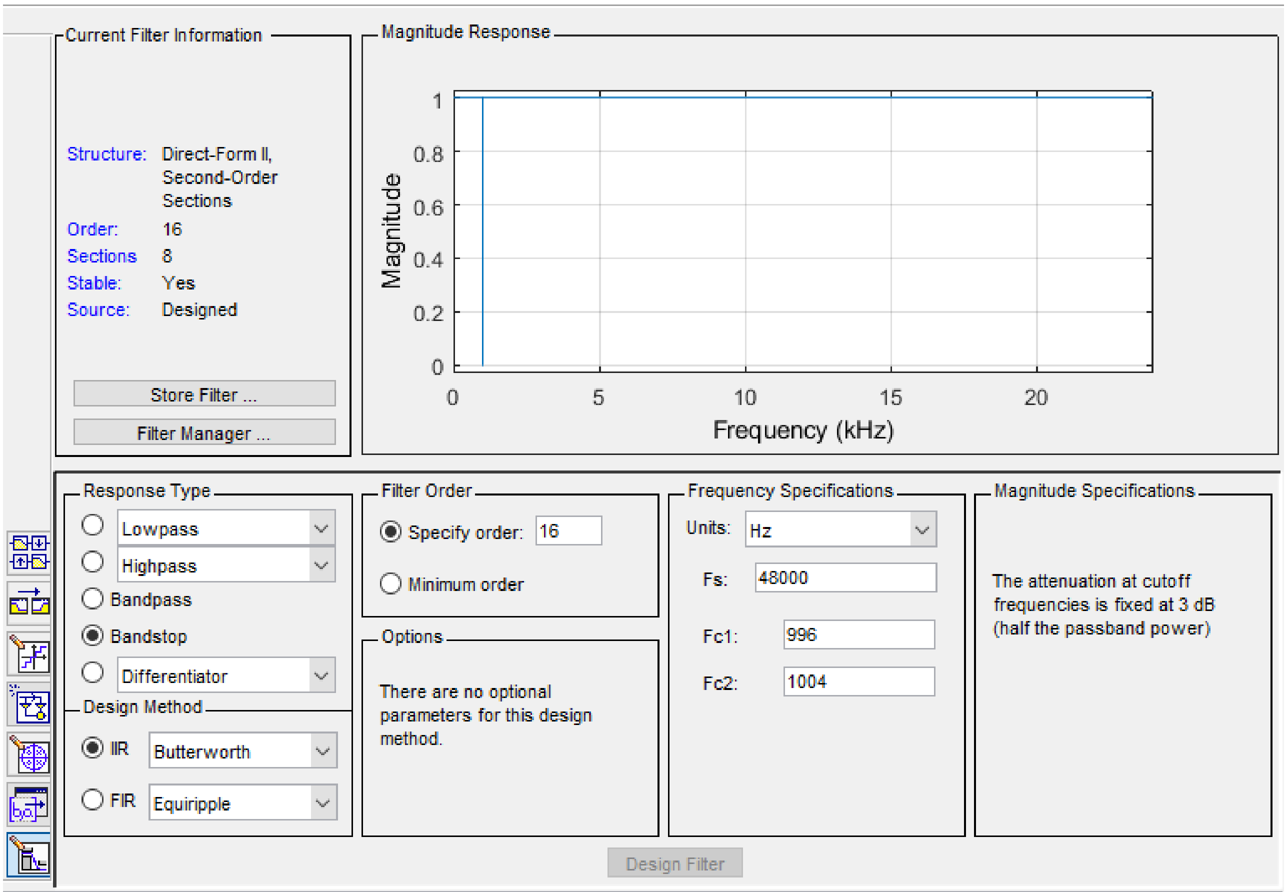

Figure 1 IIR Bandstop filter to suppress 1kHz Peak

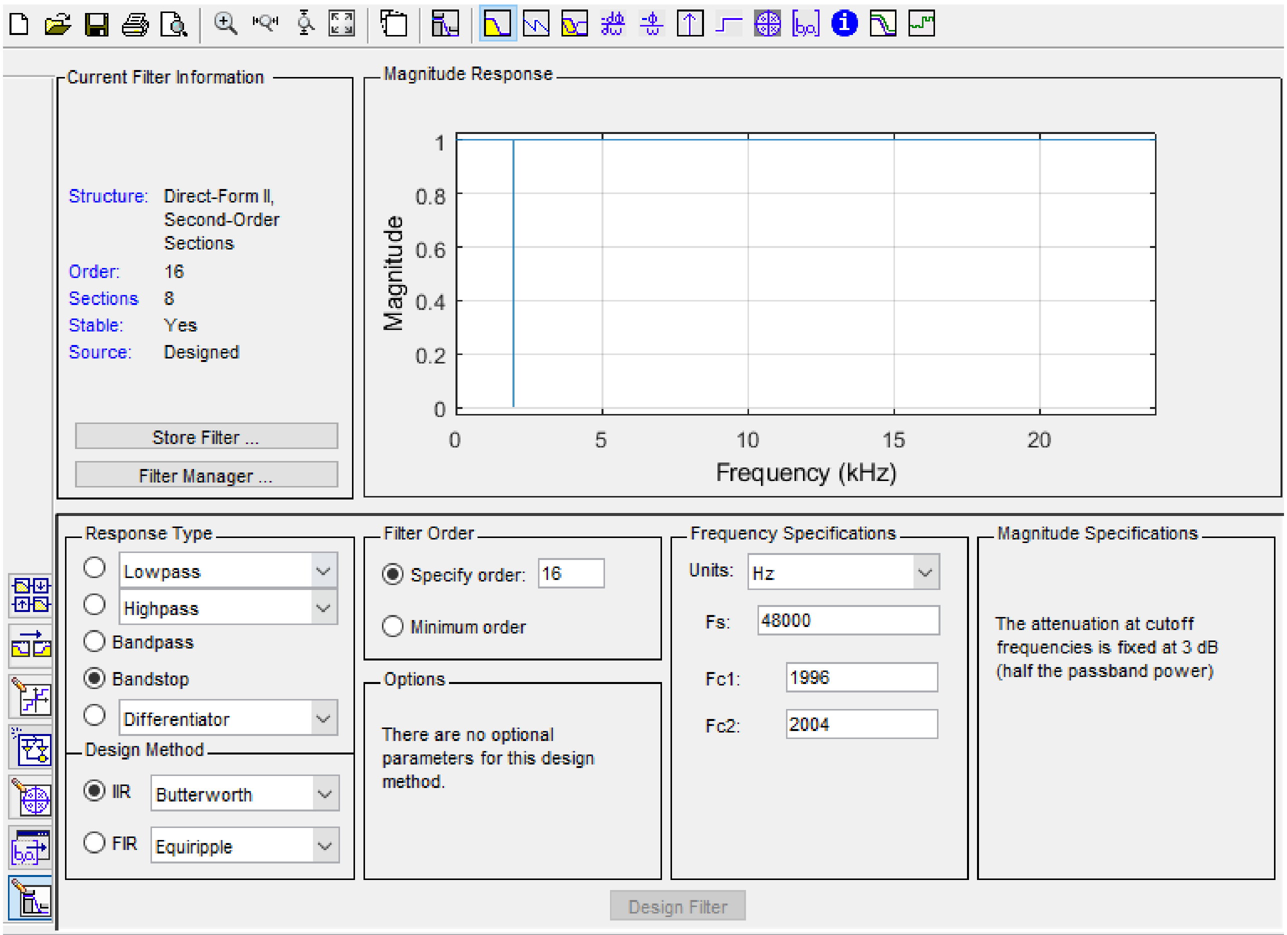

Figure 2 IIR Bandstop Fitler to suppress 2kHz peak

Once the bandstop filters are applied, the high-frequency noise will indeed be effectively suppressed, while the white noise component will persist. To address this residual white noise, the built-in filter() function offers a viable solution. However, it's important to acknowledge that no filter can completely eradicate 100% of the noise from a system; it can only attenuate it to a certain degree. Furthermore, the challenge lies in the fact that the spectrum of white noise often overlaps with the bandwidth of the audio or information signal. Hence, for post-processing, a FIR Bandpass filter, which offers additional refinement capabilities has been chosen.

FIR Bandpass Filter

The FIR (Finite Impulse Response) bandpass filter is a vital tool in signal processing used to isolate a specific range of frequencies from a signal while attenuating frequencies outside that range. Unlike IIR filters, FIR filters are characterized by a non-recursive structure, meaning that output samples are solely based on current and past input samples without feedback loops. This makes FIR filters inherently stable and predictable, with precise control over the filter's frequency response. FIR bandpass filters are widely employed in various applications, including audio processing, biomedical signal analysis, communications systems, radar, and image processing. They offer advantages such as linear phase response, which preserves the timing relationships within the signal, making them suitable for applications where phase distortion needs to be minimized. Additionally, FIR filters are more easily designed to achieve specific performance criteria and offer flexibility in filter design compared to IIR filters.

