Scope
Volume 15 Number 02 June 2025

Optimizing Signal and Image Processing: A Comprehensive Approach to Filter Design for Quality Enhancement

[1]Md. Romzan Ali, [2]Abdullah Al Zubaer, [3]Md. Shamim Imtiaz, [4]Sunjida Akter, [5]Md. Nazrul Islam, [6]Ramjan Ali, [7]Shohag Barman
[1,6]Department of Electrical and Electronics Engineering, Lalon University of Science and Arts, Kushtia, Bangladesh
[2,3,7]Department of Computer Science and Engineering, Pirojpur Science and Technology University, Bangladesh
[4]Department of Educational Technology and Engineering, Gazipur Digital University, Gazipur, Bangladesh
[5]Department of Computer Science and Engineering, Islamic University, Kushtia, Bangladesh

Corresponding Authors: **Abdullah Al Zubaer**

Abstract: This paper delves into digital filter design using the window function method, with emphasis on the superior capabilities of MATLAB for rapid and effective filter creation. Demonstrated through the design of a bandpass filter applied to a mixed sine wave signal, MATLAB's proficiency is highlighted. A simulation model, implemented in MATLAB, validates the filter's performance through observation of waveforms on Oscilloscopes. The paper further explores digital filters with complex coefficients, presenting a theory grounded in low-pass analog prototypes and digital design techniques. Comparative discussions on real and analytic signal processing reveal analogous signal operation requirements, emphasizing the efficiency of both approaches in processing information.

Key Words: Low-pass Filter, FIR filter, Band-pass Filter, SNR, Image, Operator, Derivation, Surface.

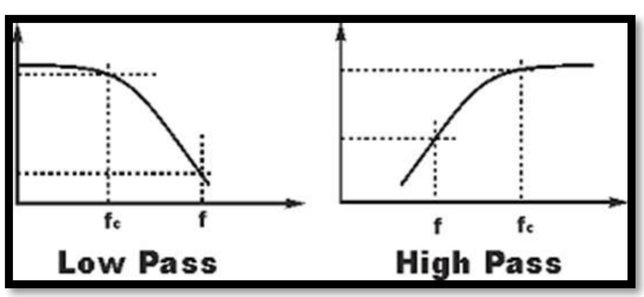
Introduction

Electronic filters are circuit components designed for signal processing, aiming to eliminate undesired frequency components, enhance desired ones, or achieve both objectives. In computer programming, a filter refers to a program or code segment responsible for scrutinizing input or output requests based on specific criteria, then processing or forwarding them accordingly [1]. Whether in electronic circuits or software, filters serve to refine and manage signals or data streams by selectively allowing or blocking certain elements based on predefined conditions. Electronic filters are circuits which perform signal processing functions, specifically to remove unwanted frequency components from the signal, to enhance wanted ones, or both.In computer programming, a filter is a program or section of code that is designed to examine each input or output request for certain qualifying criteria and then process or forward it accordingly.

Filters are electronic circuits designed for signal processing, with the primary goal of selectively eliminating undesirable frequency components from the signal. These components, often referred to as noise or interference, can distort the original signal. Filters play a crucial role in various applications, ensuring that signals are refined and tailored to meet specific requirements by allowing only the desired frequency range to pass through. The design and configuration of filters are pivotal in achieving optimal signal quality, making them essential components in electronic systems for mitigating noise and enhancing signal integrity [2].
Electronic filters can be:

- Passive or active
- Analog or digital
- High-pass, low-pass, band-pass, band-stop (band-rejection; notch), or all-pass.
- Discrete-time (sampled) or continuous-time
- Linear or non-linear
- Infinite impulse response (IIR type) or finite impulse response (FIR type),

Background Study
In modern tools, a low-pass filterselectively permits signals below a specified cutoff frequency while diminishing higher-frequency signals [3]. The filter's frequency response is contingent on its design, and it's alternatively known as a high-cut or treble-cut filter in audio contexts. Used in audio crossovers, LPFs remove high-frequency content, directing signals to low-frequency subwoofer systems. Conversely, high-pass filters allow high frequencies and attenuate lower frequencies [4]. Band-pass filters allow a specific frequency range, while band-stop filters reject a specific range. Notch filters narrow down a band-stop, and all-pass filters permit all frequencies while influencing phase based on frequency in figure 1.



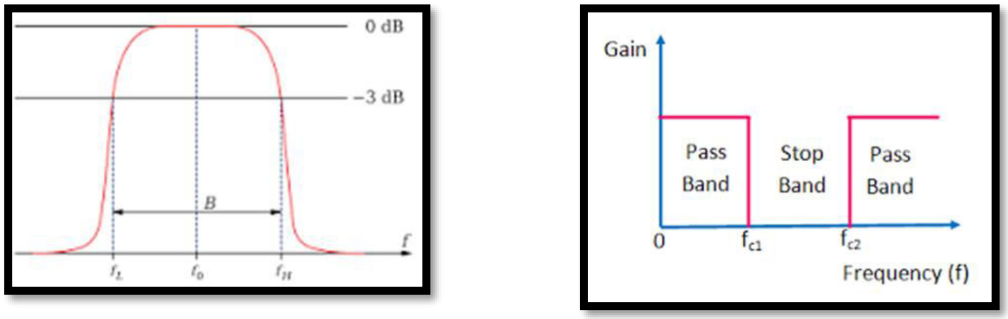


Figure 1: Low-pass, High-pass, Band-pass and Band-stop filter [Source: electronicsforu.com]

Steps for Designing

The following are several steps of designing a digital filter-

- Make sure of the property of a digital filter according to the given requirements.
- Use a discrete linear time-invariant system function to approach to the properties.
- Make use of algorithms to design the system function.
- Use a computer simulation or hardware to achieve it.

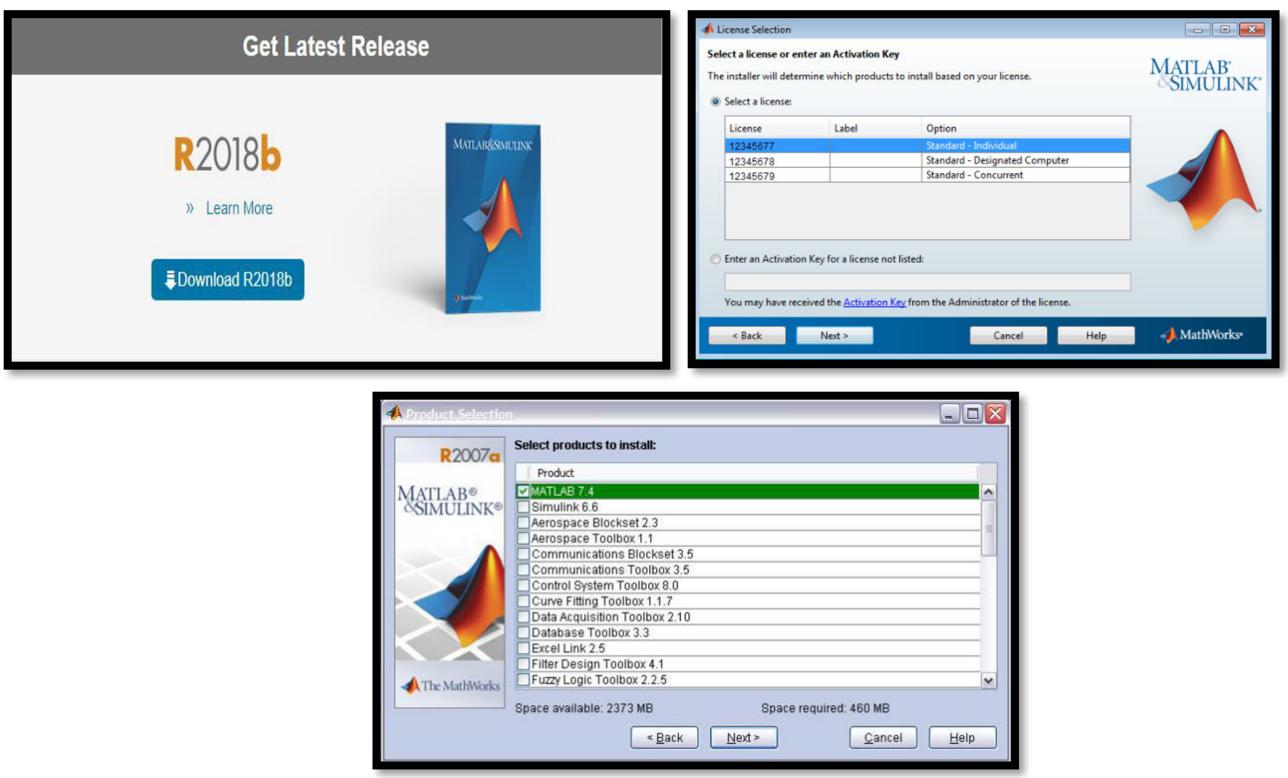


Figure 2: MATLAB [Source: Internet]

Methodology

Design of fixed frequency FIR filter

Essentials of Frequency-Selective Filter Design: Defining Pass-Bands, Stop-Bands, and Transition Bands for Uninterrupted and Attenuated Frequencies in fig 3. Acknowledging Variable Magnitude in Pass-Bands and Tolerating Nonzero Values in Stop-Bands for Practical Implementation [5]. The frequency range is meticulously split into pass-bands, stop-bands, and transition bands. Practical filters allow slight ripple in the pass-band and a small, nonzero response in the stop-band. Visualizing these variations is evident in the accompanying illustration.

