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Comparison of FIR and IIR Filters for Audio Signal Noise Reduction

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***Abstract*— An audio signal is often used as a medium for spreading information. However, this signal is easily disrupted by noise from the environment around which the signal is taken, so a filter is needed to eliminate it. In this article, sound signals will be filtered using two types of filters, namely FIR with Hamming window method and IIR with the Butterworth method. Both filters are applied as low pass types with a cut-off frequency of 4000 Hz and an order of 100. Both filters will be employed to filter a 5000 Hz generated audio noise. The SNR and execution time of the filtered signal will then be compared to determine which filter is more effective. The result shows that FIR filters perform better than IIR filters for filtering audio noise.**

***Index Terms*— Digital Filter; Finite Impulse Response- Hamming Window Method; Infinite Impulse Response- Butterworth Method.**

I. Introduction

In this era, information can be quickly disseminated, and everyone can easily access information through radio, television news broadcasts, and the Internet. Most of this information consists of a one-dimensional audio signal. Therefore, the clarity of the audio signal plays an essential role in transmitting information. Sound signals can experience disturbances in the form of noise. This noise originates from the environment where the sound signal is captured using a microphone. Noise disturbance from the environment is sometimes difficult to predict because there are various types of noise, such as motor vehicles with large cylinder volumes passing by, which cause low-frequency noise; the sound of aircraft jet engines which cause high-frequency noise; or even thunder. Not only found in audio signals, noise can be found in ECG (Electrocardiogram) signals from heart's electrical activity [1], EMG (Electromyography) from muscle's electrical activity [2], and EEG (Electroencephalography) from brain wave [3] [4]. Although information can sometimes be received, it will disrupt the receiver. That is why a filter is needed to reduce the existing noise.

A filter is a device or process that can eliminate unwanted parts of a signal from the input signal and enhance the signals [2]. A filter will only pass through or retain the desired signals [5][6]. There are various filters, such as low pass, high pass, band pass, and others. Previous comparative studies used FIR and IIR filters on ECG, EEG, EMG, and sinusoidal signals. Based on these studies, the FIR filter shows good performance when the filters are implemented to filter noise in ECG, EEG, and EMG signals due to their flexibility, faster execution time, more stable response, and lower phase distortion [1][2][3][4]. Furthermore, the IIR filter is used on sinusoidal and ECG signals because it can use lower orders for similar results and lower computational power and memory usage [7][8][9]. So in this study, FIR and IIR filters with the exact specifications will be applied to filter the audio signal.

FIR was selected due to its easiness in implementation, while IIR was chosen because it requires less memory and processing time when implemented in software [10][11]. The FIR filters and IIR will be implemented to filter audio signal data which added noise at a frequency of 5000 Hz. The noise frequency was chosen to mimic the frequency of electrical interference [12]. This research will compare the effectiveness and performance of both filters based on Signal-to-Noise Ratio (SNR) and the execution time.

II. Digital Filter

A. Audio Signals

Audio is a wave that is formed when an object vibrates. The contents of this wave consist of a combination of various frequencies, amplitude, and phase. For example, sound can be produced when a drum is struck, causing it to vibrate, and the vibrations propagate through a medium such as air. A signal is a function of one or more variables which can also be defined as changes that can be observed in an entity that can be measured [13]. Therefore, the audio signal is an electronic representation of sound waves that can be directly observed and processed. Humans can hear a sound with frequencies ranging from 20 Hz to 20,000 Hz [14].

B. Digital Filters

A digital filter is a mathematical algorithm that operates on discrete-time signals and samples, allowing it to enhance or reduce aspects of specific signals. Digital filters are more stable and relatively straightforward yet compact than analog filters [6]. These filters are widely used in signal processing and differ from analog filters, which are electronic circuits consisting of resistors, inductors, capacitors, and other components that work with continuous signals [15]. Digital filters contain analog-to-digital converters (ADC) and digital-to-analog converters (DAC) so that digital filters can have input and output as analog signals. Moreover, digital filters are commonly used in digital signal processing, control systems, and communications [16][17][18]. Digital filters can be implemented as Finite Impulse Response (FIR) filters or Infinite Impulse Response (IIR) filters [3]. The difference between FIR and IIR is presented in Table 1.

