DIGITAL FILTERING OPTIMIZATION ON NOISY HUMAN SPEECH IN ASSISTIVE LISTENING SYSTEM

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ABSTRAK

Kertas ini mewakili sistem pembatalan hingar akustik oleh algoritma penapis suai. Algoritma penyesuaian ialah kuasa dua terkecil min (LMS), kuasa dua terkecil min ternormal (NLMS) dan kuasa dua terkecil rekursif (RLS). Kertas kerja ini menyiasat pelaksanaan pengiraan LMS, NLMS dan RLS untuk hingar akustik dengan menjalankan model secara berterusan untuk tanda bunyi dan pemprosesan isyarat. Asasnya adalah pada penggunaan pengiraan NLMS dan RLS untuk mengurangkan bunyi atau kekecohan yang tidak diingini seterusnya meningkatkan kualiti isyarat bunyi yang diingini. Kaedah MATLAB Simulink sedang digunakan untuk simulasi. Penapis suai biasanya digunakan untuk membatalkan bahagian bunyi yang dicampur dengan tanda bunyi yang dikehendaki. LMS biasanya digunakan kerana kesederhanaan dan keteguhannya, namun ia mengabaikan untuk menyelesaikan kriteria penggabungan jadi di sini LMS diubah suai oleh NLMS yang merupakan sejenis algoritma LMS dan kami juga mencuba untuk RLS yang menunjukkan peningkatan yang ketara. RLS menunjukkan pameran yang lebih baik dan ia mempunyai kelajuan/kadar pertemuan yang lebih pantas daripada pengiraan LMS dan NLMS dengan kekuatan yang lebih baik untuk persekitaran yang boleh diubah dan keupayaan mengikuti yang lebih baik dan hampir boleh mendapat isyarat pertuturan yang jelas.

ABSTRACT

This paper represents the acoustic noise cancellation system by adaptive filter algorithms. The adaptive algorithms are least mean square (LMS), Normalized least mean square (NLMS) and recursive least square (RLS). This paper investigates the execution of LMS, NLMS and RLS calculations for acoustic noise by running the model continuously for sound signs and signal processing. The fundamental is on the utilization of NLMS and RLS calculations to lessen undesirable noise or commotion hence increasing desired sound signal quality. MATLAB Simulink method is being utilized for simulation. Adaptive filter is usually utilized for the undoing of the noise part which is blended with the wanted sound sign. LMS is generally utilized because of its effortlessness and robustness, however it neglects to finish merging criteria so here LMS is improvised by NLMS which is a sort of LMS algorithm and we additionally tried for the RLS which shows significant improvement. RLS shows better exhibitions and it has speedier meeting speed/rate than LMS and NLMS calculations with better strength to alterable environment and better following ability and almost can get clear speech signal.

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LIST OF SYMBOLS

dB	Decibel
μ	Step size
λ	Lambda
s	Time(seconds)

LIST OF ABBREVIATIONS

ALD	Assistive Listening Device
ANC	Active Noise Cancellation
LMS	Least Mean Square
NLMS	Normalized Least Mean Square
RLS	Recursive Least Square
RMSE	Root Mean Square Error
SNR	Signal-to-Noise Ratio

CHAPTER 1

INTRODUCTION

1.1 Project Background

Speech is one of the most natural and useful media of human communication. If a computer could properly handle speech signals in our daily lives, it could provide us with more convenient and comfortable speech services. However, when a speech signal is captured by distant microphones, background noise and reverberation contaminate the original signal and severely degrade the performance of existing speech. The presence of environmental noise or background noise poses a challenge in many speech applications and degrades system performance. Environmental noise includes sounds from traffic, industry, construction, babble or any unwanted sounds. Environmental noise degrades speech in an additive manner, resulting in noisy speech signals.

During the transmission or reception of a signal in any communication system, an unwanted signal is injected into the communication, making it unpleasant for the receiver and calling the communication's quality into doubt. Noise is a term used to describe such a disruption. Noise is an unwanted signal that interferes with the original communication signal and corrupts its properties. This change in the communication process causes the message to change. It is most commonly inputted at the channel or receiver.

Noise detection and reduction for speech applications is frequently expressed as a digital filtering issue, with clean speech estimates produced by running the noisy speech through a linear filter. With such a formulation, the key question of noise reduction becomes how to construct an effective filter capable of greatly suppressing noise while causing no discernible speech distortion. The voice signal captured by a microphone is frequently contaminated by noise from a variety of sources. Such contamination can alter the properties of voice signals, reducing speech quality and intelligibility and inflicting severe harm to human-to-machine communication systems.

This project uses human speech that has been corrupted by the high-power noise that will be recovered and enhanced by the proposed system for the better listening by the listening impaired person. Assistive Learning System is very essential and important for every impaired person that has been suffering from various problems such as deafness, blindness, handicapped and so on. With a wide range of technologies that has been very useful for this generation of our society, this assistive learning technologies can improve the lifestyle of an impaired person. This noise cancellation system is about designing pre-processing device by implementing a digital filtering optimization on noisy human speech in assistive learning system. In short words, we have to design a digital filter that has the ability to eliminate the noisy human speech.

