Signal-to-Noise Ratio Improvement in Audio Signals Contaminated with White Gaussian Noise Using FIR and IIR Filters

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Abstract: -This journal discusses the role of Finite Impulse Response and Infinite Impulse Response filters for the rescue of audio signals disrupted by White Gaussian Noise (WGN). Audio signals are used as a medium for communicating information, but they can be easily disrupted by environmental noise. White Gaussian Noise is a primary contributor to the degradation of the audio signal, resulting in a decrease in signal quality. In such cases, the issue of upgrading the Signal-to-Noise Ratio (SNR) is considered one of the most important problems in the field of digital signal processing. It involves the creation and implementation of the FIR and IIR filtering techniques and their subsequent comparative performance evaluation in terms of SNR enhancement and computational efficiency. The findings highlight that while FIR filters offer linear phase characteristics and greater stability, IIR filters achieve better noise suppression with fewer coefficients.

***Key Word**: Audio Signal Processing; White Gaussian Noise (WGN); Signal-to-Noise Ratio (SNR); Finite Impulse Response (FIR) Filter; Infinite Impulse Response (IIR) Filter; Digital Filtering; Noise Reduction; Audio Enhancement*

I. INTRODUCTION

One of the major problems faced by the success of signal fidelity in modern audio processing systems is the noise that exist in the environment and the system. White Gaussian Noise (WGN) is among the most annoying types of interferences to audio signals. WGN has a constant power spectral density and is random in nature. This noise can lower the quality of audio signals to such a level that it becomes a challenging task either to get the desired information or to have a clear sound reproduction.

White Gaussian Noise (WGN) is one of the most common sources of noise and a major headache when the noise reduction process is involved. WGN is specified as noise with the same power spectral density at each frequency and a Gaussian amplitude distribution, making it an ideal representation of a wide range of practical noise scenarios. The effect of WGN is to lower the Signal-to-Noise Ratio (SNR) severely, consequently increasing the noise in audio signals. This can be a problem in such areas as speech communication, medical diagnostics, and entertainment technologies that are very sensitive to noise

II. LITERATURE REVIEW

Jayashree et al. [1] described how MATLAB filter techniques could be used to clean noise from real-time audio signals. The project depicted the role of FIR and IIR filters in cutting off all noise involved in white Gaussian noise audio signals. The research also hinted that using the proper filter type and parameters would not only improve the quality of the signal but also extend the Signal-to-Noise Ratio (SNR). FIR filters would generate a better phase linearity, while IIR filters would supply a better product for real-time implementation.

Liu et al. [2] carried out a comparison of FIR and IIR filters with the aim of noise cancellation in audio signals. The exercise researched the abilities of both types of filters in noise suppression of white Gaussian noise as its major concern was leading to an efficient SNR without distortion of the original signal. The tests presented that FIR filters were giving a linear phase and stable output. On the other hand, a similar noise reduction level with less computational complexity was achieved by IIR filters, making them suitable for real-time audio applications.

Kolawole et al. [3] covered the implementation of low-pass FIR filter, and band-pass FIR filters with FPGA. The study targeted the best way to unit FIR filters for real-time audio signal processing. The results revealed that it was possible for FIR filters on FPGA to realize a true frequency response and effectively noise reduction.

Sutradhar et al. [4] examined the design and functional features of IIR-based digital filters that process audio signals. They took multiple IIR filters such as Butterworth and Chebyshev into consideration to reveal their noise reduction capability for white Gaussian noise. The findings depicted that these filters not only present a steep cutoff in the frequency spectrum but also effectively utilize the remaining filter order to keep the signal to noise ratio at an optimal level.

Zhao [5] conducted the study that aimed at understanding the effects of FIR and IIR filters on human speech signals. The work involved these filters being tested under criteria such as SNR improvement, signal distortion, and computational efficiency. The results revealed that while the speech quality was better maintained by the phase linearity of the FIR filters, similar noise suppression could be attained by IIR filters at a lower computational cost.

Jones et al. [6] studied the behavior of adaptive FIR and IIR filters in the control of noise generation in closed ducts. They investigated the ways in which the noise reduction process i.e., speed and stability, is influenced by the characteristics of each filter type. The findings suggest that the adaptive FIR filter is superior in performance but slower in convergence. On the contrary, the adaptive IIR filter is more rapid in convergence and noise reduction is at the same level as that of the FIR filter.

