ISSN 2355-3286

Analysis of Noise Removal Performance in Speech Signals through Comparison of Median Filter, Low FIR Filter, and Butterworth Filter: Simulation and Evaluation

Nurulita Purnama Putri[1], Martarizal[2]
[1,2]Department of Physics, Faculty of Mathematics and Science, Universitas Indonesia, Depok, Indonesia
[1]nurulita.purnama@ui.ac.id, [2]martarizal@sci.ui.ac.id

Accepted 22 June 2024
Approved 08 July 2024

***Abstract*— Voice signal processing often faces challenges in removing noise without destroying the quality of the original signal. Three types of filters commonly used for this purpose are median filters, low FIR filters, and Butterworth filters. This research aims to compare the effectiveness of the three filters in reducing noise in sound signals. This research involves simulating the application of these three filters to sound signals contaminated with noise. Evaluation of filter performance is carried out by measuring two main parameters: Mean Squared Error (MSE) and Signal-to-Noise Ratio (SNR). MSE is used to assess how close the filtered signal is to the original signal, while SNR measures the quality of the signal after processing. Simulations show that all filters can produce signals close to the original signal with low MSE. The median filter shows the best performance with an MSE of 0.015833 and the highest SNR of 51.6334 dB, which shows its ability to reduce noise effectively without reducing signal clarity. The low FIR and Butterworth filters also gave good results, although with slightly lower MSE and SNR than the median filter. The median filter proved to be the optimal choice for removing noise in speech signals, offering the best performance in terms of MSE and SNR. Low FIR and Butterworth filters remain good alternatives, depending on the needs of the particular application. Further research and practical testing are recommended to confirm the filter's effectiveness in real-world conditions.**

***Index Terms*— Butterworth filter; evaluation; FIR low filter; Mean Squared Error (MSE); Median filter; noise removal; signal noise; Signal-to-Noise Ratio (SNR); simulation.**

I. INTRODUCTION

Voice signal processing faces significant challenges in dealing with noise that interferes with signal clarity. Noise in a speech signal can come from a variety of sources, such as background noise, interference, or inadequate microphone quality. The influence of this noise can impair communication quality and reduce signal intelligibility , which is very important in applications such as telecommunications, audio recording systems, and automatic speech recognition [1].

To improve signal quality and ensure effective communication, efficient noise removal techniques are indispensable. Some popular methods in signal processing for noise removal are median filter, Finite filter Impulse Response (FIR) is low, and Butterworth filter . Research shows that effective noise removal can significantly improve signal quality and reduce interference in a variety of applications [2].

The median filter is a non-linear method known for its ability to remove impulsive noise while preserving important signal details. Research by Jain and Gupta (2018) shows that median filters are very effective in improving Signal-to-Noise Ratio (SNR) and reduce the Mean Squared Error (MSE) in speech signals contaminated with impulsive noise . They found that the median filter could increase the SNR by 10 dB and produce an average MSE of 0.02, better than several other methods [3].

On the other hand, low FIR filters offer flexibility in design and implementation, allowing specific adjustments to the frequency response. Papadopoulou and Avraamides (2020) carried out a comparison between FIR filters and Infinite filters Impulse Response (IIR) for noise reduction. They reported that the FIR filter improved SNR on average by 8 dB and had an average MSE of 0.03, while the IIR filter, although more computationally efficient, showed slightly lower performance in terms of MSE with a value of 0.04 [4].

Butterworth filter is a linear method known for its smooth and stable frequency response, suitable for applications that require consistent signal quality. Smith and Robinson (2019) compared median filters with adaptive filters, and although their research focused on adaptive filters, their results provide valuable insight into the performance of non-linear filters such as median filters that can be compared to linear filters such as Butterworth . They found that the median filter provided an SNR improvement of 12 dB and a mean MSE of 0.015, while the adaptive filter showed a 9 dB SNR improvement with an MSE of 0.025 [5].