1.2 Problem Statement

A problem statement is a brief description of the issue or issues that a project seeks to address. The problem statement specifies the current state, the desired future state, and any gaps in between. A problem statement is an important communication tool that can help ensure that everyone working on a project understands the problem and why the project is important.

As the noise produced by various ambient sources like vehicles normally lies in this frequency range, speech signals get easily distorted by the ambient noises or Additive white Gaussian noise (AWGN). The degraded speech therefore needs to be processed for the enhancement of the speech components. The aim of the speech enhancement is to improve the quality and intelligibility of degraded speech signal.

Noise can disrupt verbal communication and be stressful, distracting, and irritating. Noise makes it difficult to understand what others are saying, including hearing safe work instructions. This exchange includes face-to-face conversations, phone conversations, audible danger/warning signals, and public address system speech. A noisy atmosphere can pose a potential risk that audible alarms cannot be heard, interfering with oral communication and, as a result, interfering with the operation. Noisy human speech can cause distortion or interference between two communicators. In order to cut off high power noise, a digital filtering optimization is required to solve this problem.

1.3 Objectives

The main objective of this project is to develop an active noise cancellation (ANC) system in assistive listening system. In order to accomplish this, there are few sub objectives that must be achieved. The objectives are listed below.

1. To develop a noise cancellation system that can eliminate high-noise up to 80db.

2. To implement a suitable digital filter to optimize the speech intelligibility.

3. To analyse the processed signal using MATLAB/Simulink software.

4. To compare the performance of three types of adaptive algorithm filters.

1.4 Scope

The listening impaired person can hear the other person’s speech/voice better via implementing the noise cancellation system. Next, the hearing impaired people can communicate well with the help of suitable hearing aids device which implemented with digital filtering optimization technique. The assistive listening system will be very useful for the hearing impaired people. From this, the developed digital filtering system will be able to help the daily life activities for the people who has exposure on high-noise environment. This project focuses on developing digital filtering to reduce or eliminate noise embedded on human speech signal. The optimization technique is also considered to be implemented and the effect to the proposed system will be examined.

1.5 Thesis Outline

This project highlights on the research on digital filtering optimization on noisy human speech signal. The thesis outline consists of five chapters.

Chapter 1 covers the Introduction of the research that is digital filtering optimization on noisy human speech in assistive listening system. Therefore, Project Background, Problem Statement, Objective and Project Scope will be discussed.

Chapter 2 covers Literature Review which is about the findings based on previous research done by researchers that include methods, algorithms and devices used to develop the noise cancellation system.

Chapter 3 covers the methodology that includes the method of solution and design development of this project. The system's process and block diagram are discussed in this chapter. The application of three types of adaptive algorithms are discussed as well.

Chapter 4 which is Results and Discussion explains the outcome of the research, data analysis, system testing using Matlab Simulink and the output signal from the collected results.

Chapter 5 discusses the Conclusion for conducting this project. The shortcomings and recommendation for future works will be stated as well as impact on the society.

CHAPTER 2

LITERATURE REVIEW

2.1 Introduction

This chapter includes the review of literature that highly resembles about the researches and studies related to this project. This step is needed to ensure that my project is follows the guidelines of their case studies and researches. Apart from that, it will also able to find theoretical knowledge that will help for this project.

Literature review can define as a review of previous, relevant literature that is necessary and critical component of any academic project. An effective review establishes a solid foundation for furthering our knowledge. It enables to aid in the development of theories, closes gaps in research, and identifies areas in which more research is required. It is also regarded as the fundamental homework for all researchers and analysts that is regardless of any level.

A literature review helps to establish rapport with the audience or readers so that they feel the thesis is supported by facts and continue to read it. It also aids in the prevention of negative plagiarism and the concentration of study

2.2 Examples of articles and journals

The table below shows the related examples of articles and journals that I researched to get more info knowledge regarding my project. This articles and journals helped me throughout this semester to gained some information in terms of methodology and results for my project.

Table 1.1 Related Literature

Author name	Publication Title	Description	Methods used	Findings
Roshahliza M Ramli, Salina Abdul Samad, Ali O. Abid Noor (November 2018)	Objective and Subjective Evaluation of noise cancellation systems with selectable algorithms for speech intelligibility.	Comparison of performance between two systems using subjective measurements for speech intelligibility which is ANC and ALE.	Objective measurements=Mean Square Error (MSE), Output Signal to Noise Ratio (SNR). Subjective measurements=Listening test (MOS) technique mean opinion score, Experimental setup.	The selectable adaptation algorithms for the systems are the NLMS, AP and DSM-AP. The switching between algorithms is triggered by the value of the noise eigenvalue spread.

Amandeep Singh Dhanjal, Williamjeet Singh (Feb 2019)	Tools and Techniques of Assistive Technology for hearing impaired people	Assistive devices categorized as Assistive Listening Devices (ALD), Augmentative and Alternative Communication (AAC) and Alert systems.	five parameters is used for assistive technology for hearing aids: Speech to text, Text to Speech, Display System, Haptic or Visual Feedback.	Hearing aids, tools, sign application is very essential for physically impaired people. This enables them reduce burden on everyday activities.
Nils Westerlund, Mattias Dahl, Ingvar Claesson (June 2005)	Speech enhancement for personal communication using an adaptive gain equalizer	speech enhancement method for personal communication, where the input signal is divided into a number of subbands that are individually and adaptively weighted in time domain.	focused on speech enhancement, acting as a speech booster.	The adaptive gain equalizer (AGE) acts as a speech booster, only active when speech is present and it remains idle when only noise is present.