TABLE I. THE DIFFERENCE BETWEEN FIR AND IIR [10]

FIR Filters	IIR Filters
No phase distortion is introduced into the signal by the filter	Has a non-linear phase response
When realized non-recursively, filter results are always stable	Filter results are not guaranteed to be stable
The effect of using a certain number of bits on applying a filter, such as roundoff noise and coefficient quantization errors, are much more severe	The effect of using a certain number of bits on applying a filter, such as a roundoff noise and coefficient quantization errors, are negligible
Requires more coefficients for sharp cutoff filters	Requires less coefficient for sharp cutoff filter
Easier to synthesize filters with arbitrary frequency responses	Easier to convert analog filters to digital IIR filters

FIR is a filter in which impulse response has a finite duration, meaning it becomes zero within a finite time. This filter several Window methods, such with its characteristics, namely Rectangular, Hanning, Hamming, Blackman, and Kaiser. The Hamming method is the most used Window method, which provides the best results compared to other FIR types for the exact specifications [19]. The coefficient calculation for designing the FIR filter is given in Eqs. (1) [5].

$$h_D(n) = \frac{1}{2\pi} \backslash \int_{-\pi}^{\pi} H_D(\omega) e^{j\omega n} d\omega \tag{1}$$

where $h_D(n)$ represents the ideal impulse response with the equation is presented in Eqs (2).

$$2fc = \frac{\sin(n\omega_c)}{n\omega_c} \tag{2}$$

IIR filters are unique because they use a feedback mechanism, which requires current and previous output data. Although more challenging to design, the IIR filter is more efficient and cheaper. Furthermore, the IIR filter has feedback, so it has an infinite time limit on its impulse response [20]. IIR filters have several methods with distinct characteristics, including Butterworth, Chebyshev, and Elliptic. Based on previous research, the Butterworth method is relatively better than the Chebyshev method if a flat band response is required [21].

The coefficients for designing the IIR low pass filter use the transfer function equation given in Eqs. (3) – (5) [5].

$$H_{LP}(z) = \frac{b_0(z+1)}{z-a} \tag{3}$$

With:

$$a \approx \begin{cases} 1 - 2\pi\left(\frac{F_c}{F_s}\right) & F_c < \left(\frac{F_s}{4}\right) \\ \pi - 1 - 2\pi\left(\frac{F_c}{F_s}\right) & F_c > \left(\frac{F_s}{4}\right) \end{cases} \tag{4}$$

$$b_0 = \frac{1-a}{2} \tag{5}$$

C. Signal-to-Noise Ratio

Noise can be defined as an unwanted signal that interferes with the communication or measurement of other signals [22]. The effect of this noise on the audio signal is that it can change the information carried by the audio signal to the receiver so that the received information differs from what was previously desired [23]. This noise can come from the environment where the sound signal is picked up and cannot be predicted.

Signal-to-Noise ratio is a parameter that compares the desired signal level to the background noise level [14]. SNR is the signal strength ratio to noise strength measured in decibels (dB). A ratio greater than 0 dB or higher than 1:1 indicates there is more signal than noise. Therefore, a higher SNR value will give a better-quality signal. SNR can be calculated by using Eqs. (6).

$$SNR = 10 \log \frac{P_{signal}}{P_{noise}} \tag{6}$$

III. METHODS

The proposed method is presented in Fig. 1. A 5000 Hz noise was added to the audio signal and plotted in the time and frequency domain to see the characteristics of the audio signal. The filtering step will be performed on those signals to remove the noise. A low-pass filter is implemented to pass signals at frequencies below the cut-off value. Because the noise is at a frequency of 5000 Hz, a 4000 Hz cut-off frequency is used. Two types of digital filters are used for comparison: the Finite Impulse Response (FIR) filter with the Hamming Window method and the Infinite Impulse Response (IIR) filter with the Butterworth method. Both types of filters will use the same sampling frequency, filter order, and cut-off frequency to find out which type of filter performs better. The comparison uses the signal-to-noise ratio (SNR) and the time needed to process the signal.

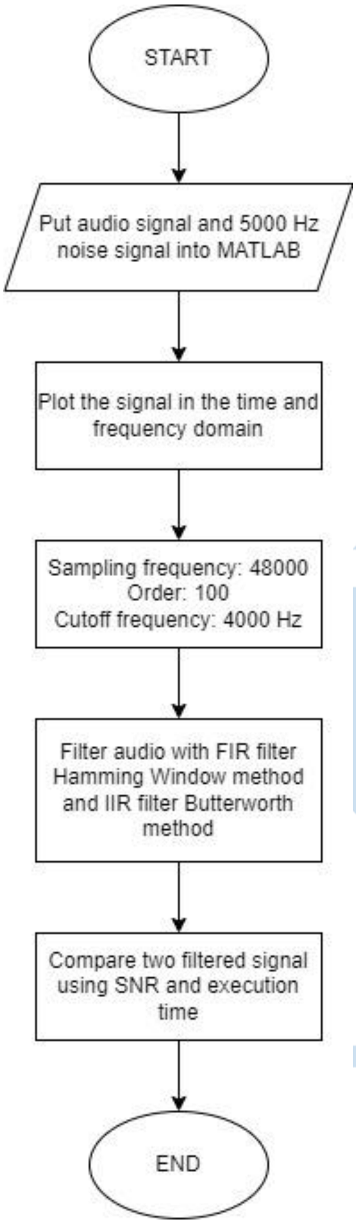

Figure 1. Research method flowchart

IV. RESULTS AND ANALYSIS

The audio signal used in this study is obtained from Red Robbo's Workshop [24]. The audio signal was then given a noise at a frequency of 5000 Hz obtained from Sonic Electronix [25]. The sound signal contaminated with the noise is plotted and find the peak of the noise signal.