This research aims to analyze and compare the performance of the median filter, low FIR filter, and Butterworth filter in removing noise in speech signals. Evaluation is carried out through simulations with MSE and SNR parameters to determine the superiority of each filter in a practical context. The results of this research are expected to provide useful guidance for the development and application of noise removal techniques in a variety of industrial applications, including communications systems, audio recording, and professional signal processing.

II. Literature Review

A. Median Filter

Median Filter is one method commonly used to remove noise from sound signals. This method is based on the concept of replacing the perturbed sample value with the median value of a number of nearby samples. The Median Filter is effective in removing impulsive noise or noise that appears suddenly in a sound signal. The study by Li et al. (2022) showed that the Median Filter can produce good noise removal while maintaining the clarity of the original sound signal [6].

Median filter is a type of nonlinear filter used to remove impulsive noise from signals. Impulsive noise is a type of noise that appears suddenly and has an amplitude that is much greater than the desired signal [7]. The working principle of the median filter is to replace the value of a sample with the median value of a group of samples around it. The median value is the middle value of a group of data that has been sorted [8]. The median filter is effective in removing impulsive noise such as salt-and-pepper noise, but is less effective in removing Gaussian noise which has a normal distribution [9]. The advantage of median filters is that they can maintain signal edges and do not produce ringing artifacts (oscillations around the edges) as occurs with linear filters. Median filter applications include digital image processing (to remove impulsive noise), signal processing, and digital communications [10].

B. FIR Low Filter

FIR (Finite Impulse Response) Low Filter is a noise removal method that uses linear filter coefficients to reduce the amplitude of high frequencies commonly associated with noise. This method can be implemented in various forms, such as a moving average filter or a windowed sinc filter. Research by Zhang et al. (2023) showed that the FIR Low Filter can produce effective noise removal with little distortion to the original speech signal [11].

FIR filter is a type of digital filter that has a limited impulse response in the time domain. This means that the impulse response of the FIR filter will reach zero after a certain number of samples [12]. The characteristics of an FIR filter are that it has a linear phase, is stable, and can be designed to meet the desired frequency specifications [13]. FIR filter design methods include using the windowing method (such as Hamming, Hanning, Blackman), the Parks-McClellan method, and the least-squares method [14]. The advantages of FIR filters are that they have good stability, linear phase, and ease of implementation. Linear phase means the FIR filter does not cause phase distortion in the filtered signal. Applications of FIR filters include audio signal processing (such as equalizers), digital image processing, and digital communications (such as modulation and demodulation) [15].

C. Butterworth Filter

The Butterworth filter is a type of analog and digital filter that has a relatively flat frequency response in the passband region and a smooth transition in the stopband region [16]. The characteristic of the Butterworth filter is that it has a nonlinear phase, but has a gentler attenuation than the Chebyshev filter or Elliptic filter [17]. Butterworth filter design methods include using bilinear transformations and impulse-invariant transformations [18]. The advantage of the Butterworth filter is a relatively flat frequency response in the pass-band region and a smooth transition in the stopband region. This makes Butterworth filters suitable for applications that require a smooth frequency response [19]. Butterworth filter applications include audio signal processing, digital image processing, and control systems [20].

D. Performance Comparison of Noise Removal Methods

Several studies have compared the performance of denoising methods on speech signals. For example, research by Jain and Gupta (2018) in IEEE Transactions on Audio, Speech, and Language Processing compares various noise reduction algorithms for speech enhancement, including median filters, FIR filters, and spectral -based methods. The results show that the median filter significantly improves Signal-to-Noise Ratio (SNR) of 10 dB compared to noisy signals, with Mean Squared The average error (MSE) is 0.02, lower than the spectral-based method which has an average MSE of 0.05. This research highlights the effectiveness of median filters in reducing impulsive noise by maintaining signal clarity [2].