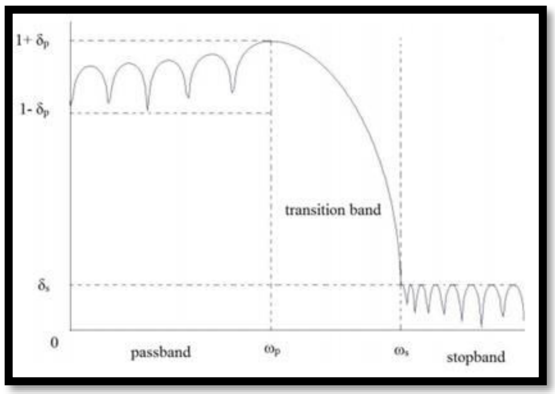


Figure 3: Fixed frequency FIR filter[Source: Internet]

Design Low-pass Filter

A low-pass filter (LPF) is a type of electronic filter designed to permit signals with frequencies below a specified cutoff frequency while attenuating higher-frequency signals shown Magnitude and Phase response of Low-pass filter in figure 4 and observation in table 1. It allows low-frequency components to pass through, effectively filtering out higher-frequency content. This filter is commonly used in various applications, such as audio systems, where it allows bass frequencies to pass while reducing or eliminating higher-frequency noise or interference. Low-pass filters play a crucial role in signal processing, offering a means to control the frequency content of a signal based on specific design parameters [6].

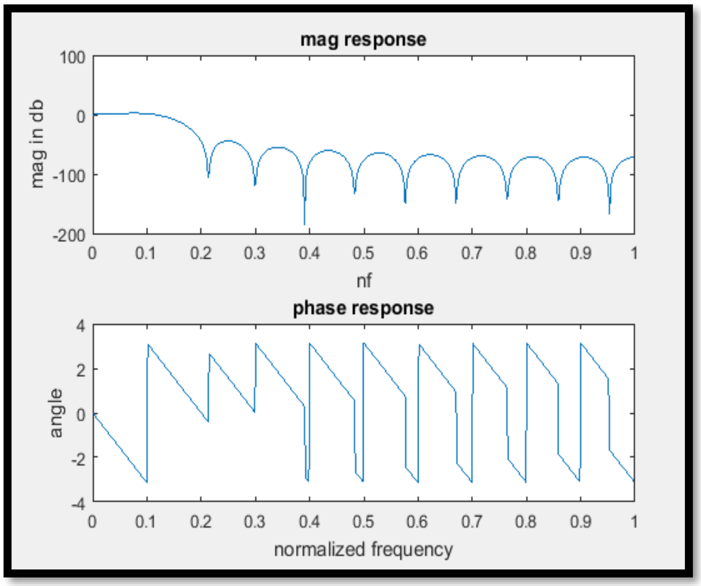
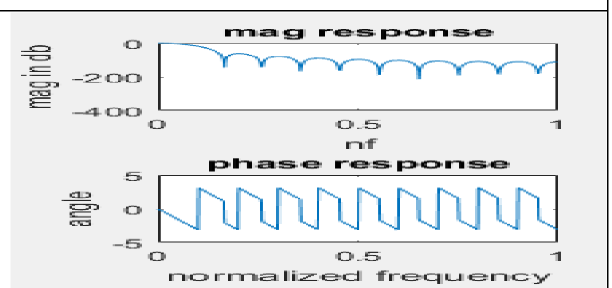
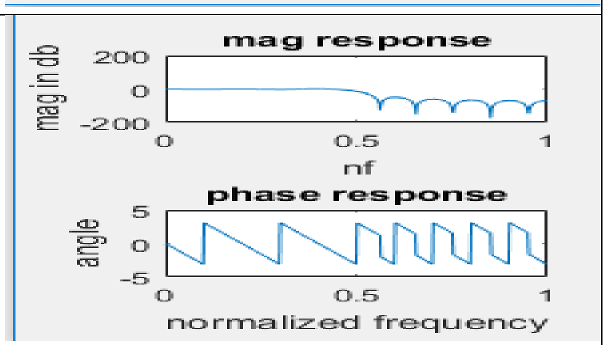
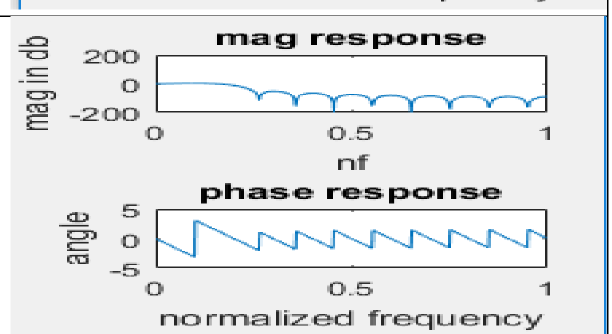


Figure 4: Magnitude and Phase response of Low-pass filter

Observation	n	fp	fs	f	Signal
1	20	500	6000	10000	
2	20	500	800	2000	
3	20	400	600	4000	

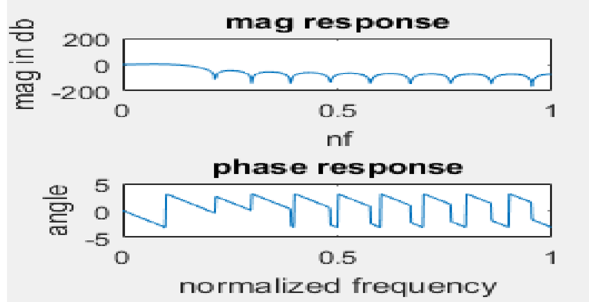
4	20	400	600	5000	

Table 1: Observationof Magnitude and Phase response of Low-pass filter

From the above observation low frequencies are passed, high frequencies are attenuated.

Design High-pass Filter

A high-pass filter (HPF) is an electronic component crafted to enable signals with frequencies surpassing a designated cutoff frequency, while suppressing lower-frequency signals. In contrast to a low-pass filter, it selectively permits higher-frequency components to pass through while blocking lower frequencies shown figure 5 and observation in table 2.

High-pass filters find widespread use in applications like audio systems, where they facilitate the passage of treble frequencies while mitigating low-frequency noise. These filters play a crucial role in signal processing, providing precise control over frequency content based on tailored design parameters, making them indispensable tools in various technological applications [7].

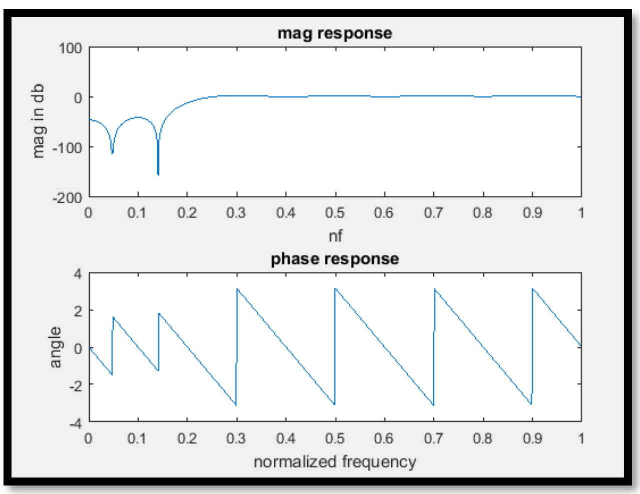


Figure 5: Magnitude and Phase response of High-pass filter

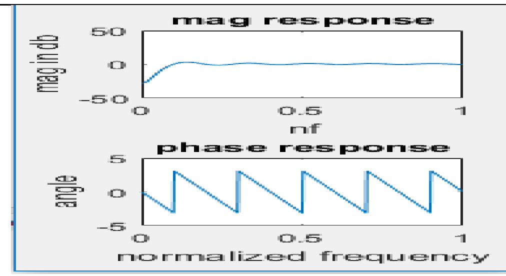
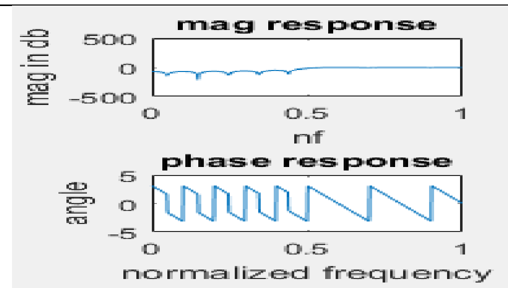
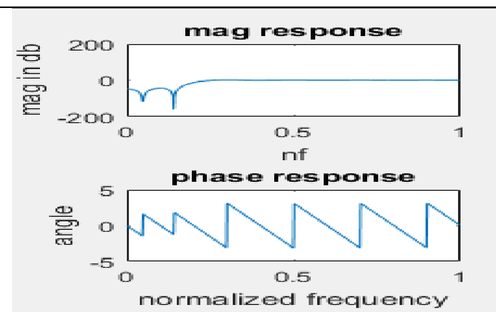
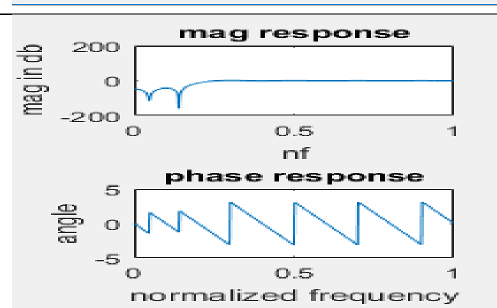
Observation	n	fp	fs	f	Signal
1	20	200	6000	10000	
2	20	250	800	1000	
3	20	200	600	2000	
4	20	250	900	2500	

Table 2: Observation of Magnitude and Phase response of High-pass filter

From the above observation high frequencies are passed, low frequencies are attenuated.