Sara Akbarzadeh, Sungmin Lee, Fei Chen and Chin-Tuan Tan (July 2019)	The effect of perceived sound quality of speech in noisy speech perception by normal hearing and hearing impaired listeners	comparison of the perceived quality ratings of noisy sentences created at different speech levels, noise levels, and noise types by normal hearing and hearing impaired listeners.	the method used for this paper is stimulus, subjects and procedure	This study investigates the effect of different factors (processing conditions; level of speech and level of noise present in noisy sentences) on the speech quality perception by NH, CI and HA subjects.

2.3 Types of Assistive Devices

Any device that assists a person with hearing loss or a voice, speech, or language disorder in communicating is referred to as an assistive device or assistive technology. These terms frequently refer to devices that assist a person in hearing and understanding what is being said more clearly or in more easily expressing thoughts. With the advancement of digital and wireless technologies, an increasing number of devices are becoming available to assist people with hearing, voice, speech, and language disorders in communicating more meaningfully and fully participating in their daily lives.

2.3.1 Assistive Listening Devices

This device helps to enhance the sounds you wish to hear or listen to, especially in noisy environments. ALDs can be used in conjunction with a hearing aid or cochlear implant to improve the wearer's ability to hear specific sounds.

Figure 2.1 Examples of assistive listening devices

2.3.2 Augmentative and Alternative Communication (AAC) Devices

This device can be able assisting persons with communication problems in expressing themselves These devices can range from a basic image board to a computer application that converts text into speech.

Figure 2.2 Examples of AAC devices

2.3.3 Alerting Devices

This gadget links to a doorbell, telephone, or alarm and generates a loud sound or flashing light to notify someone with hearing loss of an upcoming event.

Figure 2.3 Examples of alerting devices

2.3.4 Advantages & Disadvantages of Assistive Devices

Usually most of the assistive devices have their own advantages and disadvantages. It depends on how the people can be able to implement them in their daily lives. For example, one of the advantages is that students can work at their own pace. Students with disabilities may study technology at their own rate while still learning the same things as regular students. Secondly, people especially students can attain academic standards. Assistive technology has made it simpler for students to achieve academic excellence and standards. Students can hear and relate to others more effectively, which contributes to the students' overall growth and development at school.

The disadvantages are most assistive technology devices are very expensive. This is due to the complicated programmed software that is implemented in any kind of assistive technology devices. Secondly, it is time consuming. To be able to use these devices, it takes days of training to fully understand the working principles of these devices. Thirdly, some of the assistive technology devices are not globally accessible. Some of the countries including Malaysia may not be able to access high-tech assistive equipment or devices because of patent rights and other reasons.

2.4 Types of Background noise that are usually present in environment

1. Impulse Noises (Pop sounds)

Impulse noise, often known as impact noise, is characterised as extremely brief bursts of loud noise lasting less than a second. These sorts of noise are frequently associated with the demolition and construction industries. These sorts of noise are frequently associated with the demolition and construction industries. (Examples: Gunfire, Balloon popping, Explosions)

2. Broadband Noises

Broadband noise, also known as wideband noise, is noise whose sound energy is dispersed over a broad range of audible frequencies, as opposed to narrowband noise. (Examples: Frequencies and Buzzing)

3. Narrowband noises

Narrowband noise occurs when it disperses its energy over a very modest portion of the audible range. It is the opposite of Broadband/Wideband noises. (Example: RFID devices)

4. Electrical noises (Used to record sound/technology)

Electrical noise is caused by more or less random electrical impulses being connected into circuits where they are undesirable, i.e., where they disturb information-carrying signals. Noise occurs on both power and signal circuits, but it becomes an issue when it enters signal circuits. (Example: used to record sound/technology)

5. Irregular noises (traffic, rain, thunder)

Irregular noises can be defined as outer environmental noises that cause randomly in nature areas or community areas. (Example: traffic, rain, thunder)

CHAPTER 3

METHODOLOGY

3.1 Introduction

The study was conducted based on adaptive noise cancellation system also known as ANC system. This project uses adaptive filter and certain adaptive algorithms to compare the performance. An FIR based adaptive filter is used to mimic the desired filter by finding the coefficients that relate to producing the least mean square of the error signal (difference between desired and actual signal). The proposed noise cancellation system block diagram can be implemented into this project. Each part in this system has their own working principle that can able to make this project to be successful. The methods used in this proposed system are aimed to achieve the objectives of this project which will give satisfying results on the performance of control system. These methods are accomplished by implementing a new design that will lean to improve the function and efficiency.

