Noise Reduction in Signal Processing

Project Report for the course

BECE301L DIGITAL SIGNAL PROCESSING

By

MRINANK GAUR (22BEC1258)

RALF PAUL VICTOR (22BEC1222)

VINAY (22BEC1247)

Submitted to

Dr. R. RAMESH

SCHOOL OF ELECTRONICS ENGINEERING

VELLORE INSTITUTE OF TECHNOLOGY

CHENNAI - 600127

April 2024

Certificate

This is to certify that the Project work titled "**Noise Reduction in Signal Processing**" is being submitted by Mrinank Gaur 22BEC1258, Ralf Paul Victor 22BEC1222 and Vinay 22BEC1247 for the course B**ECE301L Digital Signal Processing** is a record of Bonafide work done under my guidance. The contents of this project work, in full or in parts, have neither been taken from any other source nor have been submitted to any other Institute or University.

Dr. R. RAMESH
Guide

ABSTRACT

Signal processing is an especially important part of modern technology, with applications ranging from telecommunications to biomedical engineering. But as we talk about real world signal, they are corrupted by noise, which can drastically degrade the performance of signal processing systems. In this paper we do an overview of noise reduction techniques and methods in signal processing and use MATLAB for this. We discussed various methods including like (filtering, Fourier Transform, Least Mean Square Algorithm, and Gaussian Filter Method.). Each of these techniques are evaluated based on its effectiveness in reducing noise from the original sound and preserving the integrity of the original signal. The paper concludes with a discussion on noise reduction and signal distortion, highlighting the need for adaptive and robust noise reduction techniques that can handle a wide range of noise types and signal conditions.

ACKNOWLEDGEMENT

We wish to express our sincere thanks and deep sense of gratitude to our project guide, **Dr. R. Ramesh,** School of Electronics Engineering, for his consistent encouragement and valuable guidance offered to us in a pleasant manner throughout the course of the project work.

We are extremely grateful to **Dr. Susan Elias,** Dean of School of Electronics Engineering, VIT Chennai, for extending the facilities of the School towards our project and for his unstinting support.

We express our thanks to our Head of the Department **Dr. K. Mohanaprasad** for his support throughout the course of this project.

We also take this opportunity to thank all the faculty of the School for their support and their wisdom imparted to us throughout the course.

We thank our parents, family, and friends for bearing with us throughout the course of our project and for the opportunity they provided us in undergoing this course in such a prestigious institution.

Mrinank Gaur | Ralf Paul Victor | Vinay

Mrinank Gaur | **Ralf Paul Victor** | **Vinay**

Table of contents

List of figures

1. Introduction

In signal processing, one of the most challenging this in the signal is noise. Noise, unwanted or random additions to the signal, can significantly degrade the quality and reliability of the signal, thereby affecting the performance of the entire system. That is why the implementation of noise reduction techniques is important in signal processing, which aims to minimize the effect of noise in the signal coming while preserving the integrity of the original signal. Noise can originate from various sources such as electronic devices, transmission interference, environmental factors, and more. The characteristics of noise may vary, ranging from small noise to large noise that interferes in the signal. The field of noise reduction surrounds a wide array of techniques and methods. From simple filtering methods that remove certain frequency components to more complex algorithms that adaptively adjust based on the properties of the signal and noise, these techniques form an integral part of many modern signal processing systems. This paper aims to provide a comprehensive overview of noise reduction techniques in signal processing using the Matlab code. We will investigate various methods including (filtering, Fourier Transform, Least Mean Square Algorithm, and Gaussian Filter Method.) Each technique will be discussed in detail with its advantages and limitations. The goal is not only to present these techniques but also to provide a comparative analysis that can guide researchers and engineers in choosing the most suitable method for their specific application., we hope to shed light on how noise reduction plays a vital role in enhancing signal integrity in today's digital world."

1.1 Filtering

Filtering is a class of signal processing. The defining feature of filters is the complete or partial suppression of some aspect of the signal. Most often, this means removing some frequency or frequency bands There are many different bases of classifying filters and these overlap in many ways:

1) Linear and non- linear
2) Time -variant and time-invariant
3) Causal and non-causal
4) Discrete and non-desecrate

1.2 Fourier Transform

The Fourier Transform is a powerful mathematical tool that transforms a signal from the time domain to the frequency domain, and vice versa.