Using the filterDesigner we can once more design another filter that produces a result that is closely like the expected audio if this filter is excited with the pitch-filtered signal as the input. This route is not the best method for minimizing the noise. Since both the noise and the information are correlated, a reduction of the magnitude of the noise also means the magnitude of the information will also be decreased. The filterDesigner GUI screenshot is provided below for a reference of the filter parameters. Here we used a16th-order digital Bartlett-Window bandpass FIR filter. Through experimentation, cutoff frequencies of 300Hz and 500Hz were determined to provide satisfactory audio results.

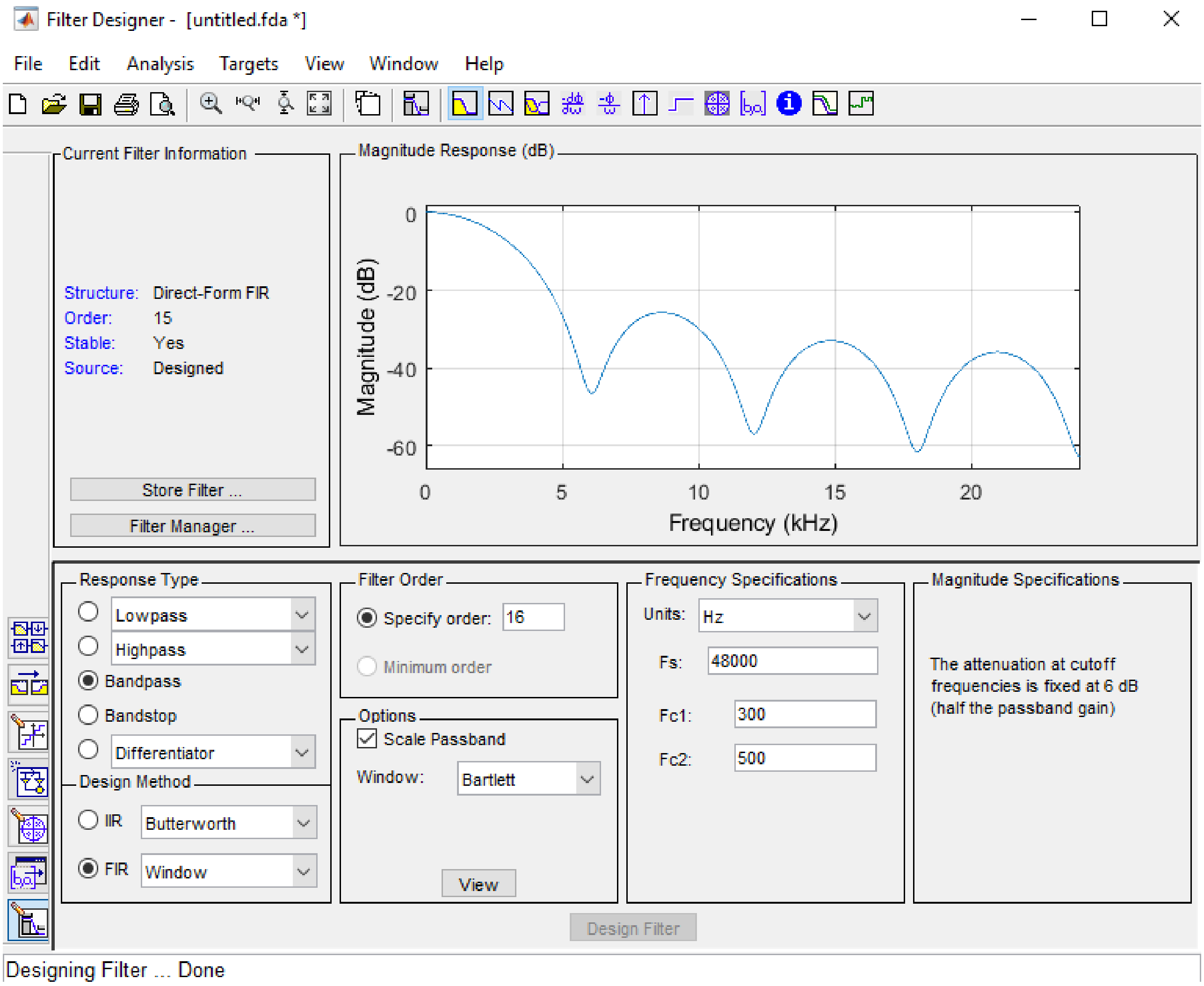

Figure 3 Post Processing using FIR Bandpass Filter

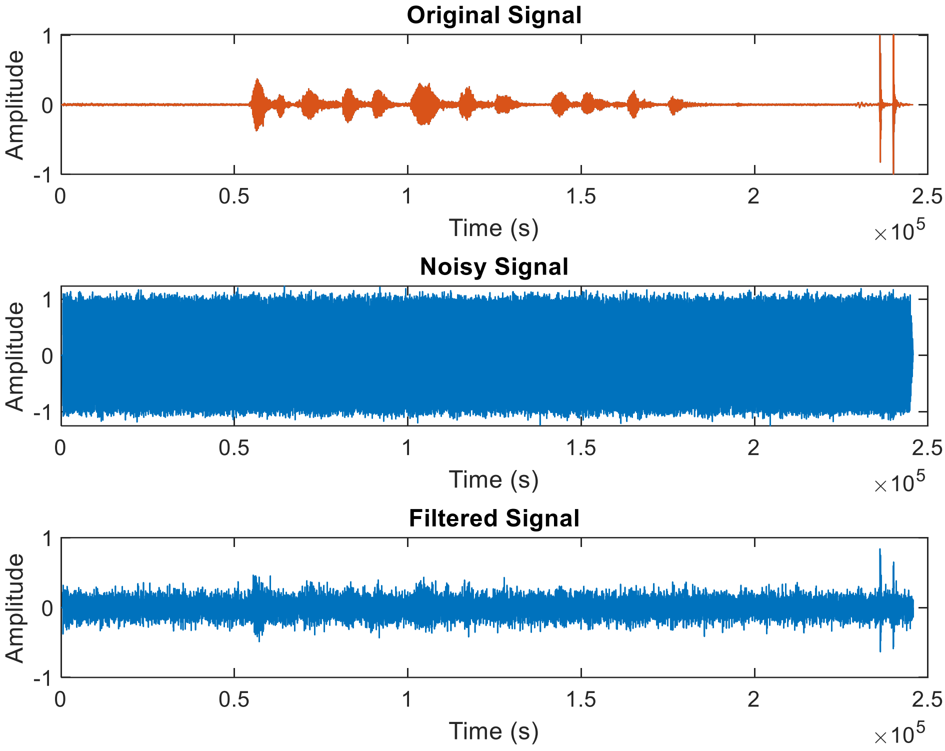

Figure 4 Original, Noisy and Filtered signal in time spectral domain.

Code:

```
%% Reset
clc;
close all;
clear;
%Hemal Sharma
%ID: 2221855
%% Read and play noiseless audio
[noiseless, sample_rate] = audioread('expected.m4a');
sound(noiseless, sample_rate); %plays sound

%% Read and play noisy audio
[noisy_data, sample_rate] = audioread('corrupted.m4a');
sound(noisy_data, sample_rate); %plays sound

%% Compare noisy and noiseless data
subplot(3,1,1);
plot(noiseless);            % Original Signal
title('Original Signal');
xlabel('Time (s)'); ylabel ('Amplitude');

subplot(3,1,2);
plot(noisy_data);           % Noisy output
title('Noisy Signal');
xlabel('Time (s)'); ylabel ('Amplitude');

%% Apply filter 1