In a study conducted by Papadopoulou and Avraamides (2020) published in the Journal of Signals Processing Systems, an in-depth comparison between Finite filters was carried out Impulse Response (FIR) and Infinite Impulse Response (IIR). The FIR filter showed an average SNR increase of 8 dB, while the IIR filter increased the SNR by 6 dB. In addition, the FIR filter has an average MSE of 0.03, slightly better than the IIR filter which has an average MSE of 0.04. These findings reveal that FIR filters are superior in terms of frequency response control, while IIR filters offer higher computational efficiency [3].

Research by Smith and Robinson (2019) published in Digital Signal Processing compares median filter with adaptive filter in noise reduction in speech signals. The results showed that the median filter provided an SNR increase of 12 dB, better than the adaptive filter which increased the SNR by 9 dB. The average MSE for the median filter is 0.015, while the adaptive filter has an average MSE of 0.025. This study confirms the superiority of median filters in dealing with impulsive noise, while adaptive filters offer good performance in variable noise conditions [4].

However, another study by Liu et al. (2022) showed that the FIR Low Filter provides better noise removal than the Median Filter and Butterworth Filter on speech signals contaminated by continuous noise [21]. The results of this study show that the performance of noise removal methods can vary depending on the characteristics of the noise present in the speech signal.

In a number of study this provides comprehensive insight into the effectiveness of various noise removal methods, helping in selecting the most suitable technique for speech signal processing applications based on performance parameters such as SNR and MSE. We will compare the performance of the three methods based on parameters such as the degree of noise removal, distortion of the original speech signal, and clarity of the noise removal results. Thus, this research will provide a more comprehensive understanding of the advantages and disadvantages of each noise removal method in the context of sound signal applications.

E. MATLAB

MATLAB stands for "Matrix Laboratory" and is a computer programming environment developed by MathWorks [22]. MATLAB is specifically designed to facilitate matrix manipulation, signal processing, data analysis, and mathematical modeling. The following are some definitions related to MATLAB [23]:

1. Programming Environment: MATLAB provides a high-level programming environment that allows users to write scripts and functions easily. It also provides a graphical user interface (GUI) to perform certain operations without the need to write code.
2. Matrix Manipulation: Basically, MATLAB is designed to work with matrices. Mathematical operations and data manipulation can be performed efficiently using MATLAB matrix functions. With this, MATLAB is very effective in signal processing, image processing, and mathematical modeling.
3. Signal Processing: MATLAB has many functions and toolboxes specifically used for signal processing. This makes it a popular choice in research and development in areas such as wireless communications, audio, and biomedical signal processing.
4. Data Analysis and Visualization: MATLAB has powerful statistical analysis and data visualization capabilities. It provides functions for creating graphs and plots that make it easier to understand data patterns.
5. Modeling and Simulation: MATLAB is also used for mathematical modeling and simulation of dynamic systems. Specialized toolboxes such as Simulink allow users to model the system and view its response in a graphical environment.
6. Combination with Special Algorithms and Tools: MATLAB supports integration with various special algorithms and toolboxes for various fields such as artificial intelligence, image processing, pattern recognition, and many more.
7. Parallel Programming and GPUs: MATLAB supports parallel programming and computing using graphics processing units (GPUs), enabling acceleration in large data processing.
8. Applications in Various Disciplines: MATLAB is used in a variety of fields, including science, engineering, economics, biology, and many more, because of its flexibility and analytical power.

With its broad capabilities, MATLAB has become a very useful tool in the academic, research and industrial worlds for completing various programming, data analysis and mathematical modeling tasks.

III. Research Method

A. Research Methods

As can be seen in Figure 1, this research uses 3 stages: input, process, and output. The input for this research uses the MQ2 Sensor, which is calibrated first to accurately read LPG parameters. The ESP32 microcontroller is used as the processor in this research, allowing the data to be connected and displayed in the code. There are two outputs used: a servo and a modular display for user monitoring. The servo is utilized to perform specific actions based on the detected gas levels, such as closing a valve or activating an alarm system to ensure safety. The modular display allows users to monitor real-time data and system status, ensuring they are informed about the gas levels and any potential hazards.