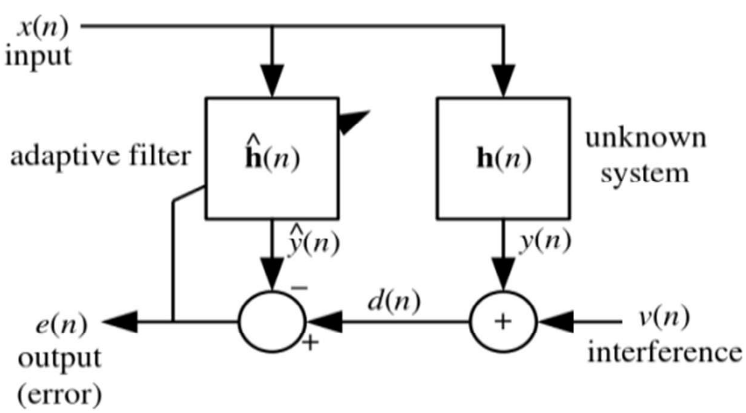
The Fourier Transform of a continuous-time function x(t):

$$X(\omega) = \int_{-\infty}^{+\infty} x(\mathrm{t}).\, e^{-i\omega t} dt$$

This equation transforms the signal from the time domain (x(t)) to the frequency domain (X(ω)). The variable ω represent angular frequency.

1.3 Least Mean Square (LMS) Algorithm

Least mean squares (LMS) algorithms are a class of adaptive filter used to mimic a desired filter by finding the filter coefficients that relate to producing the least mean square of the error signal (difference between the desired and the actual signal)



$$\mathbf{x}(n) = [x(n), x(n-1), \ldots, x(n-p+1)]^T$$

$$\mathbf{h}(n) = [h_0(n), h_1(n), \ldots, h_{p-1}(n)]^T, \quad \mathbf{h}(n) \in \mathbb{C}^p$$

$$y(n) = \mathbf{h}^H(n) \cdot \mathbf{x}(n)$$

$$d(n) = y(n) + \nu(n)$$

$\hat{\mathbf{h}}(n)$ estimated filter; interpret as the estimation of the filter coefficients after n samples

$$e(n) = d(n) - \hat{y}(n) = d(n) - \hat{\mathbf{h}}^H(n) \cdot \mathbf{x}(n)$$

Fig1.3.1: Least Mean Square Algorithm Block Diagram

1.3.1 Signal Model

• The observed signal is given by: x[n]=d[n]+v[n]

• x[n] is the noisy observed signal.

• d[n] is the clean signal you want to estimate.

• v[n] is the noise or interference.

1.3.2 LMS Estimation

• You will estimate the clean signal d[n] using the LMS algorithm.

• The estimated signal, denoted as d^[n], is updated iteratively.

1.3.3 Error Signal

• The error signal e[n] is defined as the difference between the estimated signal and the observed signal: e[n]=x[n]−d^[n];

1.4 Gaussian Filter Method

A Gaussian filter will have the best combination of suppression of high frequencies while also minimizing spatial spread, being the critical point of the uncertainty, these properties are important in areas such as digital telecommunication systems The filters are then categorized into several groups based on their applications for state estimation. These groups involve linear optimal filtering, nonlinear filtering, adaptive filtering, and robust filtering.

2. Methodology

In this report the researcher using some method to reduce the noise in the signals that are Couse due some external interference and some of the due to the internal interference in the receiving signals. In this paper the researcher is using Matlab calculate the total noise in the signals and reduce the noise from the signals using methods and technique that are as following (filtering, Fourier Transform, Least Mean Square Algorithm, and Gaussian Filter Method.) Using Matlab, the researcher can demonstrate the reduction of the noise from the signals and evaluate the output signals after reducing the noise from the signals. Least Mean Square (LMS) Algorithm: The LMS algorithm is an adaptive filter that adjusts its filter coefficients to minimize the mean square error.

2.1 Signal-to-Noise Ratio (SNR)

$$SNR_{dB} = 10.\,log_{10}(\frac{P_{signal}}{P_{noise}})$$

• SNR (dB) is the Signal-to-Noise Ratio in decibels.