Design Band-pass Filter

A band-pass filter is an electronic component engineered to enable signals within a designated frequency range, termed the passband, to pass through while suppressing frequencies outside this range. It amalgamates attributes from both low-pass and high-pass filters [8]. Widely utilized in applications like audio systems and communication devices, the band-pass filter facilitates the isolation and processing of signals within a predefined frequency band shown figure 6 and observation in table 3. Its significance lies in scenarios demanding the extraction or isolation of specific frequency components for subsequent processing or analysis in diverse technological applications.

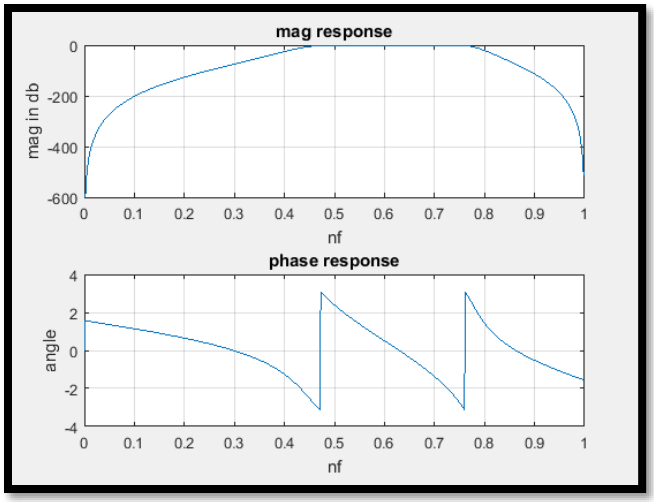


Figure 6: Magnitude and Phase response of Band-pass filter

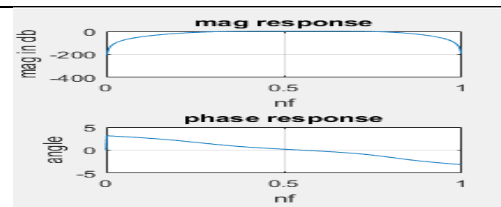
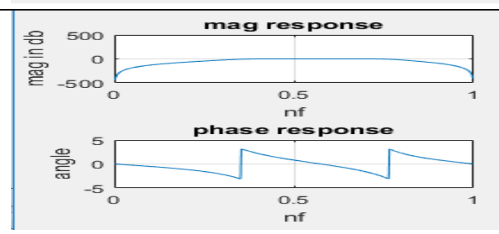
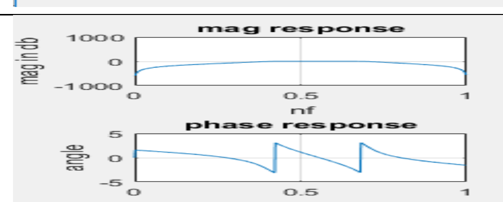
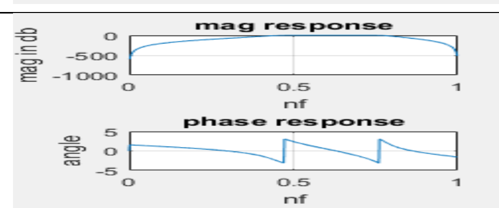
Observation	rp	rs	fp	fs	f	Signal
1	0.2	20	200	600	1500	
2	0.4	30	300	650	1700	
3	0.5	30	400	700	2000	
4	0.5	40	400	700	1800	

Table 3: Observation of Magnitude and Phase response of Band-pass filter

From the above observation only, frequencies in a frequency band are passed.

Expected Results and Discussion
Noisy Signals Filtering

Noisy signals filtering involves refining or extracting pertinent information from a signal tainted by unwanted noise. Amplitude spectrum of noisy signal shown in figure 7and Normalized frequency, magnitude response of the signal and filtered signal shown in figure 8. Factors like interference, environmental conditions, or transmission errors contribute to signal noise. Various filtering techniques, such as low-pass, high-pass, band-pass, and adaptive filters, are deployed to diminish or eliminate this noise, enabling a clearer representation of the original signal [9]. The objective is to enhance the signal-to-noise ratio, rendering the signal more valuable for analysis or subsequent processing. In essence, noisy signals filtering optimizes signal quality by mitigating the adverse effects of unwanted noise. Observation of filtered signal shown in table 4.

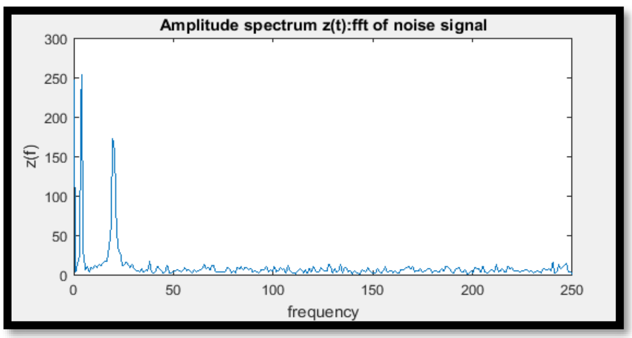


Figure 7: Amplitude spectrum of noisy signal

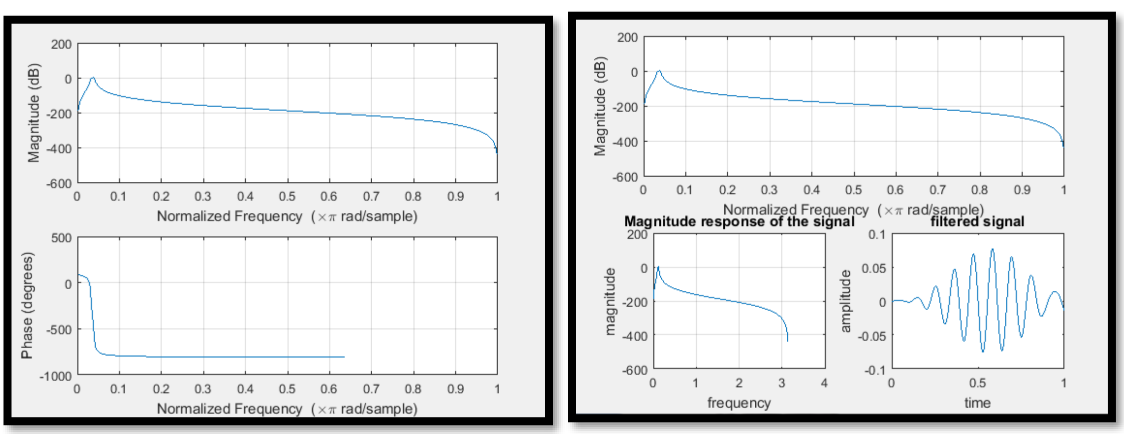


Figure 8: Normalized frequency, magnitude response of the signal and filtered signal

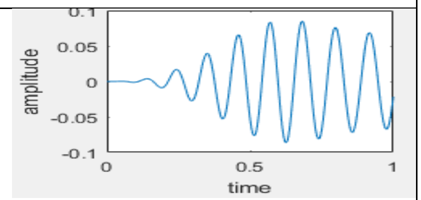
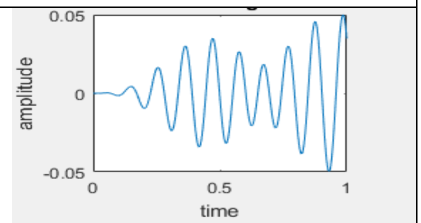
Observation	Fs	f	x	Filtered signal
1	500	20	sin(2*pi*f*n)	
2	500	20	sin(2*pi*f*n)+sin(2*pi*f*n/5)	

Table 4: Observation of filtered signal

Table 4 reveals that the filtered signal-1 allows the passage of a 20Hz signal, whereas filtered signal-2 only permits the transmission of a 20Hz signal, effectively eliminating the 2.5Hz signal.

Image Filtering

a) b)

Figure 9: a) Image before filtering b) Image after filtering

Lowpass Filtering Image

In image processing, lowpass filtering entails applying a filter to retain lower-frequency components while diminishing higher-frequency ones shown in figure 9 (Source of girls imagesare from Internet). The goal is to minimize or eliminate high-frequency noise and intricate details, yielding a smoother image. Commonly used for tasks like image smoothing, blurring, and noise reduction, lowpass filtering employs methods such as mathematical operations, convolution with a lowpass filter kernel, or frequency domain techniques like Fourier transforms [10]. This technique finds applications in image enhancement, compression, and pre-processing images for subsequent analysis or feature extraction shown in figure 10.

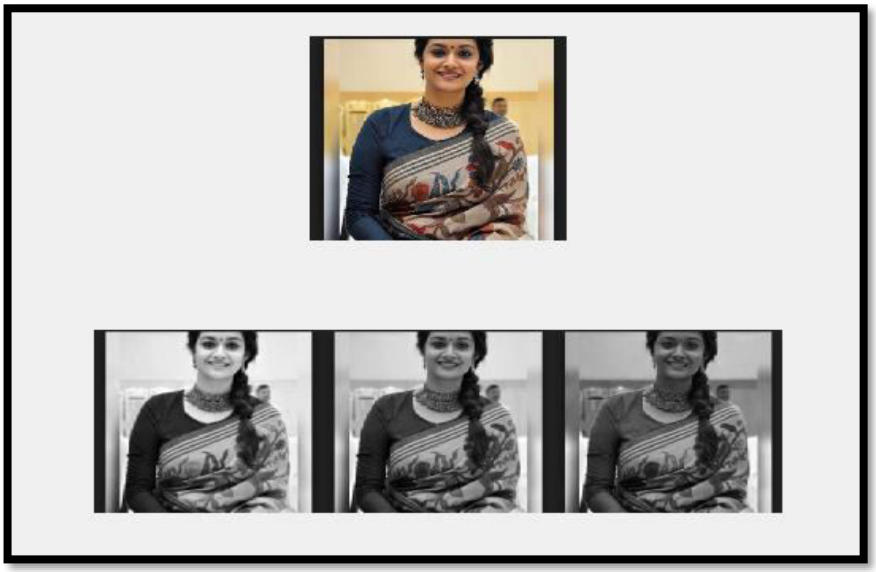


Figure 10: Lowpass filtering image

Lowpass filters in the Fourier domain selectively remove high-frequency components, allowing low frequencies to pass through. In essence, lowpass filtering results in image blurring as it emphasizes the retention of lower-frequency information while suppressing higher frequencies.