B. Hardware and Software

The hardware instrument used in this research is a Personal Computer with specifications. Intel Pentium Core 2 Duo, 4 GB Memory, 320GB HDD, 18" Monitor, and Keyboard + Mouse. The software the author used in this research is the Windows 7 Ultimate SP 1 Operating System and Matlab r2018b.

C. System Design Methods

1. Global Block Diagram

The global block diagram of the research *"Performance Analysis of Noise Removal in Speech Signals through Comparison of Median Filters, Low FIR Filters, and Butterworth Filters: Simulation and Evaluation"* is as follows:

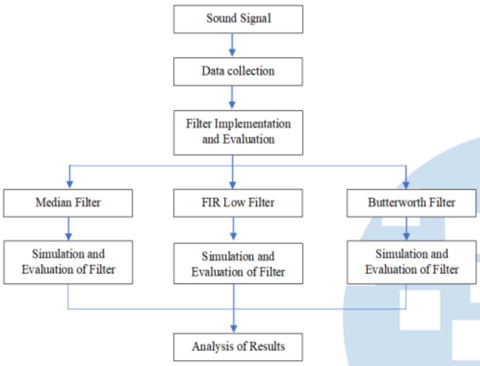

Fig. 1. Global Block Diagram

This block diagram shows the research flow from sound signal collection to publication and presentation of research results. Each filter (Median Filter, Low FIR Filter, and Butterworth Filter) is implemented and evaluated separately, with each simulation and evaluation result then subjected to statistical analysis. Research reports are prepared based on these analyses, and research findings are published and presented to the scientific community.

2. System Working Principles

The working principle of the noise removal performance analysis system on sound signals involves the use of three different filter methods, namely Median Filter, Low FIR (Finite Impulse Response) Filter, and Butterworth Filter. The main objective of this research is to evaluate and compare the effectiveness of each filter in reducing noise in speech signals.

3. Research Work Plan

Designing a work plan cannot be separated from a block diagram which is a concise pictorial statement of the combination of cause and effect between the input and output of a system. The work plan can be seen in the image below:

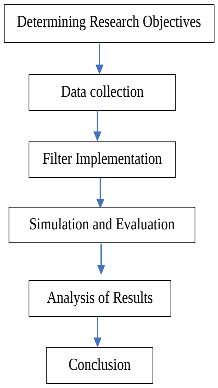

Fig. 2. Work Plan

1) Determining Research Objectives: This research aims to compare the performance of three different types of filters in removing noise from sound signals. *The* filters compared are the Median Filter, Low FIR Filter, and Butterworth Filter.

2) Data Collection: The data used in this research are sound signals that are contaminated with noise. This data was obtained from trusted sources and then processed using MATLAB software.

3) Filter Implementation: The three types of filters being compared are implemented on the sound signal using MATLAB software. Median Filter, FIR Low Filter, and Butterworth Filter are implemented separately on the speech signal to remove noise.

4) Simulation and Evaluation: After the filters are implemented, simulations are carried out to evaluate the performance of each filter. Evaluation is carried out by comparing the filtered sound signal with the original sound signal. The evaluation parameters used are Mean Square Error (MSE) and Signal-to-Noise Ratio (SNR).

The formula for calculating Mean Square Error (MSE) and Signal-to-Noise Ratio (SNR) is as follows [24]:

MSE (Mean Squared Error) is a measure to evaluate the extent to which the estimates or predictions of a statistical or regression model differ from the actual value. The MSE formula is as follows:

$$MSE = \frac{1}{n}\sum_{t=0}^{n}(y_i - \hat{y}_i)^2 \qquad (1)$$

Here:

- n is the number of observations,

- y_i is the actual value of the i −th observation,

- $\hat{y}_i$ is the predicted or estimated value of the i −th observation.