• P_{signal} is the power of the signal.

• P_{noise} is the power of the noise.

$$P_{signal} = \frac{1}{N}\sum_{i=1}^{N} x(i)^2$$

Where x(i) represents the individual samples of signal, and N is the total number of samples.

$$P_{noise} = \frac{1}{N}\sum_{i=1}^{N} n(i)^2$$

Where n(i) represents the individual samples of signal, and N is the total number of samples.

2.2 LMS Algorithm

The Least Mean Square (LMS) algorithm is a widely used method in signal processing and adaptive filtering for estimating a desired signal or minimizing the error between the estimated signal and the actual signal.

2.3 MATLAB Code

```
% Load the audio file

[y, Fs] = audioread("dammm.wav");



% Generate noisy signal by adding noise

SNR = 10; % Signal-to-Noise Ratio (in dB)

noisy_signal = awgn(y, SNR);



% Create an adaptive filter

filter_length = 256;

mu = 0.01; % LMS step size

lms_filter = dsp.LMSFilter(filter_length, 'StepSize', mu);



% Initialize variables

desired_signal = y;

error_signal = zeros(size(y));

output_signal = zeros(size(y));
```

```
% Perform adaptive noise cancellation
for i = 1:length(y)
    [output, error] = step(lms_filter, noisy_signal(i), desired_signal(i));
    output_signal(i) = output;
    error_signal(i) = error;
end

% Play the original, noisy, and denoised audio signals
sound(y, Fs); % Original signal
pause(length(y)/Fs + 1); % Pause for original signal playback
sound(noisy_signal, Fs);
pause(length(noisy_signal)/Fs + 1); % Pause for noisy signal playback
sound(output_signal, Fs);

% Create the time vector
t = (0:length(y)-1)/Fs;

% Plot the results
figure;
plot(t, y, 'b');
title('Original Signal');
```

```
xlabel('Time');
ylabel('Amplitude');
legend('Original Signal');

figure;
plot(t, noisy_signal, 'g');
title('Noisy Signal');
xlabel('Time');
ylabel('Amplitude');
legend('Noisy Signal');

figure;
plot(t, output_signal, 'r');
title('Corrected Signal');
xlabel('Time');
ylabel('Amplitude');
legend('Corrected Signal');
```