Noise Remove from Image

Removing noise from an image is a common task in image processing. Several methods can be employed, depending on the characteristics of the noise and the desired outcome shown in figure 11 and 12 (color).

Figure 11: Original and filtered images

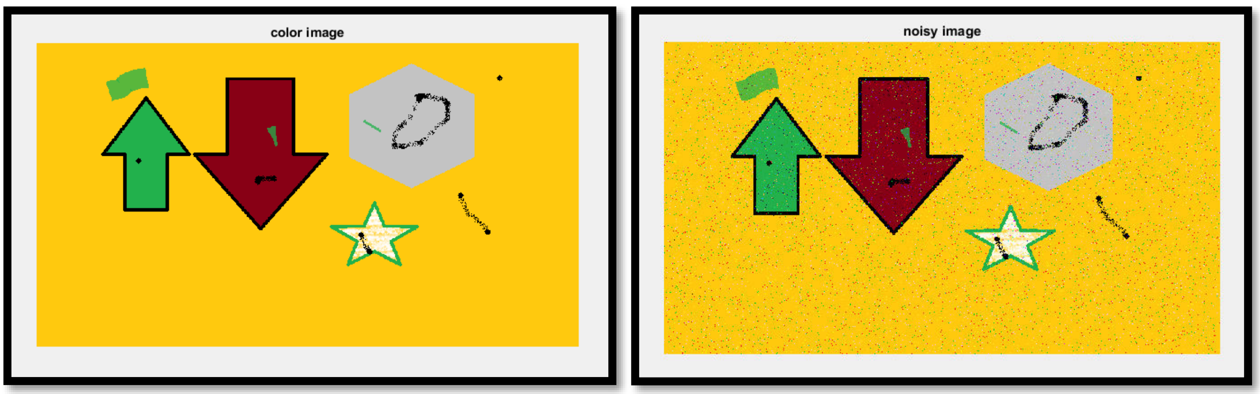


Figure 12: Colorand noisy image

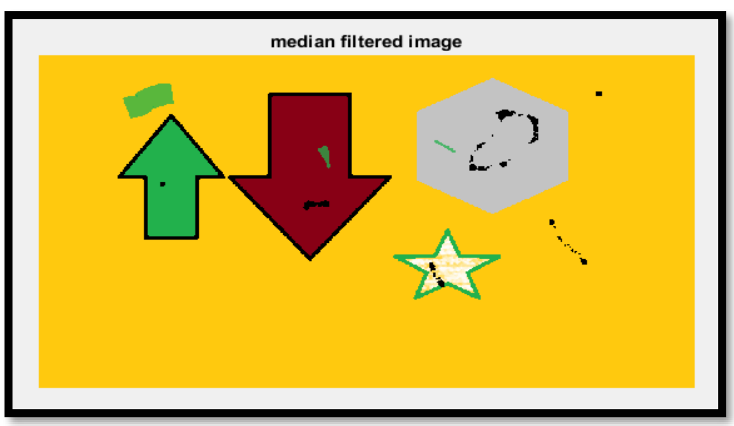


Figure 13: Median filtered image

Noise Remove without Added Salt & Pepper Noise

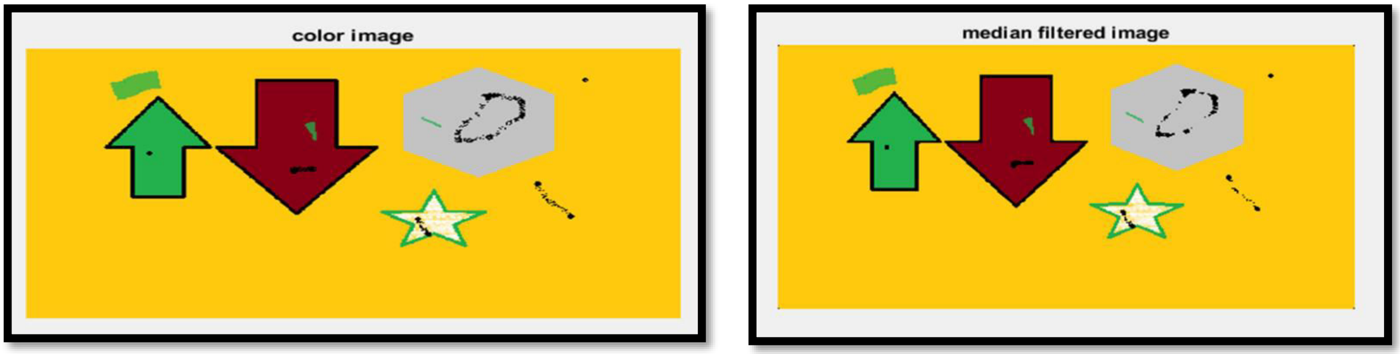


Figure 14: Color and Median filtered image without added Salt & Pepper noise

The computational workload and time expenditure in signal processing predominantly revolve around calculating the median within each window in figure 14 (without salt and pepper) and figure 13 (with salt and pepper). For extensive signals like images, the efficiency of this median calculation significantly influences algorithmic speed. While a naive implementation involves sorting the entire window, the use of the selection algorithm proves more efficient as it targets only the middle value. Additionally, certain signals, especially those represented with whole numbers like images, benefit from histogram medians. Updating the histogram across windows and extracting the median from it proves notably streamlined in comparison to traditional methods.

First and Second Order Partial Derivative Filtering

In contemporary image processing, pixel averaging across a region often results in detail blurring. Leveraging modern tools, the concept of differentiation, akin to its role in integration, can be harnessed to counteract this effect and enhance image sharpness. The application of derivative filters (in figure 15) to digital images enables the utilization of brightness change rates, offering opportunities for contrast enhancement, edge detection, and boundary delineation [11]. Out comes are shown in figure 16 (Original image), 17 (1st order derivative), 18 (2nd order derivative) and 19 (Log based image).

-1	-2	-1
0	0	0
1	2	1

-1	0	1
-2	0	2
-1	0	1

a)

1	0
0	-1

0	1
-1	0

b)

-1	-1	-1
0	0	0
1	1	1

-1	0	1
-1	0	1
-1	0	1

c)

Figure 15: a) Sobel operator b) Roberts operators c) Prewitt operator

Figure 16:Original cameraman image

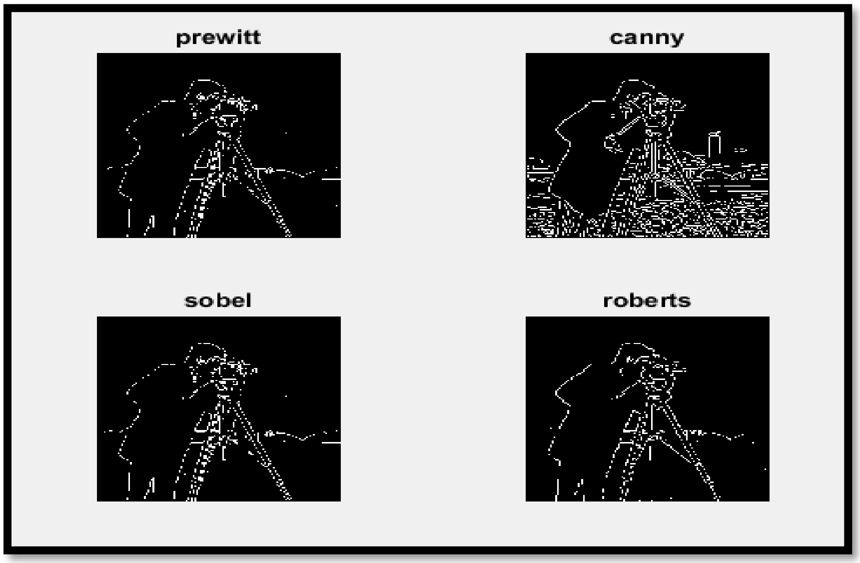


Figure 17: 1st order derivative

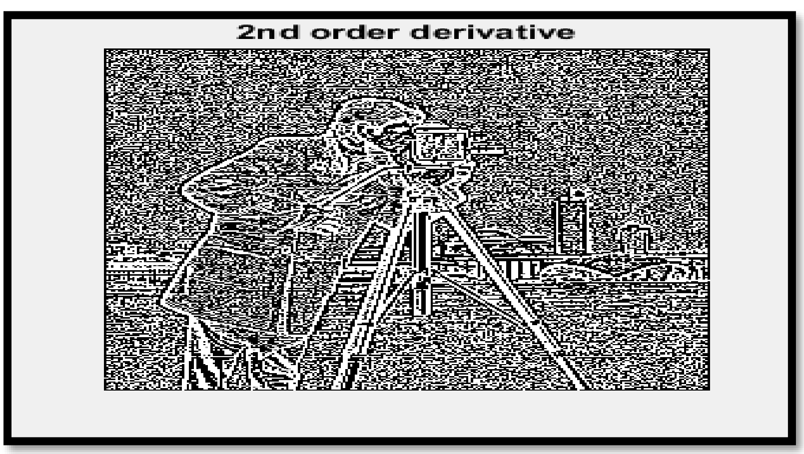


Figure 18:2ndorder derivative

Figure 19: Log based image

Edge Detection

Modern tools for image processing employ a range of advanced mathematical techniques for edge detection, designed to identify points in a digital image where brightness undergoes significant changes or, formally, exhibits discontinuities. These points, often organized into curved line segments known as edges, play a crucial role in various applications. The challenge of detecting discontinuities in one-dimensional signals is analogous to step detection, while observing signal changes over time is termed change detection [11].