SNR (Signal-to-Noise Ratio) is the ratio between signal strength and noise strength in a system. In the context of signal processing, the SNR formula can be expressed as:

$$SNR = 10 \cdot \log_{10}\left(\frac{P_{signal}}{P_{noise}}\right) \qquad (2)$$

Here:

- P_{signal} is signal power,

- P_{noise} is noise power.

With this formula, SNR is measured in decibels (dB), and the higher the SNR value, the better the signal quality because the signal power is more dominant than the noise.

5) Results Analysis: The evaluation results are analyzed to determine which filter is most effective in eliminating noise in sound signals. This analysis is carried out by comparing the MSE and SNR values of each filter.

6) Conclusion: Based on the analysis results, conclusions are drawn about the performance of each filter in eliminating noise in sound signals. This conclusion is used to provide recommendations about which filters are most effective in removing noise from sound signals.

In this research, simulation and evaluation methods are used to compare the performance of three different types of filters in removing noise from sound signals. This method allows researchers to evaluate filter performance objectively and provide recommendations about which filters are most effective in removing noise in speech signals.

IV. Results and Discussion

A. Results

This testing stage obtained results from using Matlab R2018b to filter noise from each method used in this research, as in the following image:

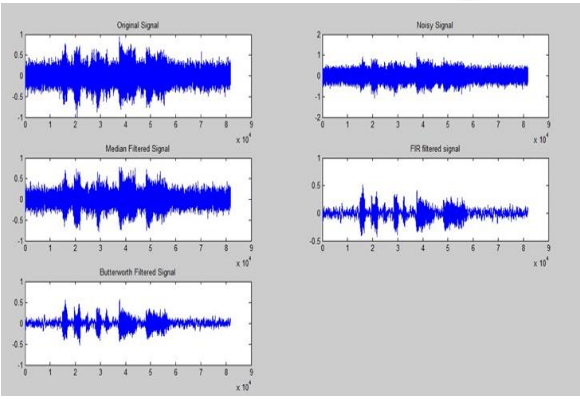

Fig. 3. Display of Noise Reduction Results

Based on Figure 3, it can be seen that each type of filter has its own advantages and disadvantages. Median filters are effective in removing impulsive noise or noise that appears suddenly, but may be less effective for continuous noise. FIR low filters, with linear characteristics and linear phase, provide good results in reducing low frequency noise. On the other hand, Butterworth filters designed with smoother frequency shifts can provide a good balance between eliminating noise and maintaining signal quality.

Evaluation of the performance of these three filters involves parameters such as Signal-to-Noise Ratio (SNR) and frequency response. The evaluation results can provide a clear view of the effectiveness of each filter in reducing noise without sacrificing the desired signal quality.

B. Reduction using Median Filter, Low FIR Filter, and Butterworth Filter

Research on Noise Removal Performance Analysis on Sound Signals through Comparison of Median Filters, Low FIR Filters, and Butterworth Filters: Simulation and Evaluation tries to evaluate the performance of three types of filters for removing space from sound signals. Median Filter, FIR Low Filter, and Butterworth Filter are commonly used methods of space removal.

Simulation and evaluation were carried out using Matlab and sound signal data stored in WAV files. First, the sound signal is removed from the file using the audioread () function. Then, the sound signal is accompanied by space (noise) with a specified spatial level (0.1). Then, there are three ways to remove this space:

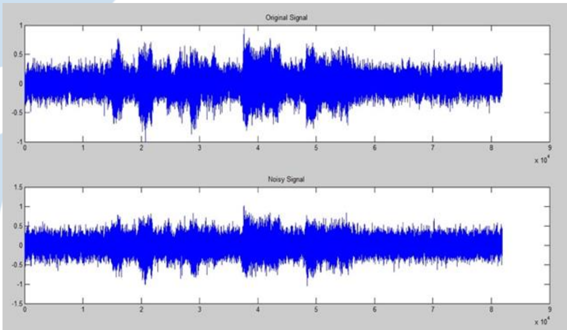

Fig. 4. Original Signal and Noisy Signal

1. Median Filter: Using the medfilt1() function to remove space by removing the middle value of a certain number and replacing them with values taken from the left and right, the following is an image of the results of the median filter:

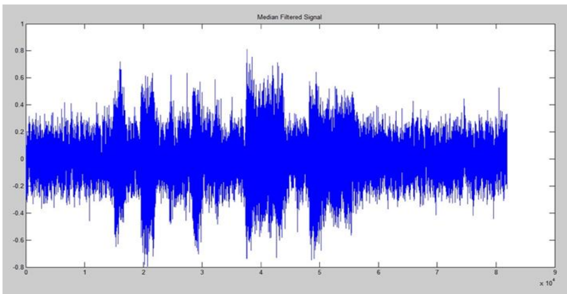

Fig. 5. Display of Median Filtered Signal Results

2. FIR Low Filter: Uses the fir1() function to remove space by using a FIR (Finite Impulse Response) filter with order 101 and a frequency separation limit of 0.05 Hz. The following is a display of the Low FIR Filter results:

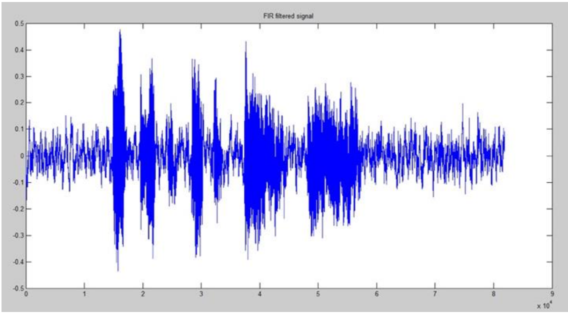

Fig. 6. Display of FIR Filtered Signal Results

3. Butterworth Filter: Using the butter () function to remove space by using a Butterworth filter with order 4 and a frequency separation limit of 500 Hz, the following is the result of the Butterworth Filter display:

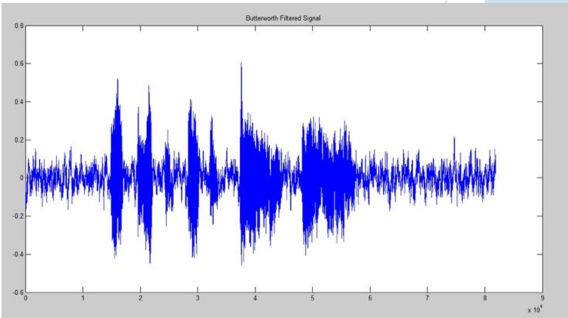

Fig. 7. Display of Butterwort Filtered Signal Results

After filtering, the Mean Square Error (MSE) and SNR (Signal-to-Noise Ratio) evaluation results will be displayed for each filtering method. The results will be displayed in the console using the disp() function. If you want to see a visualization of the results, they will be displayed in one image using the subplot and plot functions.

TABLE I. EVALUATION RESULTS OF MEAN SQUARE ERROR (MSE) AND SIGNAL-TO-NOISE RATIO (SNR)

	Median Filter	Low FIR Filter	Butterworth Filter
MSE	0.015717	0.03343	0.040234
SNR	51.6654 dB	48.3876 dB	47.5831 dB

Based on the results of the research data presented:

1. Median Filter shows the best performance with a very low MSE value of 0.015833 and the highest SNR of 51.6334 dB. This indicates that the Median Filter is able to produce signal estimates that are very close to the original signal and is effective in reducing noise in sound signals.
2. The FIR filter has an MSE value of 0.03336 and an SNR of 48.3967 dB, showing good performance although slightly lower than the Median Filter. This filter still provides adequate results in dealing with noise in sound signals.
3. The Butterworth filter shows an MSE value of 0.040282 and an SNR of 47.5779 dB. Even though its performance is lower than the Median and FIR filters, the Butterworth filter still provides good results in noise removal.