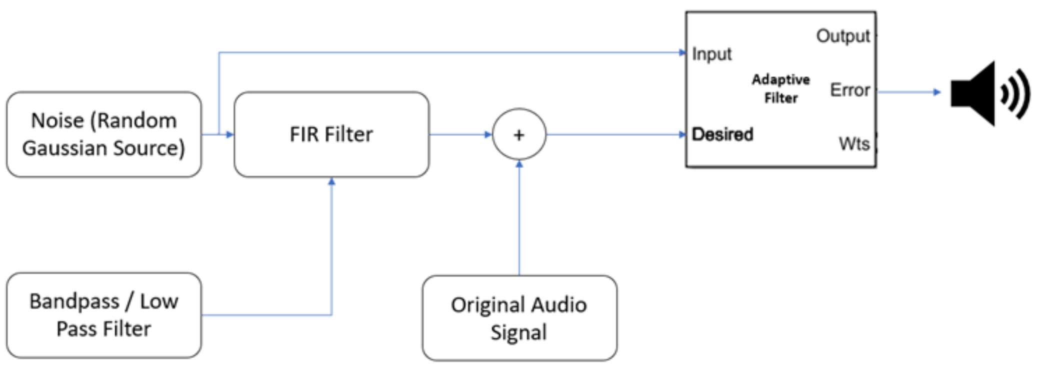

Figure 3.1: Block Diagram Overview

3.2 Adaptive Noise Cancellation System

Random Gaussian Noise of 80dB is generated by using Simulink block. The noise is input into the adaptive filter Input port as reference signal. FIR filter used to shade/filter the noise (using either Bandpass or Lowpass). The shaded noise is then mixed with the original audio signal to produce noise polluted audio signal which is feeded into the Desired port of the adaptive algorithm filter. The sample model uses adaptive filter to expel the unwanted noise from the noise polluted audio signal. Theoretically, over time the adaptive filter in the model will channels out the unwanted noise so that the original audio signal will be obtained. The error output from the LMS, NLMS and RLS filter block will be used to listen to the filtering process output and the root mean square error are calculated.

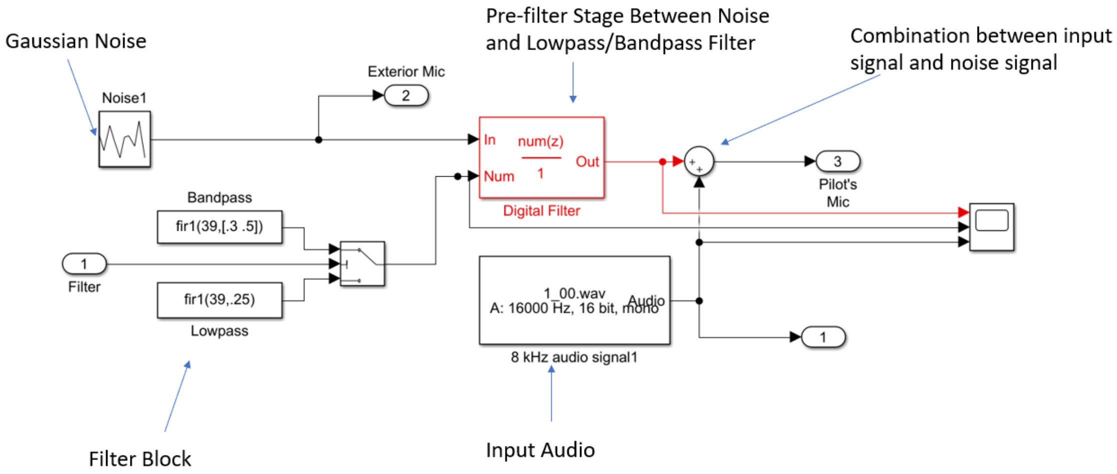

Figure 3.2: Block Diagram Sub (Acoustic Environment)

Figure 3.3 below displays the main adaptive noise cancellation system which the output signals are generated by using Matlab Simulink. Each adaptive algorithm filters are connected to respective output speakers to display the output signal after noise is filtered. The adaptive algorithm filters are Least Mean Square (LMS), Normalized Least Mean Square (NLMS) and Recursive Least Square (RLS).The root mean square error block will act as a catalyst for the original audio signal and output signal after noise filtered.