In contemporary fields like image processing, machine vision, and computer vision, edge detection serves as a foundational tool, especially in feature detection and extraction.

Common algorithms, such as Sobel, Canny, Prewitt, and Roberts methods shown in figure 20, 21 and 22 and operators are shown in figure 23, contribute to the effectiveness of modern edge detection for tasks like image segmentation, data extraction, and boundary identification.

Figure 20: Image segmentation using the Sobel method

Figure 21: Image segmentation using the Canny method

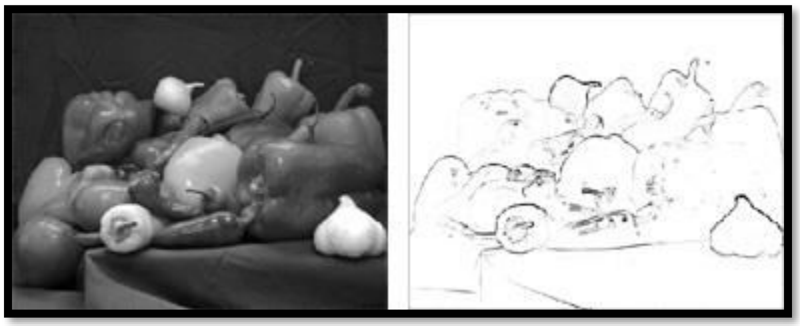


Figure 22: Image segmentation using a Fuzzy Logic method

-1	-2	-1
0	0	0
1	2	1

-1	0	1
-2	0	2
-1	0	1

a)

1	0
0	-1

0	1
-1	0

b)

-1	-1	-1
0	0	0
1	1	1

-1	0	1
-1	0	1
-1	0	1

c)

Figure 23: a) Sobel operator b) Roberts operators c) Prewitt operator

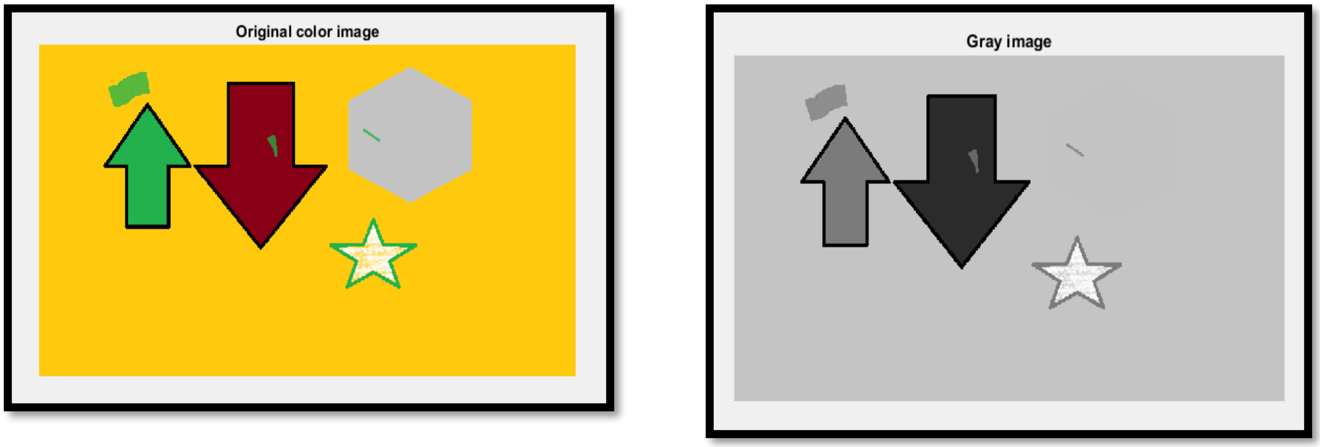


Figure 24: Original colorand Gray image

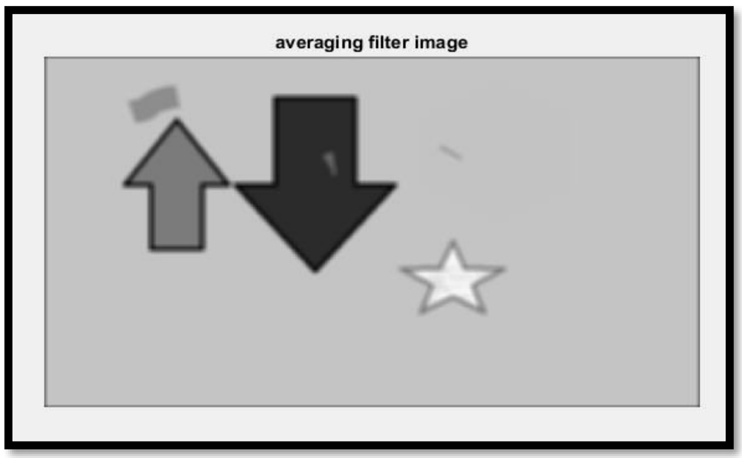
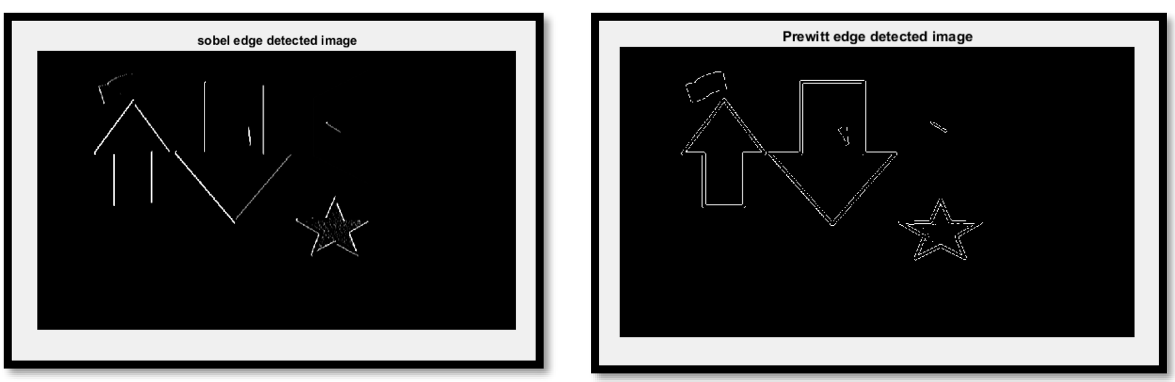


Figure 25: Average filter image



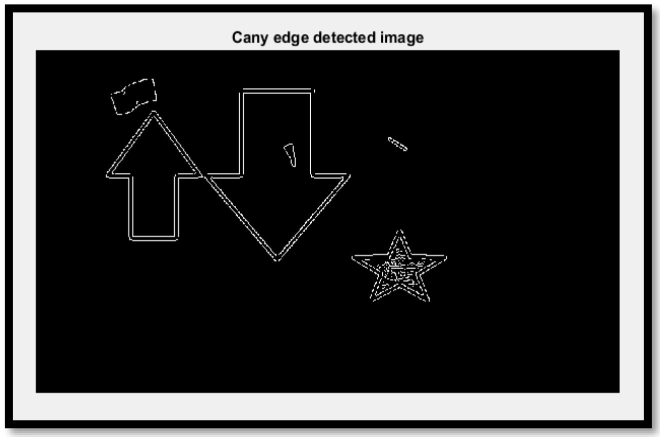


Figure 26: Sobel, Prewitt and Cany edge detected image

In contemporary image processing, the Canny edge detector operator stands out as a superior tool for edge detection compared to alternatives like the Sobel and Prewitt edge detector operators. Known for its advanced algorithmic approach, the Canny operator excels in accurately detecting and highlighting edges in images shown in figure 26. This modern tool surpasses its counterparts by providing more refined and effective edge detection capabilities, making it a preferred choice in various applications within the field of computer vision and image processing shown in figure 24 and 25.

Gaussian Noise

In contemporary data analysis and signal processing, Gaussian noise is a prevalent form of statistical noise characterized by a probability density functionidentical to the normal distribution [12]. This distribution, commonly referred to as the Gaussian distribution, imparts Gaussian noise with specific characteristics shown in figure 27 and surface in figure 28. In practical terms, this means that the values assumed by the noise follow a Gaussian distribution, making it a key consideration in modern tools and methodologies for tasks such as data analysis, signal processing, and various applications in machine learning and statistics.