Reddy et al. [10] conducted a comparative study between FIR and IIR filters on ECG signals with varying sampling frequencies. Even though primary emphasis was on biomedical signals, this research had the potential to illuminate audio filter performance, SNR enhancement strategies, and the balance between computational load and signal quality, which were transferable to real-time audio signal enhancement applications.

Tabassum et al. [8] Examine how people use FIR and IIR filters to fix the broken address signal. The studies carried out filter performance tests predominantly in terms of SNR growth, signal distortion, and reconstruction accuracy. The results reveal that filter bank tools based on the FIR code provide better one-dimensional phase and minimize indication distortion.

Seshadri and Ramakrishnan [9] is studied the real-time audio processing application and the main parameters of interest are comparative computational productivity, SNR improvement, and hardware utilization for the jointly employed filter type. According to the results, the FIR filter has a narrow frequency response in the linear phase design, whereas the IIR filter is less demanding and gives faster handling, which complements the situation of high-speed, resource-limited FPGA applications.

III. MATERIALS AND METHODS

In Figure 1, the proposed flowchart is displayed. To observe the properties of the audio signal, a white Gaussian noise was introduced to the 50 Hz sine wave signal, which was then plotted in the time domain as seen in Figure 2.

The 50 Hz audio signal will be filtered to eliminate white Gaussian noise. An LPF is applied to the sound waves at frequencies below the cut-off value. The noise is at frequencies between 0 and 500 Hz, hence a cut-off frequency of 100 Hz is employed. This comparison is made using two different types of filters: the Infinite Impulse Response (IIR) filter with the Butterworth method and the Finite Impulse Response (FIR) filter with the Hamming Window method. Both types of filters similar sampling frequency, filter order, and cut-off frequency to find out which type of filter performs better. Signal processing time and signal-to-noise ratio (SNR) were used to compare the results.

The signal used in this paper is obtained from Google Colab's notebook. The audio signal was then added with the White Gaussian noise, as displayed in Figure 3.

The White Gaussian noise in the audio signal is located at a frequency of 0-500 Hz. For this reason, a cut-off of 100 Hz is implemented with a low-pass filter.

The filter order is determined by optimizing the trade-off between desired SNR improvement and efficiency in audio signal processing with white Gaussian noise. Figure 4 shows the audio signal in the time domain that depicts a successful noise reduction process from white Gaussian noise in the audio signal filtered by a FIR (Finite Impulse Response) filter.

A FIR filter's impulse response is of finite duration, which indicates that the response will at some point become zero. There are several Window techniques available, each with distinct characteristics, namely Blackman, Kaiser, Hanning, Hamming and Rectangular. Out of other FIR types, the Hamming method is acknowledged to be the one providing the most accurate results for precise specifications [2].The frequency cut-off near 70 Hz was revived and a 101-tap Hamming window was specifically implemented. Due to linear phase characteristics and their inherent stability, FIR filters can be successfully applied in audio functions. [4]