3. Results and Discussion

3.1 MATLAB Output

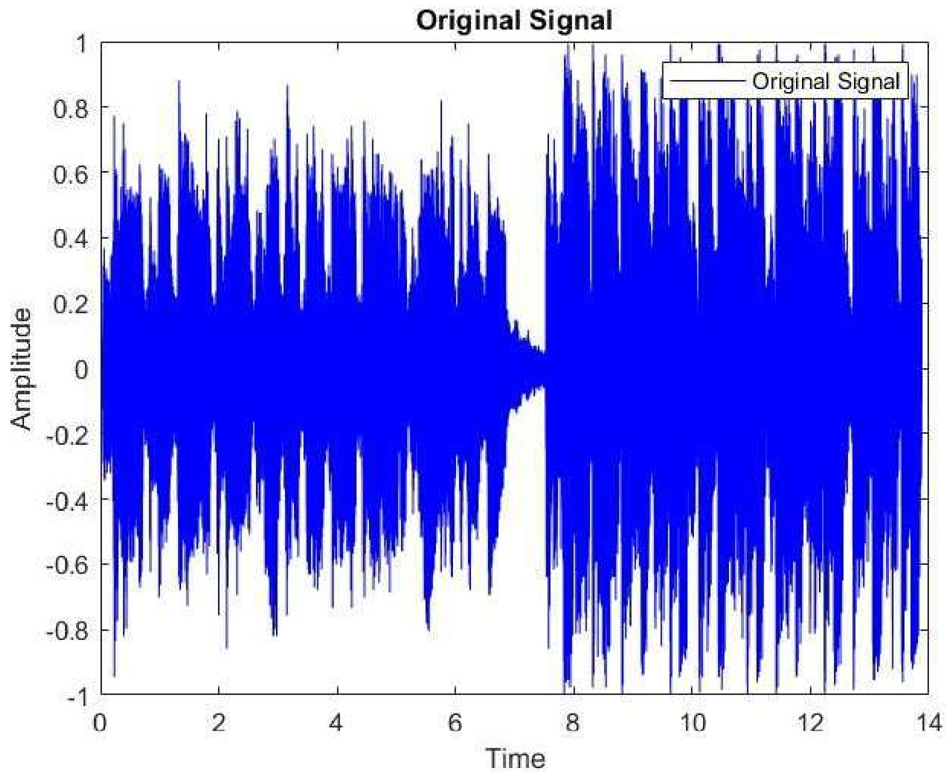


Fig3.1: Original Signal

Analysis

Here in Fig3.1 we can see that the signal is noticeably clear and there is no distortion in the signal and there is no noise in the signal, but it is an ideal case when the signal is clear, in the real case it is not possible.

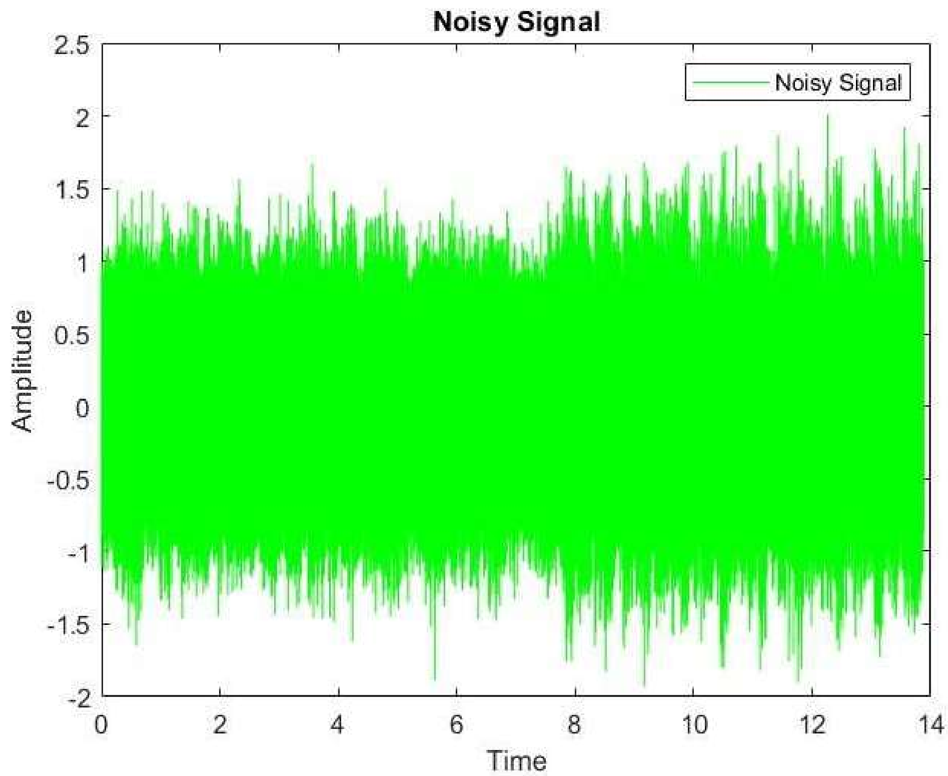


Fig3.1: Noisy Signal

Analysis

Here in Fig3.2 to demonstrate the noise in the signal we add some noise in the signal by using some noise to signal ratio and add to the clear signal to get some noise/distortion into the signal. Here you can see that the signal is more disturbed, and it is not as smooth as it was in the clear signal. And it causes technical problems or analytical issues when we transfer this signal to the system. That why it is important to remove the noise from the signal