The probability density function of a Gaussian random variable is given by:

$$p_G(z) = \frac{1}{\sigma\sqrt{2\pi}} e^{-\frac{(z-\mu)^2}{2\sigma^2}}$$

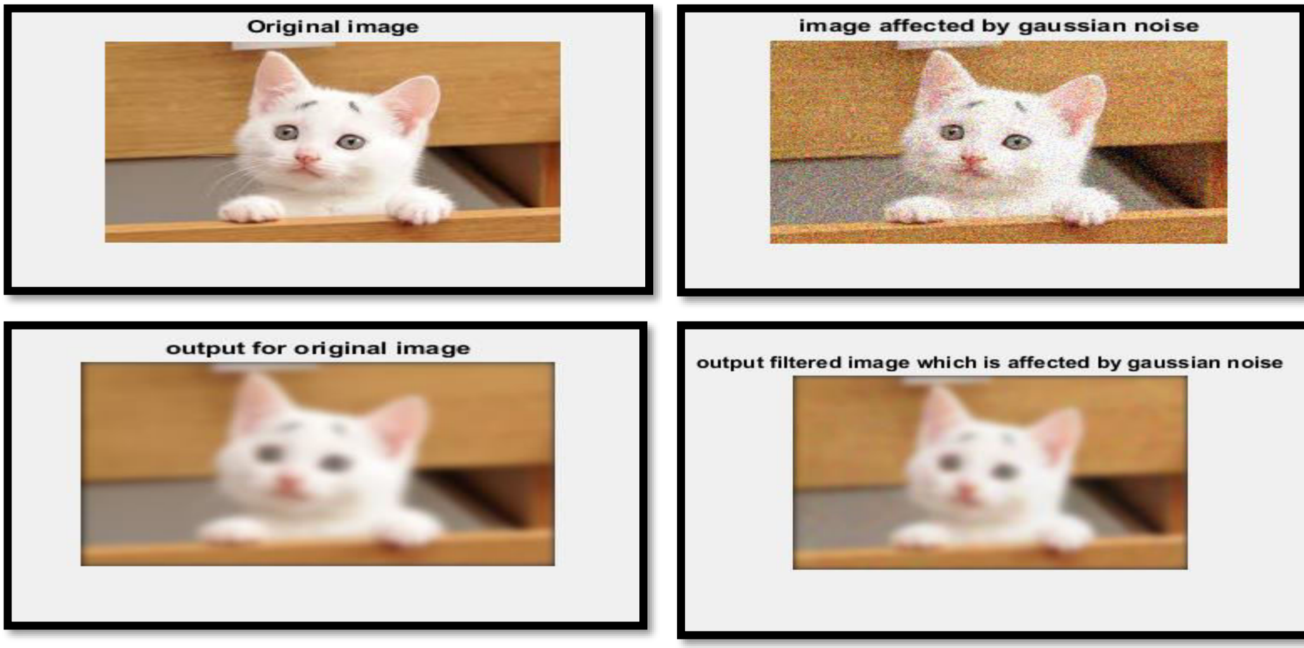


Figure 27: a) Original image b) Image affected by Gaussian noise c) output for original image d) output filtered image which is affected by Gaussian noise

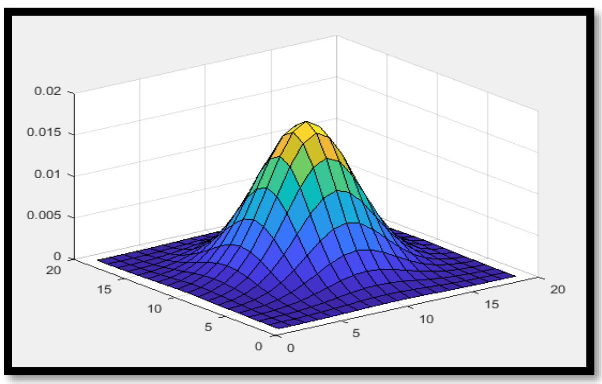
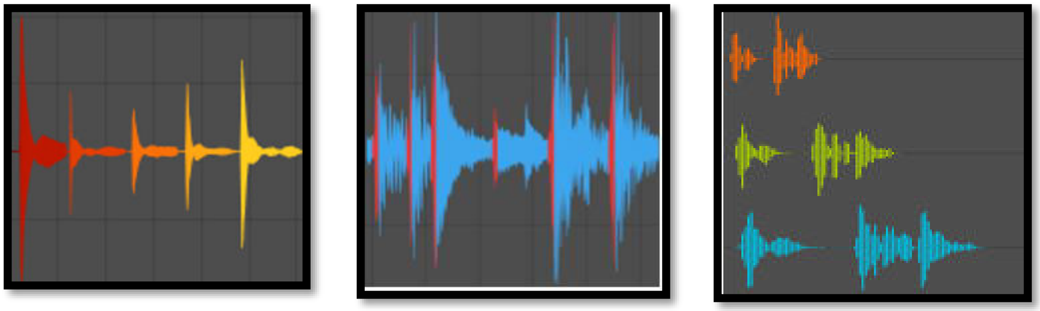


Figure 28: Surface Design

In contemporary image processing using modern tools, the phenomenon of blurring is attributed to the Gaussian filter acting as a low-pass filter. Low-pass filters, prevalent in the Fourier domain, selectively remove high-frequency components while allowing low frequencies to pass through. Essentially, this implies that the filter emphasizes the retention of lower-frequency information, resulting in the overall effect of image blurring. Understanding the role of low-pass filtering, particularly through Gaussian filters, is crucial in various applications where image clarity and detail preservation are key considerations.

Audio Filtering

An audio filter is a frequency dependent amplifier circuit, working in the audio frequency range, 0 Hz to beyond 20 kHz. Audio filters can amplify (boost), pass or attenuate (cut) some frequency ranges [13]. Many types of filters exist for different audio applications including hi-fi stereo systems, musical synthesizers, sound effects, sound reinforcement systems, instrument amplifiers and virtual reality systems.Out comes are shown in figure 29-34.



Audio file-1:

audio file.wav

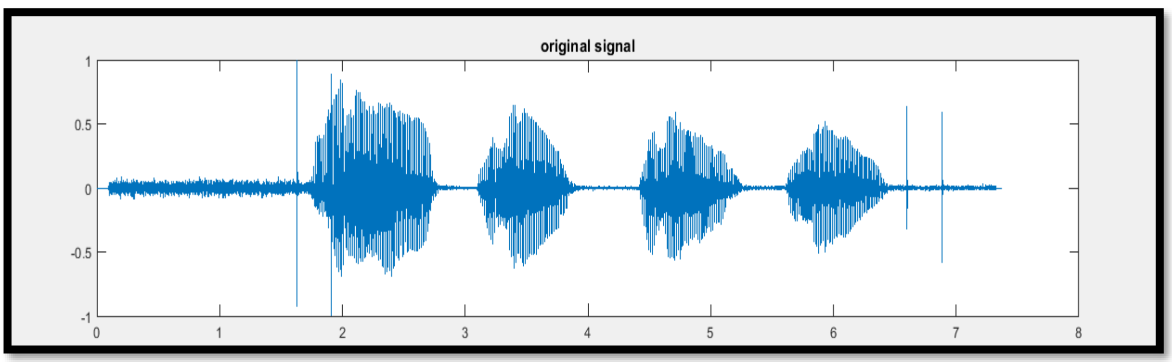


Figure 29: Original signal

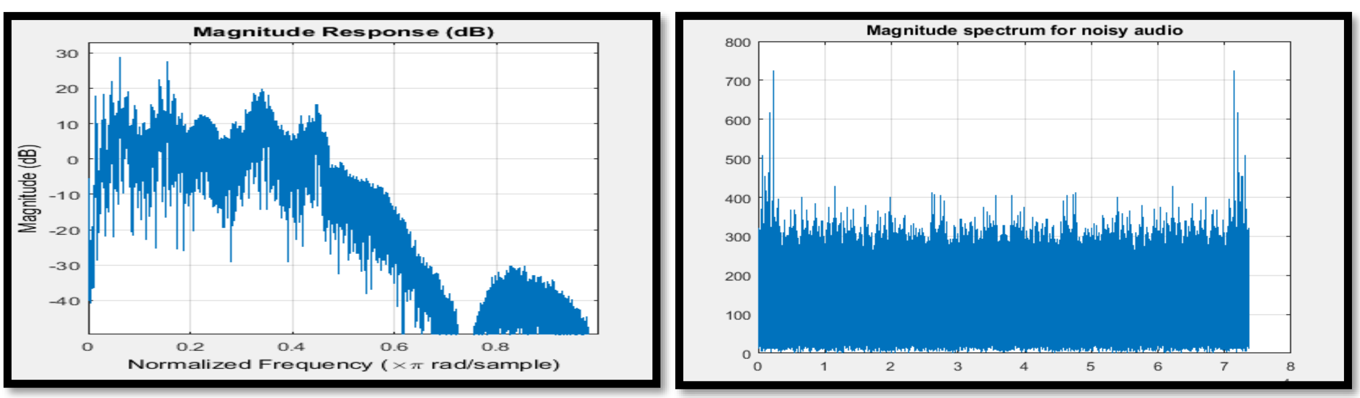


Figure 30: Magnitude response and Magnitude spectrum

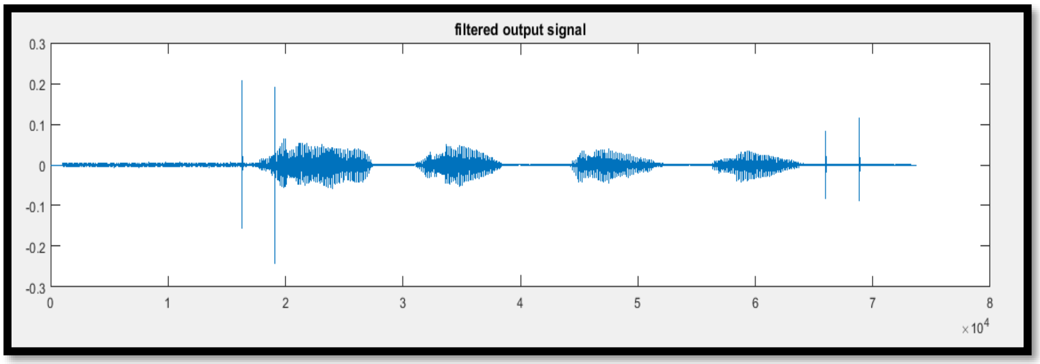


Figure 31: Filtered output signal

Audio file-2:

rana.wav

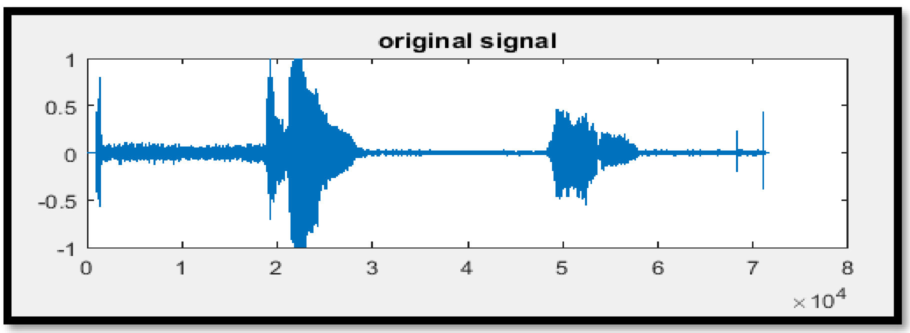


Figure 32: Original signal

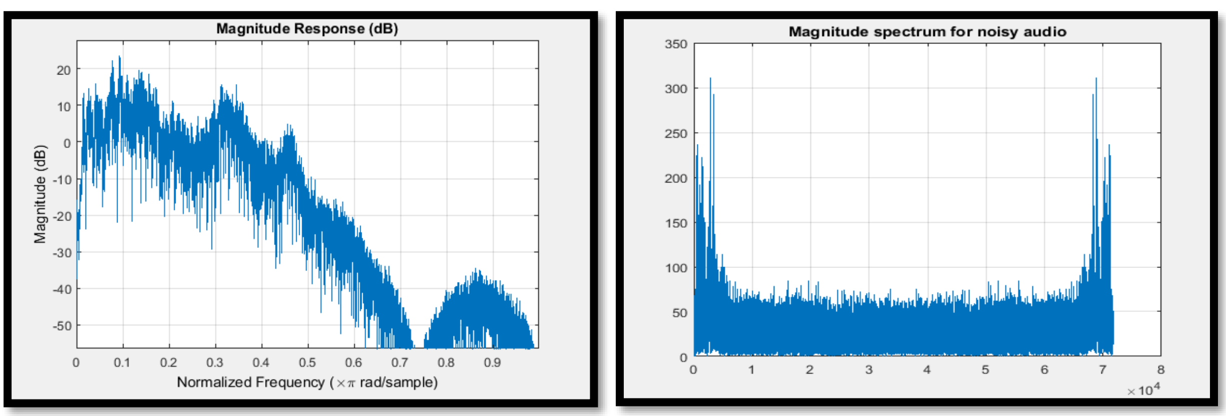


Figure 33: Magnitude response and Magnitude spectrum

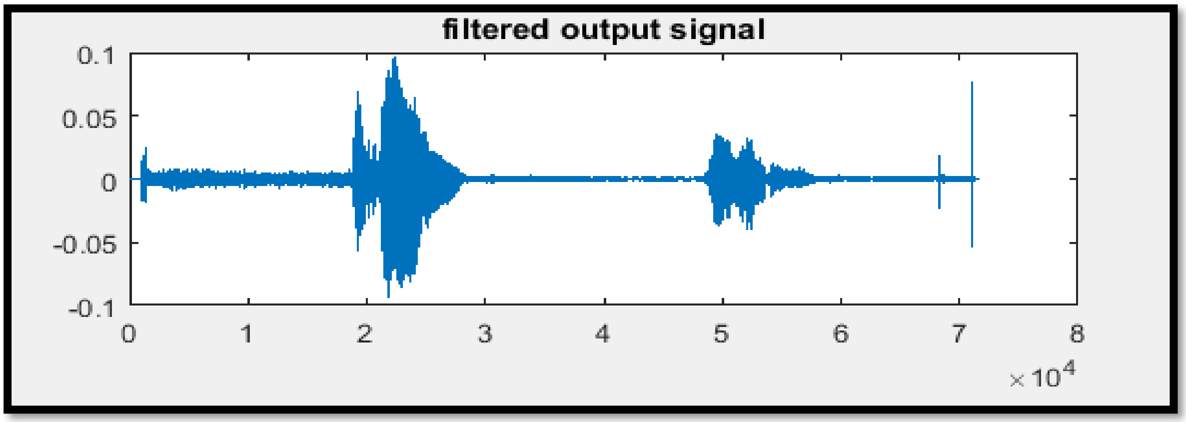


Figure 34: Filtered output signal

In the realm of modern audio processing tools, the output signal from Audio file-2 exhibits a smoother quality compared to Audio file-1. This distinction arises from strategic choices in processing parameters: while Audio file-1 incorporates a noise percentage of .50, Audio file-2 adopts a more refined approach with a lower noise percentage of .10. Additionally, the use of a low-pass filter in Audio file-2 further contributes to the enhanced smoothness of the output signal. These nuanced adjustments in noise management and filtering underscore the precision and effectiveness of contemporary audio processing techniques.

Voice Filtering

This paper discusses human voice filtering. The filtering was done by attenuating human voice's frequencies within a song's frequencies using filter design. Low pass and high pass filter design is one of filter design that is used in this research. The method for the human voice filtering is offered by using a low pass filter design simulated in MATLAB. The specified low pass is based on the human voice's frequencies [14]. The algorithm was done to filter the human voice from a voice recording. The result is then optimized to find the optimal result for the filtering. The methods for the optimization are used to simulate the band-stop filter design with different stop-band, FFT the result to show its frequency spectrum, and repeat the process until the heard voice has the optimal result. Based on the methods, a comparison between various stop-band specifications is presented to find the most optimal one. The chosen low pass is used for male voice filtering and female voice filtering to see if it is suitable for both kind of voices. Out comes are shown in figure 35-38.

Voice Recording-1:

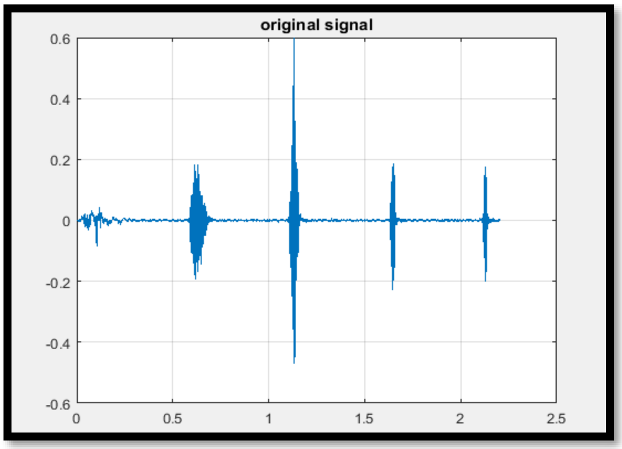


Figure 35: Original Signal

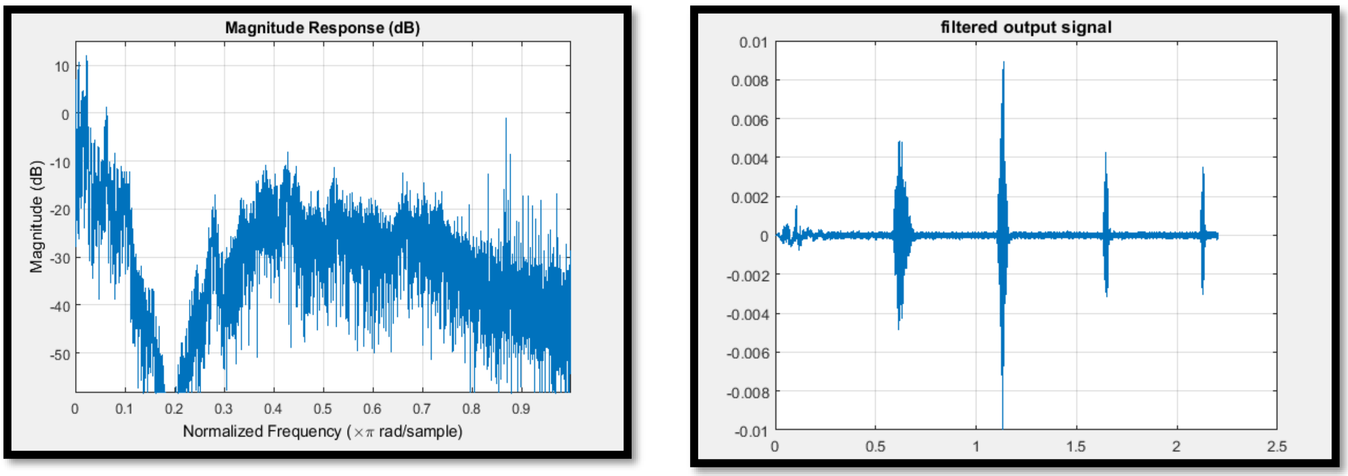


Figure 36: Magnitude response and filtered output signal

Voice Recording-2:

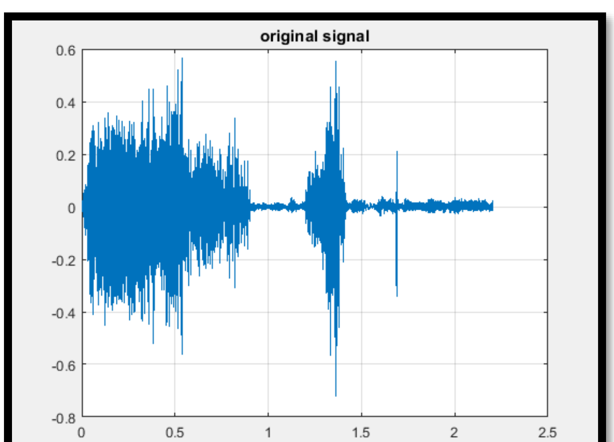


Figure 37: Original Signal

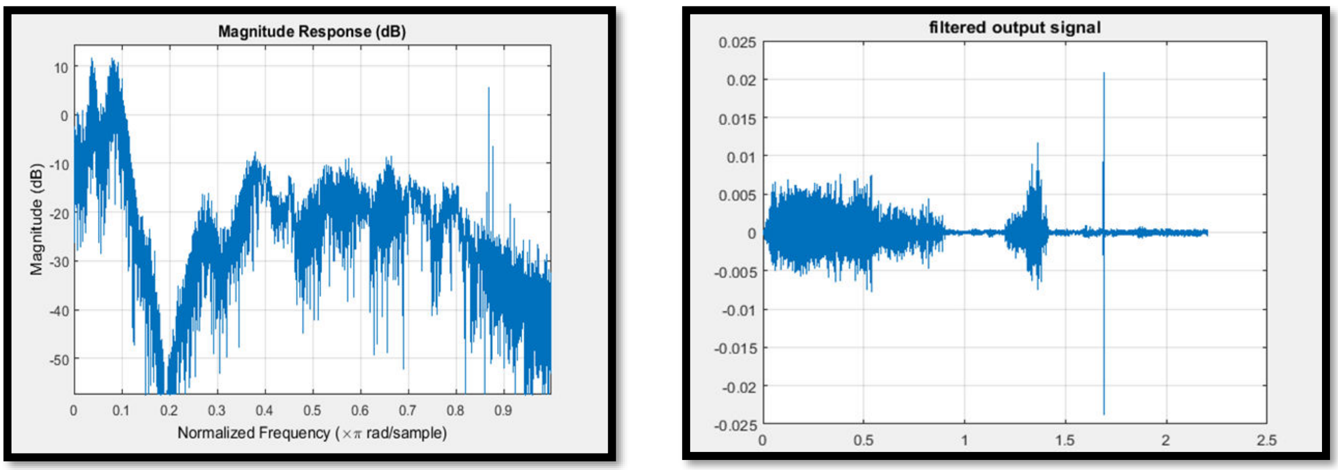


Figure 38: Magnitude response and filtered output signal

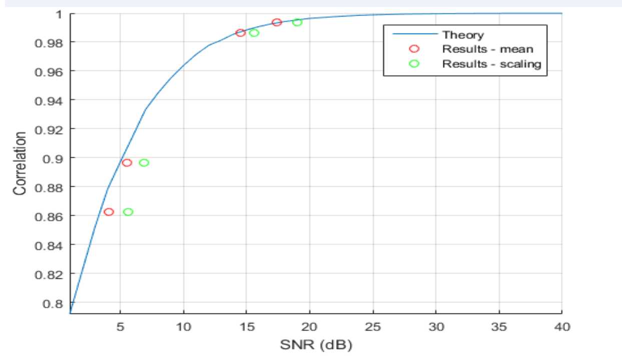
In the contemporary landscape of audio processing tools, addressing noise in a single-microphone recording often involves techniques such as spectral subtraction, particularly effective for mitigating constant noise sources like fan or engine noise. Alternatively, advanced methods leverage statistical approaches and perceptual models of speech. In scenarios with multiple microphones, blind source separation techniques come into play, aiming to isolate speech signals from various sources. It's important to note that achieving perfection remains a challenge, with current outcomes representing a delicate balance between noise suppression and maintaining clarity in the desired speech signal. Striking this balance involves navigating the trade-off between heightened noise reduction and potential degradation of the target signal.

Signal-To-Noise Ratio

SNR (Signal-to-Noise Ratio) in modern tools assesses the ratio of desired signal level to background noise, expressed in decibels. A ratio above 1:1 (greater than 0 dB) signifies more signal than noise [15].

Applied across various domains like electrical signals, isotope levels, biochemical signaling, and financial trading, SNR metaphorically extends to information quality in conversations.

In communication channels, SNR, bandwidth, and channel capacity are interlinked by the Shannon–Hartley theorem, defining SNR as the ratio of signal power to background noise power.



When, corr=1, snr=30

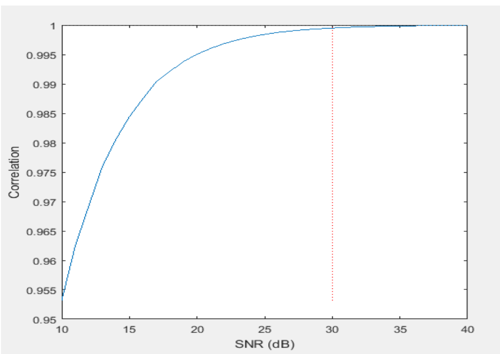


Figure 39: Correlation measurement

When, corr=0.9988, snr=20.44

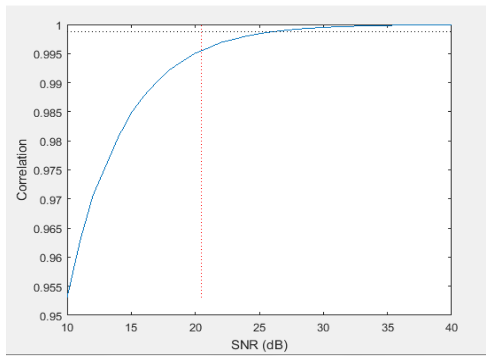


Figure 40:Correlation measurement

When, corr=0.7080, snr=24.44

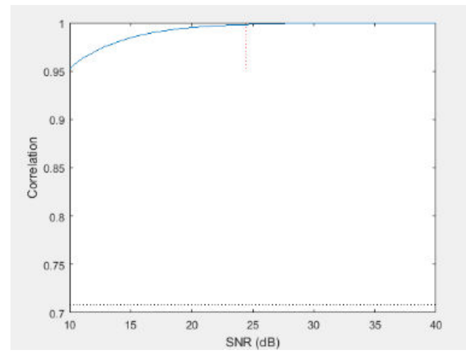


Figure 41: Correlation measurement

In the context of modern toolsfigure 39, the 1 correlation measurement is denoted by the dotted black horizontal line, while the 30 dB SNR result is marked by the dotted red vertical line.Infigure 40, the dotted black horizontal line signifies a correlation measurement of 0.9958, while the dotted red vertical line denotes a 20.44 dB SNR result.Infigure 41, the dotted black horizontal line indicates a correlation measurement of 0.7080, while the dotted red vertical line represents a 24.44 dB SNR result.

The obtained Pearson's correlation coefficients (1.0, 0.9988, and 0.7080) suggest a strong linear relationship between the original signal and the oscilloscope recording. However, the calculated SNR values (30 dB, 20.44 dB, and 24.44 dB) appear lower than expected. This discrepancy could be attributed to additional noise introduced during the digital-to-analog conversion. Further investigation into the signal processing steps and potential sources of noise may be needed to reconcile the observed correlation and SNR values.

Conclusion

In conclusion, filter design is a crucial aspect of signal and image processing, and the choice of filters depends on the specific characteristics and requirements of the data being processed. The effectiveness of filtering methods should be evaluated based on the trade-off between noise reduction and preservation of important information in the signal, voice,audio or image.

Availability of Data and Materials

Accessing data and materials for a project on "Optimizing Signal and Image Processing: A Comprehensive Approach to Filter Design for Quality Enhancement" can be crucial for its success. The availability of data and materials can vary depending on the specific requirements of our project. ImageNet, explore academic journals and research papers in the field of signal and image processing, some commercial databases offer high-quality datasets for a fee etc.

Reference:

1. Digital Signal Processing Committee of the IEEE Acoustics, Speech, and Signal Processing Society, eds. Programs for Digital Signal Processing. New York: IEEE Press, 1979.
2. R. Matei, "Closed-Form Design of 2D Filters with Elliptical and Circular Frequency Response," 2020 European Conference on Circuit Theory and Design (ECCTD), Sofia, Bulgaria, 2020, pp. 1-4.
3. L. Van, I. Khoo, P. Chen and H. H. C. Reddy, "Symmetry Incorporated Cost-Effective Architectures for Two-Dimensional Digital Filters," in IEEE Circuits and Systems Magazine, vol. 19, no. 1, pp. 33-54, Firstquarter 2019.
4. R. Matei, "Design Procedure for 2D Gaussian Anisotropic Filters," 2020 IEEE 63rd International Midwest Symposium on Circuits and Systems (MWSCAS), Springfield, MA, USA, 2020, pp. 174-177.
5. HamidrezaMafi, M. Yargholi, M. Yavari and S. Mirabbasi, "Digital Calibration of Elements Mismatch in Multirate Predictive SAR ADCs," in IEEE Transactions on Circuits and Systems I: Regular Papers, vol. 66, no. 12, pp. 4571-4581, Dec. 2019.
6. TalitaT. A. Borges, L. B. Soares and C. Meinhardt, "A Fine-grained Methodology for Accuracy-configurable and Energy-efficient Gaussian Filters Design," 2020 33rd Symposium on Integrated Circuits and Systems Design (SBCCI), Campinas, Brazil, 2020, pp. 1-6.
7. Peter Apostolov, A. Stefanov and I. Nedialkov, "Efficient Two-Dimensional Filter Synthesis," 26th Telecommunications Forum (TELFOR), Belgrade, pp. 1-4, 2018.
8. A. Croisier, D. J. Esteban, M. E. Levilion, and V. Rizo, “Digital filter forPCM encoded signals,” U.S. Patent 3 777 130, Dec. 4, 1973.
9. K. K. Parhi, VLSI Digital Signal Processing Systems: Design and Implementation. New York: Wiley, 1999.