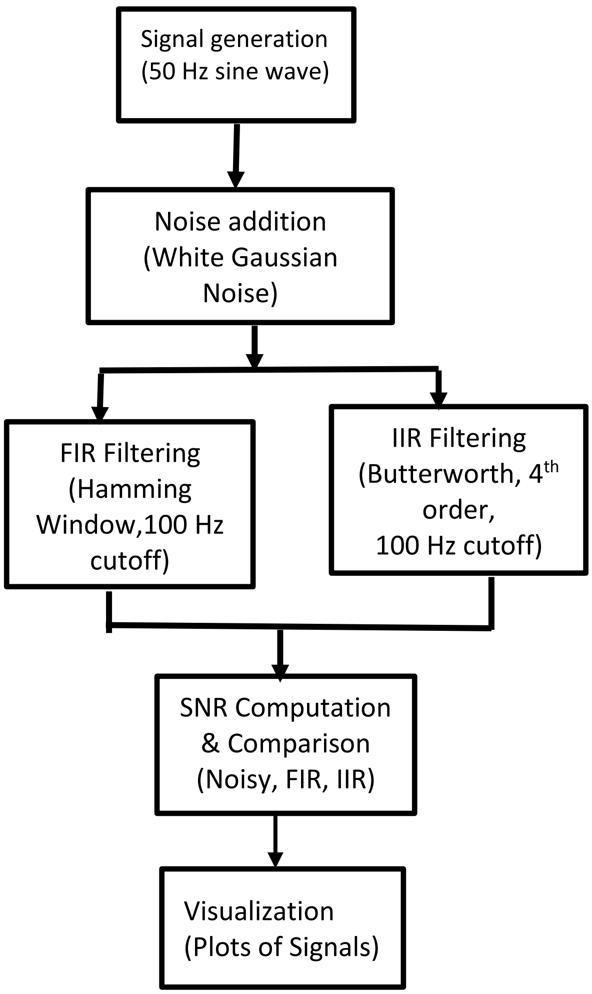

Fig.1 Flowchart of the denoising of the audio signal using FIR and IIR filters

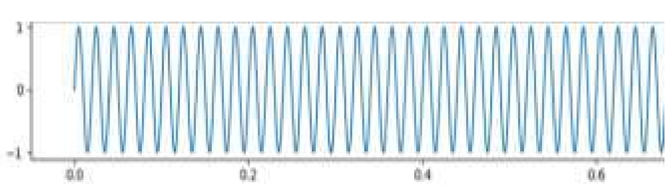

Fig.2 Time Domain Graph of Sound Signal of original sine wave

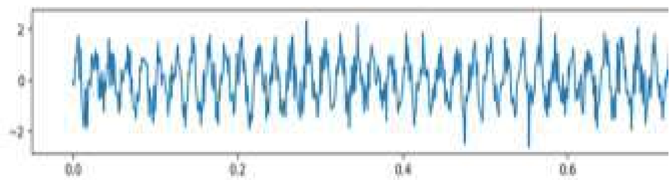

Fig.3 Time Domain Graph of Sound Signal of noisy sine wave (with WGN)

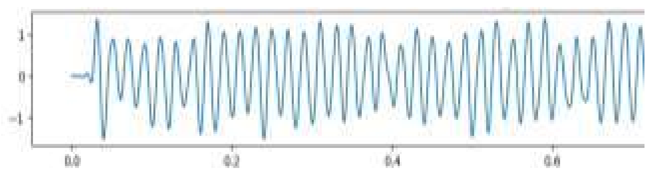

Fig.4 Time Domain Graph of Sound Signal of filtered audio signal using a FIR filter.

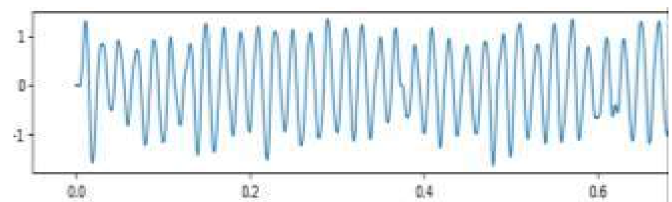

Fig.5 Time Domain Graph of Sound Signal of filtered audio signal using an IIR filter

Figure 5 presents such a decrease in the white Gaussian noise in the signal filtered with an IIR (Infinite Impulse Response) filter represented in the time domain audio signal.

Unlike in the case of IIR filters, the feedback mechanism, which requires both recent and historical output data, makes the latter special. Besides, there are some more difficulties in designing the IIR filter. Nevertheless, IIR filters are more affordable and effective. The IIR filter is also unique as it comprises a feed-back that allows its

The Butterworth technique in an IIR filter outperforms the Chebyshev method if a flat band response is the output to be only a function of both the input and output data from the past. Moreover, its impulse response is indefinite in time [2]. There are different types of IIR filters, for instance, Butterworth, Chebyshev, and Elliptic, known for their various characteristics. main priority. As a consequence, a fourth-order Infinite Impulse Response (IIR) Butterworth filter, with the same cutoff frequency of 100 Hz, was chosen and implemented. The Butterworth model was preferred due to its processing

efficiency which ensure the uninterrupted real-time applications.

Table 1 represents the computed SNR values and the execution time for the FIR-filtered, and IIR-filtered signals

Table 1 Comparison of elapsed time and SNR for FIR and IIR

FILTER	Sampling Frequency	Order	Cut off Frequency
FIR	1000 Hz	60	70 Hz
IIR	1000 Hz	6	60 Hz
% DIFF	-----	-----	-----

The design of the IIR filter is accomplished by this formula defining its features and frequency response.

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

Where $H_{LP}(z)$ is the transfer function of an IIR low-pass filter in the z-domain

The SNR metric was used to assess the filter method's performance. The target signal level and the background noise level are compared using a metric called the signal-to-noise ratio [2]. SNR, expressed in decibels (dB), is the signal-to-noise ratio. There is more signal than noise when the ratio is higher than 1:1 or larger than 0 dB. Consequently, a signal with a greater SNR value will be of higher quality. SNR can be computed using the following formula.