my_filter1 = filter1; % filter1 is a bandstop filter to remove the 1000Hz noise
filtered_data1 = filter(my_filter1, noisy_data);

%% Play filtered audio 1
sound(filtered_data1, sample_rate); %plays sound

%% Apply filter 2
my_filter2 = filter2; % filter2 is the bandstop filter to remove the 2000Hz noise
filtered_data2 = filter(my_filter2, filtered_data1); % Apply filter 2 to the output of
filter 1

%% Play filtered audio 2
sound(filtered_data2, sample_rate); %plays sound

%% Apply filter 3
my_filter3 = filter3; % filter3 is a FIR bandpass filter
filtered_data3 = filter(my_filter3, filtered_data2); % Apply filter 3 to the output of
filter 2

%% Plot filtered signal 3
subplot(3,1,3);
plot(filtered_data3);           % Filtered output 3
title('Filtered Signal');
xlabel('Time (s)'); ylabel ('Amplitude');

%% Play filtered audio 3
sound(filtered_data3, sample_rate); %plays sound
```

Conclusion:

In conclusion, the methodology presented in this report showcases an effective approach to denoising corrupted audio signals through a combination of IIR Butterworth bandstop filters and a FIR bandpass filter. Initially, efforts were directed towards suppressing the high-intensity peaks using bandstop filters based on specific filter specifications. However, the filtered audio still exhibited residual noise. Subsequently, the application of a FIR bandpass filter allowed for further refinement of the audio. This approach successfully isolated desired frequency components while attenuating residual noise, ultimately preserving the integrity of the audio signal and resulting in a cleaner and more intelligible output.