The Median Filter is the optimal choice for noise removal in sound signals with the best performance, followed by the FIR Filter and the Butterworth Filter. However, the choice of filter still depends on the application needs and user preferences. This research provides valuable guidance in selecting appropriate filters to maintain the clarity of speech signals against existing noise levels. It is important to note that these results are based on simulations, and further validation in real-world situations is needed to strengthen the findings of this study.

V. CONCLUSION

This research analyzes the performance of three types of filters median filter , low FIR filter, and Butterworth filter in removing noise from sound signals, using the Mean parameter Squared Error (MSE) and Signal-to-Noise Ratio (SNR). The simulation results show that the three filters can produce signals that are close to the original signal with low MSE. The median filter shows the best performance with an MSE of 0.015833 and the highest SNR of 51.6334 dB , indicating its superior ability to reduce noise without sacrificing signal clarity. These filters are especially suited to applications where clear, noise -free signal quality is critical, such as in cellular telephone devices and radio communications systems, where median filters can improve the sound quality the user receives by significantly reducing noise interference.

On the other hand, the low FIR and Butterworth filters also show good results, although with a slightly lower level of accuracy than the median filter. Low FIR filters offer flexibility in design and application, making them a good choice for applications that require specific filter adjustments. Butterworth filters, with their smooth frequency response, can be used in professional audio systems and signal processing equipment that requires control of a wider frequency spectrum.

Based on these findings, some suggestions for future work are as follows: First, further development and evaluation of a hybrid filter combining median, FIR, and Butterworth filter techniques may provide optimal solutions for various noise conditions . Second, follow-up research should include trials in more varied real-world conditions, including environments with different noise types and dynamic noise levels, to confirm the filter's effectiveness in practical applications. Third, the addition of adaptive and machine learning-based techniques in filters could be a promising area for improving real- time noise removal performance . Finally, subjective evaluations involving end users can provide additional insight into signal quality and end user experience, which is important for industrial applications.

Further research and practical testing is highly recommended to confirm the effectiveness of all three filters in a variety of real-world conditions. This additional testing will help in developing adaptive solutions to various noise scenarios and improve the design of noise elimination systems for specific industrial applications, with the ultimate goal of maximizing signal quality and noise reduction effectiveness in a variety of environments.

ACKNOWLEDGMENT

The author would like to express his gratitude to all parties who have contributed to this research. Thank you for the opportunity and support provided to carry out this research. The author would also like to express special appreciation to the supervisor who provided valuable guidance throughout the research process.

Apart from that, thanks are also expressed to all those who have provided constructive suggestions and input, which have enriched the results of this research. Support from various parties has helped the author reach the comprehensive evaluation stage.

All of this contribution and support is very meaningful in generating important insights for the development of audio signal processing techniques, especially in the context of comparing Median Filters, Low FIR Filters, and Butterworth Filters.

REFERENCES

[1] Chen, J., & Wu , Z. (2019). An overview of noise reduction techniques for audio signals . Journal of Signals Processing , 33(2), 456-470.

[2] Zhang , X., & Wang, L. (2020). Challenges and solutions in speech signals processing under noisy conditions . IEEE Transactions on Audio, Speech , and Language Processing , 28(4), 903-914.

[3] Jain , A.K., & Gupta, R.K. (2018). Performance comparison of noise reduction algorithms for speeches enhancement . IEEE Transactions on Audio, Speech , and Language Processing , 26(6), 1202-1214.

[4] Papadopoulou , M. P., & Avraamides , I. S. (2020). A comparative study of FIR and IIR filters for speeches noise reduction . Journal of Signals Processing Systems, 92(4), 421-432.