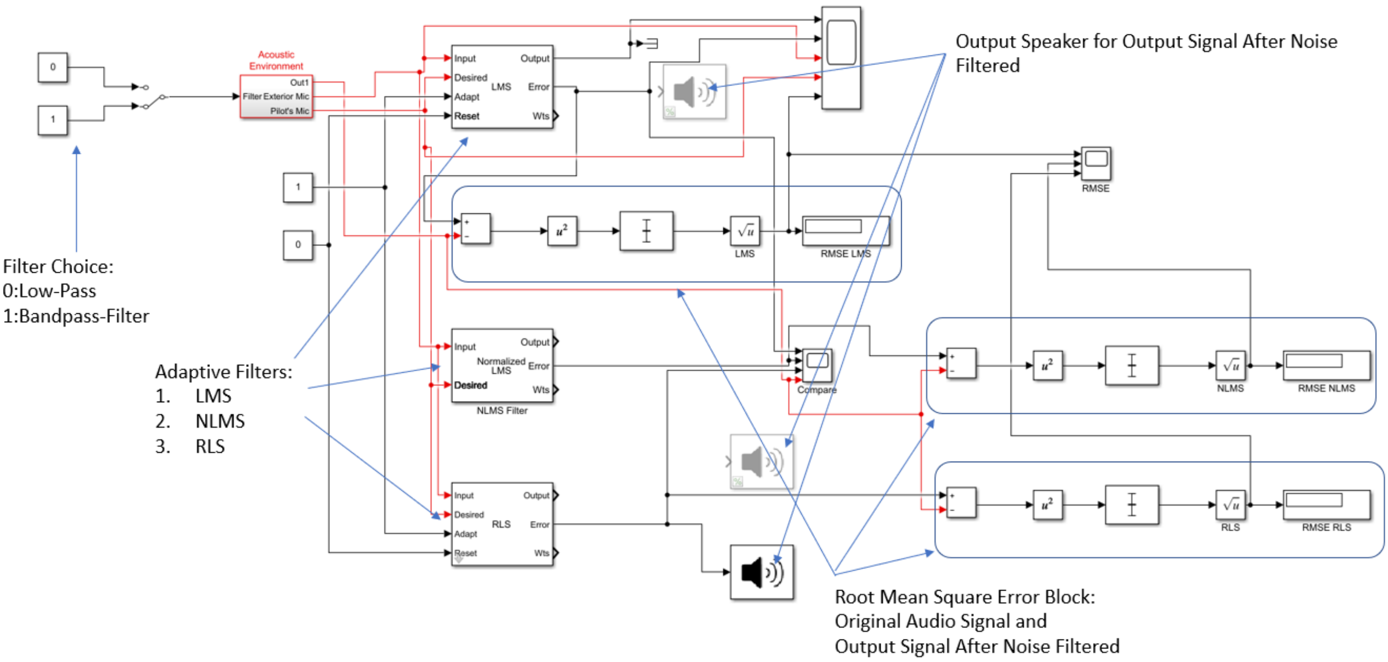

Figure 3.3: Block Diagram (System)

3.3 Adaptive Algorithm Filters

3.3.1 Least Mean Square (LMS)

The least mean square algorithm employs a technique known as "method of steepest descent" to estimate outcomes continually by adjusting filter weights. The least mean square method, based on the notion of algorithm convergence, gives specific learning curves helpful in machine learning theory and practise. Below shows the coefficients and principles of this adaptive algorithm.

a) Based on a gradient search algorithm

b) Output signal, $y(n) = \mathbf{w}(n)\mathbf{x}(n)$, $y(n) = \mathbf{w}\mathrm{T}(n)\mathbf{x}(n)$.

c) The error signal calculated as the difference between the required and the output signal: $e(n) = d(n) - y(n)$,

d) Recursion of filter: $\mathbf{w}(n + 1) = \mathbf{w}(n) + 2\mu e(n)\mathbf{x}(n)$, (2) (3.1)

e) Where: w(n) is the current weight value vector, w(n+1) is the next weight value vector, x(n) is the input signal vector, e(n) is the filter error vector and μ is the convergence factor which determine the filter convergence speed and overall behaviour.

f) The filter step size is parameter that can improve the adaptive filter convergence rate.

3.3.2 Normalized Least Mean Square (NLMS)

The Normalized Least Mean Square (NLMS) method belongs to the gradient class of adaptive algorithms and provides a solution to the Least Mean Square (LMS) algorithm's delayed convergence. The Normalized Least Mean Square method is a version of the LMS algorithm in which the learning rate is normalised with respect to the power of the input signal. Below shows the coefficients and principles of this adaptive algorithm.

a) The normalized LMS (NLMS) algorithm is a modified form of the standard LMS algorithm

b) The NLMS algorithm updates the coefficients of an adaptive filter by using the following equation:

c) $\mathbf{w}(n+1) = \mathbf{w}(n) + \mu e(n)\, \mathbf{x}(n) \;\; \| \mathbf{x}(n) \| \; \mathbf{2}$ (3) (3.2)

d) NLMS algorithm is almost identical to LMS, the only difference is a time-varying step size μ(n).

3.3.3 Recursive Least Square (RLS)

The recursive least squares (RLS) algorithm is a recursive application of the well-known least squares (LS) regression algorithm, in which each new data point is used to modify (correct) a previous estimate of the parameters from some linear (or linearized) correlation thought to model the observed system. The approach enables the dynamical application of LS to real-time time series.

a) Calculates the output signal of the adaptive filter according to the following formula: $y(n) = \mathbf{w}\mathrm{T}\ (n-1)\mathbf{x}(n)$

b) Determines error estimate e (k) using the equation: $e(n) = d(n) - y(n)$. (3.3)

c) Updates the filter coefficients according following equations: $\mathbf{w}(n+1) = \mathbf{w}\mathrm{T}(n) + e(n)\mathbf{K}(n)$, where k(n) is the filter coefficient vector, K(n) is a gain vector that is defined as: $\mathbf{K}(n) = P(n)\mathbf{u}(n)\ \lambda + \mathbf{u}\ \mathrm{T}(n)P(n)\mathbf{u}(n)$, where P (k) is the inverse correlation matrix of the input signal, where λ is a small positive constant very close to, but smaller than 1.

3.4 MATLAB Simulink Application

MATLAB is a high-performance programming language used in technical computing. It combines computing, visualisation, and programming in a user-friendly environment in which problems and answers are stated in common mathematical notation. The typical uses of this application are math & computation, algorithm development modelling simulation, data analysis, signal processing and so on.