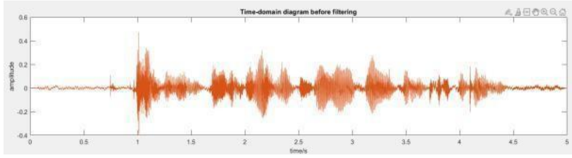

Figure 2. Time Domain Graph of Sound Signal With Noise Before Filtering

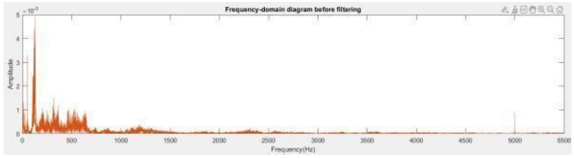

Figure 3. Frequency Domain Graph of Sound Signal With Noise Before Filtering

The time and frequency domain of the unfiltered signal is presented in Figure 2 and Figure 3, respectively. The noise in the signal is located at a frequency of 5000 Hz. For this reason, a cut-off of 4000 Hz is employed with a low pass filter so that frequencies lower than the cut-off will be passed on. The specification of the implemented FIR and IIR filter is shown in Table 2. Two types of filters are used in the form of FIR Hamming Window and IIR Butterworth filters, each with the exact specifications as in Table 2. The filter order is selected based on the trials carried out in the pre-study to get the optimal precise audio signal results on both filters.

TABLE II. THE SPECIFICATION FIR HAMMING AND IIR BUTTERWORTH

Specs	FIR Hamming	IIR Butterworth
Sampling Frequency (Hz)	48.000	48.000
Order	100	100
Cut-off Frequency (Hz)	4000	4000

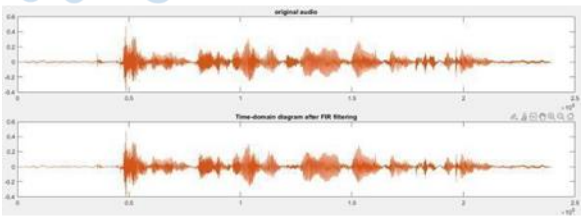

Figure 4. Comparison of time domain noisy audio signals before (top) and after (bottom) filtering using FIR filters

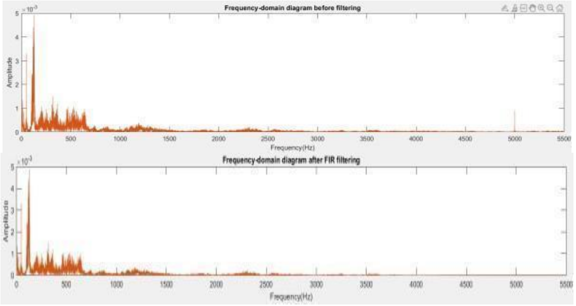

Figure 5. Comparison of frequency domain noisy audio signals before (top) and after (bottom) filtering using FIR filters

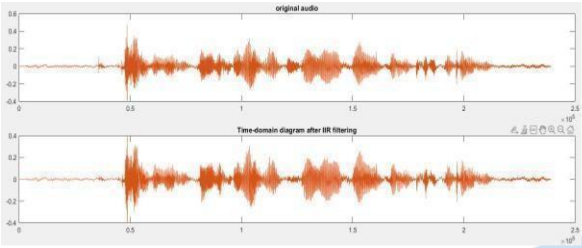

Figure 6. Comparison of time domain audio noisy signals before (top) and after (bottom) filtering using the IIR filter

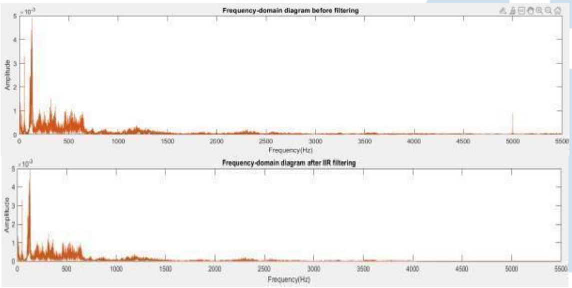

Figure 7. Comparison of frequency domain audio noisy signals before (top) and after (bottom) filtering using the IIR filter

The time domain signal presented in Figure 4 and Figure 6 shows that noise in the signal filtered using both FIR and IIR has decreased. It can be seen that the thickness of the graph generated in the time domain before being filtered is thicker than that of the time domain that has been filtered. This difference in the thickness of the graphs proves the loss of 5000 Hz noise from the signal.