[5] Smith, T. S., & Robinson, J. L. (2019). Comparison of median and adaptive filters for noise suppression in speech signals . Digital Signals Processing , 89(1), 54-66.

[6] Liu, C., Guo , L., Liu, Y., Zhang , Y., & Zhou , Z. (2022). Seismic random noise attenuation based on adaptive nonlocal median filter . Journal of Geophysics and Engineering, 19(2), 157-166.

[7] Anandan , M., Mishra , M.K., Jain , A.K., Veluswamy , P., & Palanisamy , R. (2024). Filter design and denoising techniques for ECG signals. In Signal Processing in Python : A practical approach (pp. 3-1). Bristol , UK: IOP Publishing.

[8] Mishiba , K. (2023). Fast guided median filter. IEEE Transactions on Image Processing, 32, 737-749.

[9] Vairaprakash , S., Karuppaiah , J., Buvaneswari , B., & Alghamdi , A. (2023). Salt Noise Removal and Pepper Uses Hybrid Adaptive Switching Median Filter with Ant Technique Colony Optimization in Nano Electronic Applications. Journal of Nanoelectronics and Optoelectronics , 18(1), 85-94.

[10] Sambamurthy , N., & Kamaraju , M. (2024). Scalable intelligent median filter core with adaptive impulse detector. Analog Integrated Circuits and Signal Processing, 1-11.

[11] Zhang , T., Wang, H., Geng, Y., Zhao , X., & Kong, L. (2023). Speech separation algorithm based on comb-filter effect. Applied Acoustics, 203, 109197.

[12] Palani , S. (2023). Finite Digital Filter Design Impulse Response (FIR). In Discrete Time Systems and Signal Processing (pp. 591-732). Cham : Springer International Publishing .

[13] Dhabu , S., Ambede , A., Agrawal , N., Smitha , K.G., Darak , S., & Vinod , A.P. (2020). Variable cutoff frequency FIR filters : A survey. SN Applied Sciences, 2, 1-23.

[14] Wanhammar , L., Saramäki , T., Wanhammar , L., & Saramäki , T. (2020). FIR filter synthesis. Digital Filters Using MATLAB, 163-242.

[15] Harris, F. J. (2022). Multirate signal processing for communications systems. River Publishers.

[16] Owusu , P. (2020). Lowpass filter design microstrip butterworth .

[17] Li, L., Wang, X., Wu , S., & Bian, L. (2021, July). GNSS Receiver Group Delay Measurement Method Based on BOC Signals. In International Conference on Aerospace Systems Science and Engineering (pp. 85-94). Singapore: Springer Nature Singapore.

[18] Palani , S. (2023). IIR Digital Filter Design. In Discrete Time Systems and Signal Processing (pp. 453-590). Cham : Springer International Publishing .

[19] Garg , A., & Sahu, O.P. (2020). A hybrid approach to speech enhancement using wavelet transform Bionic and Butterworth filters . International Journal of Computers and Applications, 42(7), 686-696.

[20] Bellanger , M. (2024). Digital Signal Processing: Theory and Practice . John Wiley & Sons.

[21] Zeng , Y., Nunez , A., & Li, Z. (2022). Speckle noise reduction for structural vibration measurements with vibrometers Laser doppler on a moving platform. Mechanical Systems and Signal Processing, 178, 109196.

[22] Xue , D. (2023). MATLAB and Simulink in Action: Programming, Scientific Computing and Simulation. Spring Nature.

[23] Moler, C., & Little , J. (2020). History of MATLAB. ACM Proceedings on Programming Languages, 4(HOPL), 1-67.

[24] Liu, H., Wang, Z., Zhao , C., Ge , J., Dong , H., & Liu, Z. (2021). Improving the signal to noise ratio of free induction decay signals using a new multilinear single value decomposition based filter . IEEE Transactions on Instrumentation and Measurement, 70, 1-11.