For this project, I used the MATLAB Simulink software to develop the Simulink model of adaptive noise cancellation system. MATLAB Simulink is the suitable software to attain the respective Simulink blocks of this adaptive noise cancellation system. In terms of developing the block diagram of this adaptive noise cancellation system, this software is the most efficient and more user friendly. It is also quite easy to use this software because most of the functions and block are already set there.

From this MATLAB Simulink software, the simulation can be run by generating the noise cancellation system. From that, can get the output results in the form of filtered signal waveforms. The simulation is done three times with each of adaptive algorithm filters. The filtered signal is obtained along the filtered human speech signal.

Figure 3.4 Logo of MATLAB Simulink Application

3.5 Root Mean Square Error (RMSE)

The Root Mean Square Error (RMSE) is used to calculate the standard deviation of the residuals (prediction errors). Residuals measure how distant the data points are from the regression line; RMSE measures how spread out these residuals are. For all three types of adaptive filters, the RMSE is determined between the original audio signal and the filtered audio signal (LMS, NLMS and RLS).

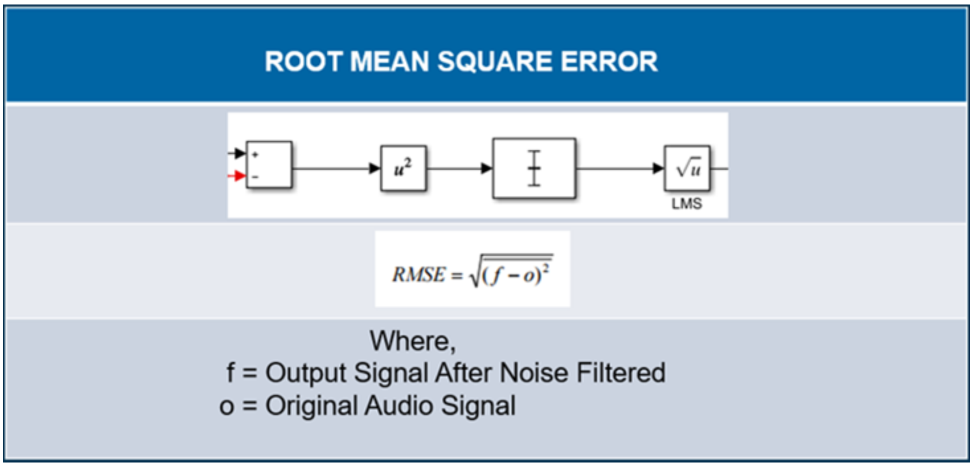

Figure 3.5 Simulink block and formula of RMSE

CHAPTER 4

RESULTS AND DISCUSSION

4.1 Introduction

This chapter will summarises the findings of the study, data analysis, system testing using MATLAB Simulink, and the output signal derived from the acquired data. The results contains the Root Mean Square Error (RMSE), Signal-to-Noise Ratio (SNR), the display of output audio signal and the convergence of the waveforms. The results are achieved by observing each convergence of the adaptive algorithm filters. In the results, we also obtained the audio listening of the noise suppressed human speech signal. The human audio speech signal was listened three times with each adaptive algorithm filters (LMS, NLMS, RLS).

4.2 Root Mean Square Error (RMSE)

RMSE between the original audio signal and the filtered audio signal were calculated for all three types of adaptive filters namely LMS, NLMS and RLS. RLS RMSE approaches zero faster as compared to the other two methods, followed by NLMS and LMS. RLS approaches zero roughly around 3 second mark, NLMS around 13 seconds, whereas LMS more than 30 seconds. Findings shows that the RLS algorithm can filter out the signal with minimal error faster as compared to the other two algorithms.