Discussion:

Method Choice:

The decision to use IIR Butterworth bandstop filters and a FIR bandpass filter in conjunction for denoising audio signals stemmed from their complementary functionalities and distinct filtering characteristics. Initially, the Butterworth bandstop filters were chosen for their ability to attenuate specific frequency bands, aiming to address the high-intensity peaks at 1kHz and 2kHz frequencies, identified as prominent noise artifacts. However, the initial attempts at noise suppression fell short of desired results, revealing residual noise in the filtered audio. Consequently, an adaptive approach was taken by applying a FIR bandpass filter. This decision was made recognizing the FIR filter's potential to further refine the audio by isolating desired frequency components while continuing to attenuate residual noise.

Improvements:

Experimenting with various filter types and parameters is essential to optimizing noise removal while minimizing distortion in audio signals. Despite the application of IIR Butterworth bandstop and FIR bandpass filters, residual white noise persists, necessitating further refinement. Exploring alternative filter types such as Chebyshev or Elliptic filters may offer sharper roll-off characteristics for improved noise suppression in specific frequency bands, while optimizing parameters like filter order, cutoff frequencies, and transition bandwidths is crucial for balancing noise reduction and signal fidelity. Integration of advanced techniques like adaptive filtering or spectral subtraction could also enhance denoising performance by dynamically adjusting filter parameters or estimating noise spectra from the corrupted signal. Through systematic experimentation and evaluation, the denoising approach can be iteratively refined to achieve optimal noise reduction with minimal distortion.

In addition to exploring different filter types and parameters, increasing the filter order presents another avenue for enhancing denoising performance. By elevating the order of the bandpass filter to 160, satisfactory results were achieved. This adjustment can potentially lead to sharper frequency selectivity and more effective suppression of unwanted noise components, thereby improving the overall quality of the denoised audio signal. Further experimentation with increased filter orders and comprehensive analysis of their impact on noise reduction and signal fidelity can provide valuable insights for optimizing denoising techniques in audio signal processing applications.

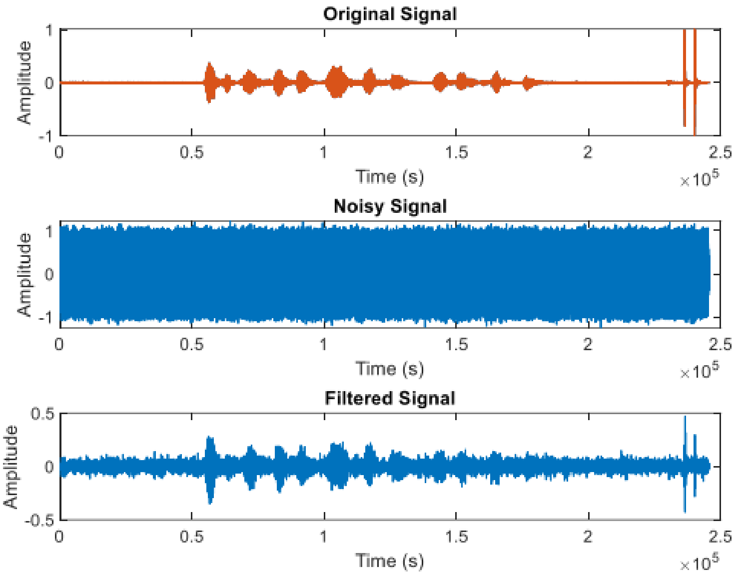

Figure 5 Using bandpass filter of order 160

Further Analysis

Future analysis could expand upon this work by examining the effects of different filter orders and stopband attenuations on the filtered signal's quality. Additionally, exploring the impact of environmental factors on the noise profile and investigating the potential for machine learning algorithms to dynamically adjust filter parameters could offer valuable insights for optimizing denoising techniques in audio signal processing.