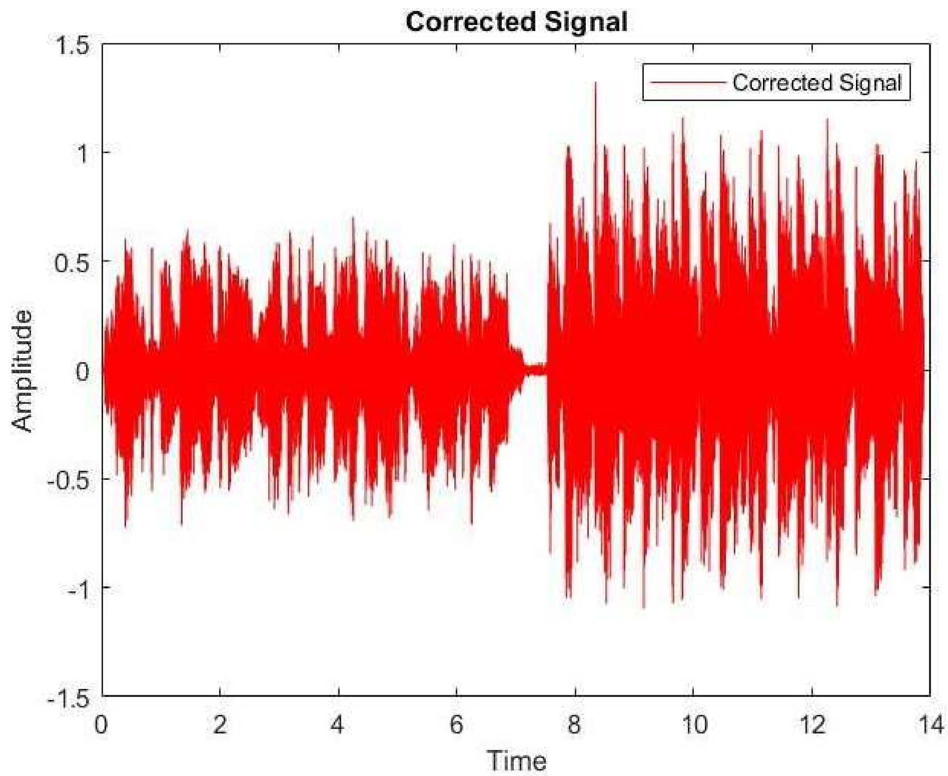


Fig3.1: Corrected Signal

Analysis

Here in Fig 3.3, you can see the signal is now clearer than it was in the above signal. After doing the Least Mean Square (LMS) Algorithm, to the signal the noise gets cancelled from the signal, and it became clearer, and we can use this to system. We can compare the signal of original and cleared signal after correction Numerical Analysis: In the provided MATLAB code, a sinusoidal signal contaminated with Gaussian noise is generated. The noisy signal is then processed using a moving average filter with a specified window size. The original signal, noisy signal, and the corrected signal after applying the moving average filter are plotted for comparison. The window size determines the extent of smoothing applied to the noisy signal. 13 Conclusion: In conclusion, from the ab

3.2 Conclusion

In conclusion, from the above result we can see that it is especially important to reduce the noise from the signal so that a system can work more efficiently. By using the Matlab and LMS method we reduce the noise from the signal and make it clear. The noise reduction in signals is a critical aspect in many fields such as audio processing, image processing, and seismic exploration. Various methodologies like Discrete Fourier Transformation (DFT) Technique, Gaussian Filter Method, Least Mean Square (LMS) Algorithm, Wavelet Decomposition and Reconstruction Method, Nonlinear Wavelet Transform Threshold Method, are used to achieve this. Each method has its own strengths and weaknesses and is suitable for different scenarios. The choice of method depends on the specific characteristics of the signal and the noise. Despite the challenges, effective noise reduction can significantly improve the quality of the signal and the reliability of the data derived from it. It is an active area of research with new techniques and improvements being constantly developed.

References

[1]Vedansh Thakkar. “Noise cancellation using least mean square algorithm” IOSR journal of Electronic and Communication Engineering (IOSR-JECE), vol.12, no.5,2017, pp,64-67

[2] “Simulation for noise cancellation using LMS adaptive filter”: Jia-Haw Lee et al 2017 IOP Conf. Ser.: Mater. Sci. Eng. 211 012003

[3] SIGNAL NOISE REDUCTION AND FILTERING: Tatiana Kelemenová; Ondrej Benedik; Ivana Koláriková ,(PDF) SIGNAL NOISE REDUCTION AND FILTERING (researchgate.net)

[4]” A Review on Noise Cancellation by LMS Adaptive Filters” G. Sunil Kumar , Dr.A.A.Ansari ,Dr.S.M.Ramesh, Vol 12, Issue 11, NOV /2021