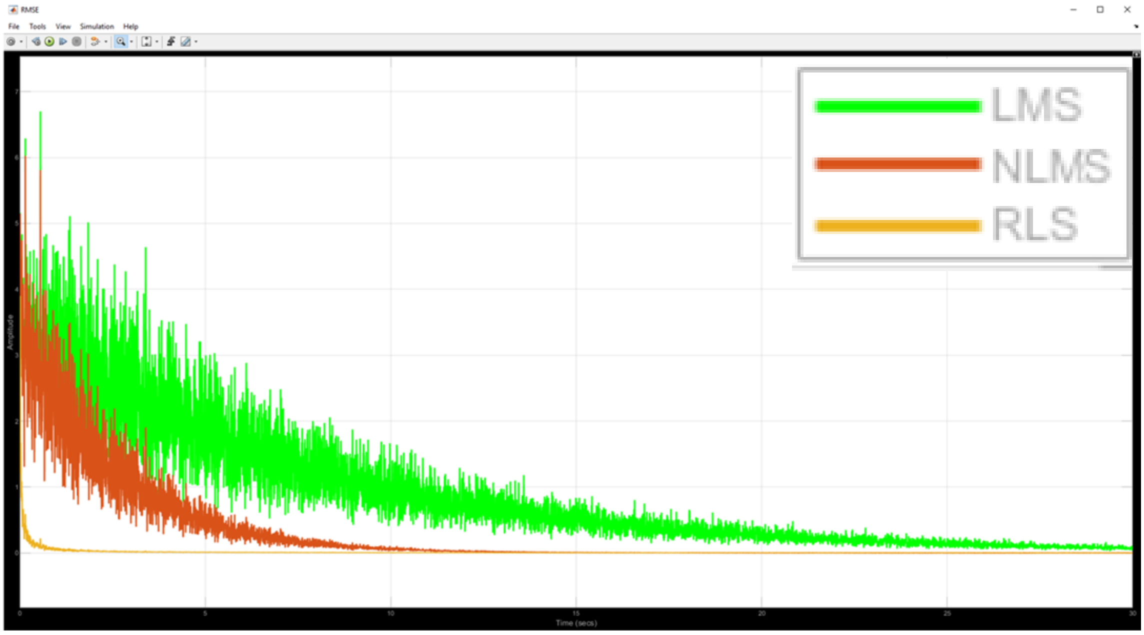

Figure 4.1 RMSE signal for three adaptive algorithm filters

4.3 Output Audio Signals

The output audio signals are also obtained after run the simulation. The waveform for each adaptive algorithm is displayed in terms of different indicating colours. The Acoustic Environment is the human audio speech signal. From the observation the amplitude for LMS filter (yellow) is higher than others which indicates the noise is still diverging.

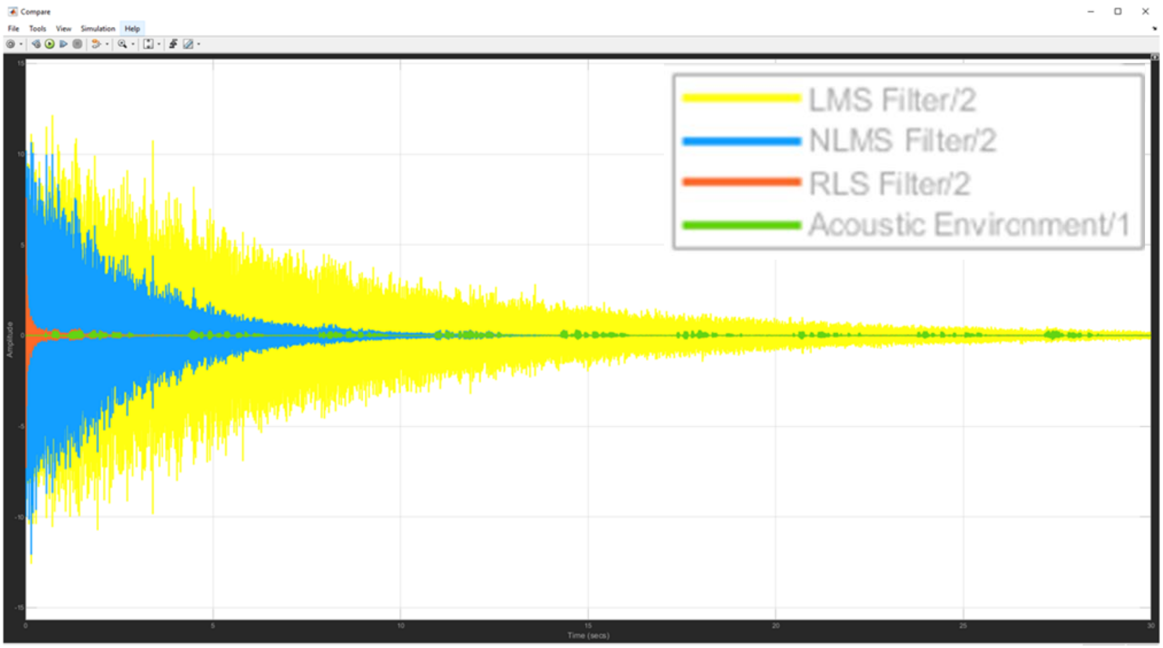

Figure 4.2 output audio signals

4.3.1 The convergence of RLS

RLS output audio signal convergence rate is the fastest among all three adaptive filters. RLS converges around the 7.5 seconds mark with minor deviation from the original audio input signal, while LMS and NLMS still have lot of divergence as compared to audio input signal at this time interval.