The result in the frequency domain, as seen in Figures 5 and 7, shows a missing signal around the frequency of 5000 Hz. This result shows that both Hamming Window and IIR Butterworth FIR filters can remove noise at a cut-off frequency of 4000 Hz. This result proves that FIR and IIR can eliminate noise at the desired frequency, but the graphs produced by the two filters do not show significant differences. The difference between these filters can be seen in the SNR and execution time. The compassion of the SNR and the execution time of both filters is given in Table 3.

TABLE III. COMPARISON OF EXECUTION TIME AND SNR OF IIR FILTER WITH FIR FILTER

	Without Filter	IIR	FIR
Elapsed Time (s)	–	0,817776	0,213473
SNR (dB)	28.4766	31,4774	31,4782

TABLE IV. EXECUTION TIME FORM MULTIPLE EXECUTIONS

Execution	IIR	FIR
1	0.784473	0.16293
2	0.79077	0.168706
3	0.78685	0.165422
4	0.804127	0.161626
5	0.789127	0.152496
6	0.854054	0.160514
7	0.857525	0.161091
8	0.845725	0.169508
9	0.884793	0.169061
10	0.778916	0.156219
11	0.846445	0.168094
12	0.843364	0.171907
13	0.845329	0.161352
14	0.848843	0.183132
15	0.835147	0.174549
Average	0.8263658667	0.1657738

As seen in Table 3, the time required by the FIR filter is less than that required by the IIR filter, with the value of 0,165773 seconds and 0,826365 seconds, respectively. The execution time in Table 3 is the average result of 15 execution trials, with each execution time result given in Table 4. The time difference is insignificant because both filters require a short processing time. The SNR of the signal filtered using the FIR filter is 31.4782 dB, more significant than the IIR filter, which has an SNR of 31.4774 dB. The results show that the FIR filter with the Hamming Window method is more effective than the IIR filter with the Butterworth method in filtering the noise in the audio signal. Although the difference in execution time and SNR is minimal, implementing both filters in a more extended audio signal will give a more diverse result.

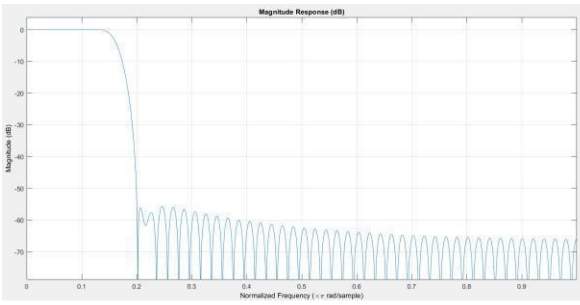

Figure 8. Magnitude response of FIR Hamming Window filter

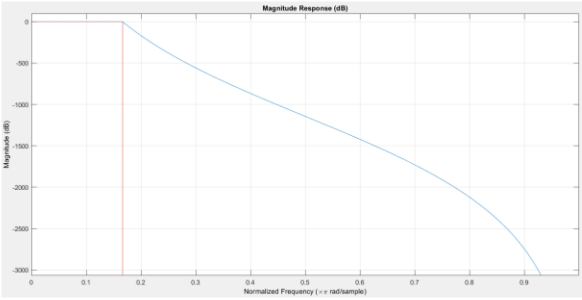

Figure 9. Magnitude response of IIR Butterworth filter

The magnitude response of both the FIR and IIR filter are presented in Figure 8 and Figure 9, respectively. In Figure 8, it can be seen that the cut-off frequency of the audio signal is immediately attenuated. In contrast, the signal is already attenuated in Figure 9 from before the cut-off frequency. In Figure 8, the magnitude response of FIR filter results has more ripples than the results of the IIR filter in Figure 9. However, the FIR filter has a shorter transition band than the IIR filter's transition band. Nevertheless, both filters can filter the noise efficiently.

V. CONCLUSIONS

This study compares the performance of the FIR Hamming method, and IIR Butterworth implemented in a low-pass filter for noise reduction in the audio signal. The result shows that both filters can filter noise effectively and produce clear audio signals. However, the FIR filter with the Hamming Window method is more recommended to be used in filtering 5000 Hz noise in the sound signal because the SNR result by the FIR filter with the Hamming Window is higher than the SNR of IIR filtered signals. In addition, the FIR filter with Hamming Window method has a shorter execution time of 0,165773 seconds than IIR filters with the Butterworth method, which has an execution time of 0,826365 seconds. This time execution difference will be very influential in filtering audio signals that have a long duration.

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