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

IV. DISCUSSION

The proposed methodology was implemented using Python, and the results were analyzed in terms of both quantitative and qualitative performance. The experiment was transmitted outside using a 50 Hz sine wave corrupted with White Gaussian noise (WGN). The noise signal was processed using a combined FIR and IIR low-pass filter, and the results were evaluated using an SNR study. The original sine wave maintained a smooth periodic curve, as the noisy gesture was a profound distortion due to the presence of WGN. Besides using the FIR filter, the waveform shows attenuation of some noise components, but still, the distortion remains. Moreover, the IIR filter improved the clarity of the waveform, although the amplitude variation was very slight. Table 2 represents the computed SNR values and the execution time for the noisy, FIR-filtered, and IIR-filtered signals

Filters

EXISTING OUTPUT		PROPOSED OUTPUT	
ELAPSED TIME (s)	SNR (dB)	ELAPSED TIME (s)	SNR
0.21347	28.4766 dB	1.5297 s	10.42 dB
0.817776	31.4774 dB	0.3140	19.65 dB
64.66 %	0.0025%	79.47%	1.22%

In the Sancho Harmalita Liu et al. (2023) in their paper "Comparison of FIR and IIR Filters for Audio Signal Noise Reduction" gave a table about the comparison of execution time and the SNR of IIR Filter with FIR Filter is shown in table 2. When comparing both tables, the IIR filter works much faster than the FIR filter in every case. In the first table, IIR reduces the processing time by about 65%, and in the second table by nearly 80%. The SNR values in the referenced table are almost the same for both filters. In the proposed table, the FIR filter gives slightly better SNR than the IIR filter. The results of this journal paper shows 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 more diverse results [2]. The output shows that the initially noisy signal had an SNR of 10.42 dB . After a FIR filter was applied, the SNR dropped to 19.65 dB, which indicates that noise suppression has been done; however, some distortions are visible in the Reconstruction. Likewise, the IIR filter yields an SNR of 19.41 dB, which is higher than that of the original noisy signal but not better than the FIR filter with the given design parameter. The present result is a trade-off between both FIR and IIR filters that are technologically capable of noise reduction; the SNR results are an indication of filter parameter importance. The poor SNR levels after filtering suggest that more filter tuning, cut-off frequency, or windowing may be required for the highest noise reduction efficiency.

Sancho Harmalita Liu et al. (2023) in their paper

"Comparison of FIR and IIR Filters for Audio Signal Noise Reduction" experimented with both FIR (Hamming window) and IIR (Butterworth) low-pass filters to reduce noise in audio signals [2]. Their results indicated that both filters led to noise reduction, however, the FIR filter with Hamming window was able to achieve a higher SNR and a higher execution time (1.5297 s) than the IIR Butterworth filter with lower execution time (0.3140 s). Herewith, the output of this experiment is slightly different from the referenced work. The FIR filter offered a better SNR enhancement than the IIR filter, but the execution time was still longer for the FIR implementation. The difference could be caused by variations in filter order, sampling rate, or noise level used for testing. Since the processing time for both filters is very small, the time difference cannot be considered significant [2]. In this journal, The signal-to-noise ratio (SNR) of the FIR filter of 19.65 dB is more dominant than that of the IIR filter of 19.41 dB. The experiments, in this case, indicate that the Hamming Window technique for FIR filter is more effective than the Butterworth method for the IIR filter at noise removal from audio signals.

V. CONCLUSION

This study provides a comparative evaluation of FIR and IIR filters for improving an audio signal corrupted by White Gaussian noise. By applying both filtering techniques to a noisy sine wave and evaluating the results through Signal-to-Noise Ratio (SNR), it was observed that both filters successfully improved signal quality. The FIR filter designed with the Hamming window provides durability and continuity of the phase of the signal, while the IIR Butterworth filter achieves effective noise suppression with low system requirements. These results suggest that the choice of filter depends on the application of the FIR filter is advantageous when accuracy and phase one-dimensionality are crucial, whereas the IIR filter is advantageous when computational efficiency is a priority. Overall, the work emphasizes that digital filtering remains a reliable approach for noise reduction and audio signal enhancement.

VI. REFERENCES

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