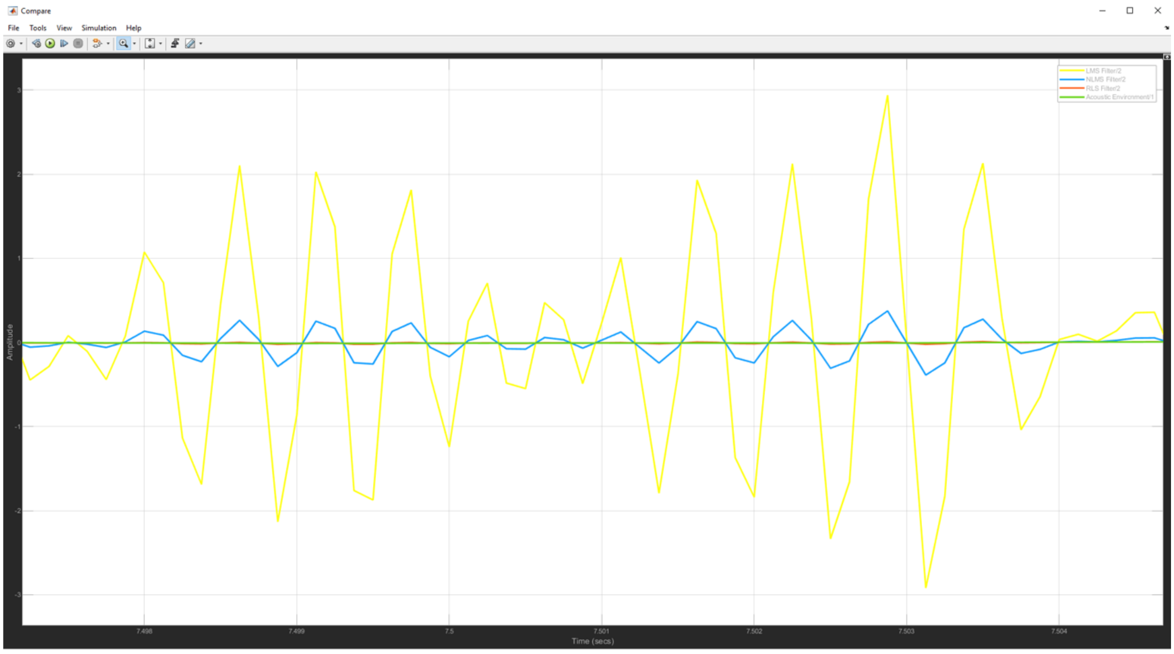

Figure 4.3 the result of RLS algorithm converges at 7.5 seconds mark

4.3.2 The convergence of NLMS

NLMS converges around the 14 seconds mark with minor deviation from the original audio input signal, while LMS still have lot of divergence as compared to audio input signal at this time interval.

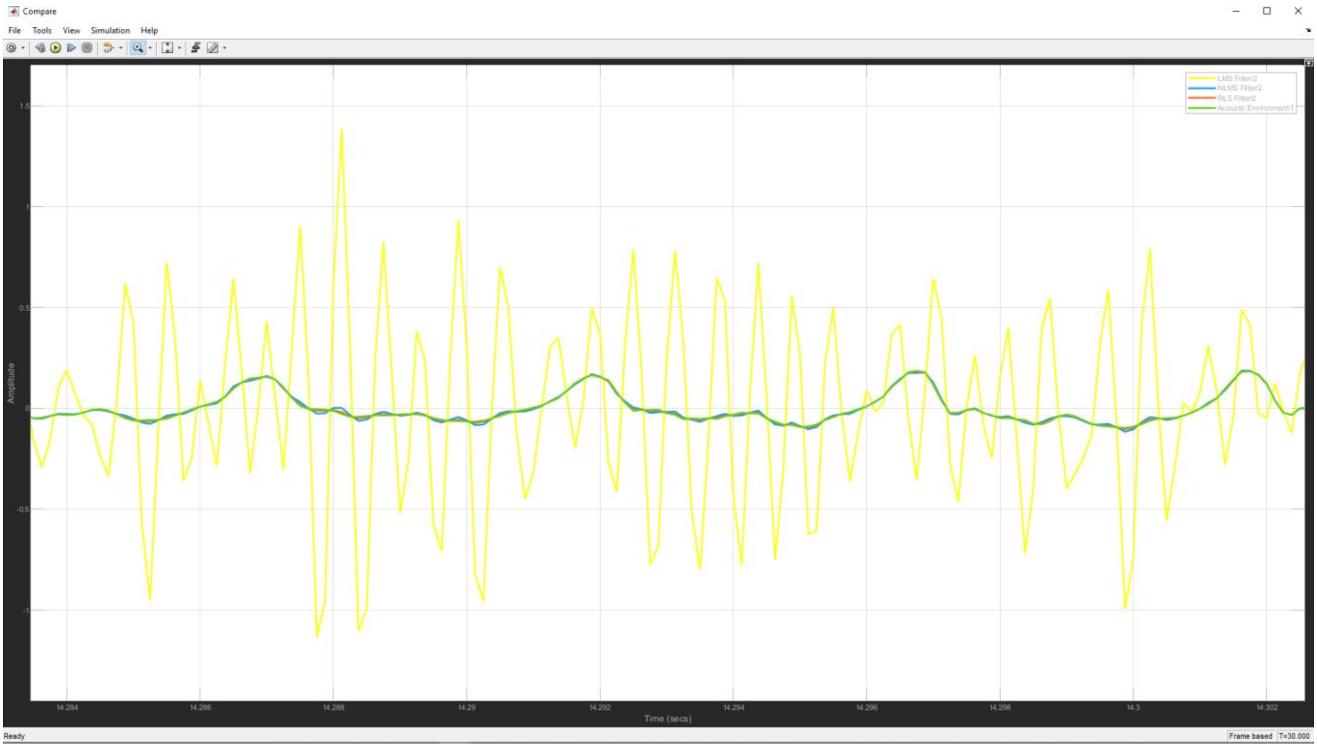

Figure 4.4 the result of NLMS algorithm converges at around 14 seconds mark

4.3.3 The convergence of LMS

Even after the 30 seconds mark, the LMS filter still have not converges with the original audio signal and have significant difference between the original audio signal.

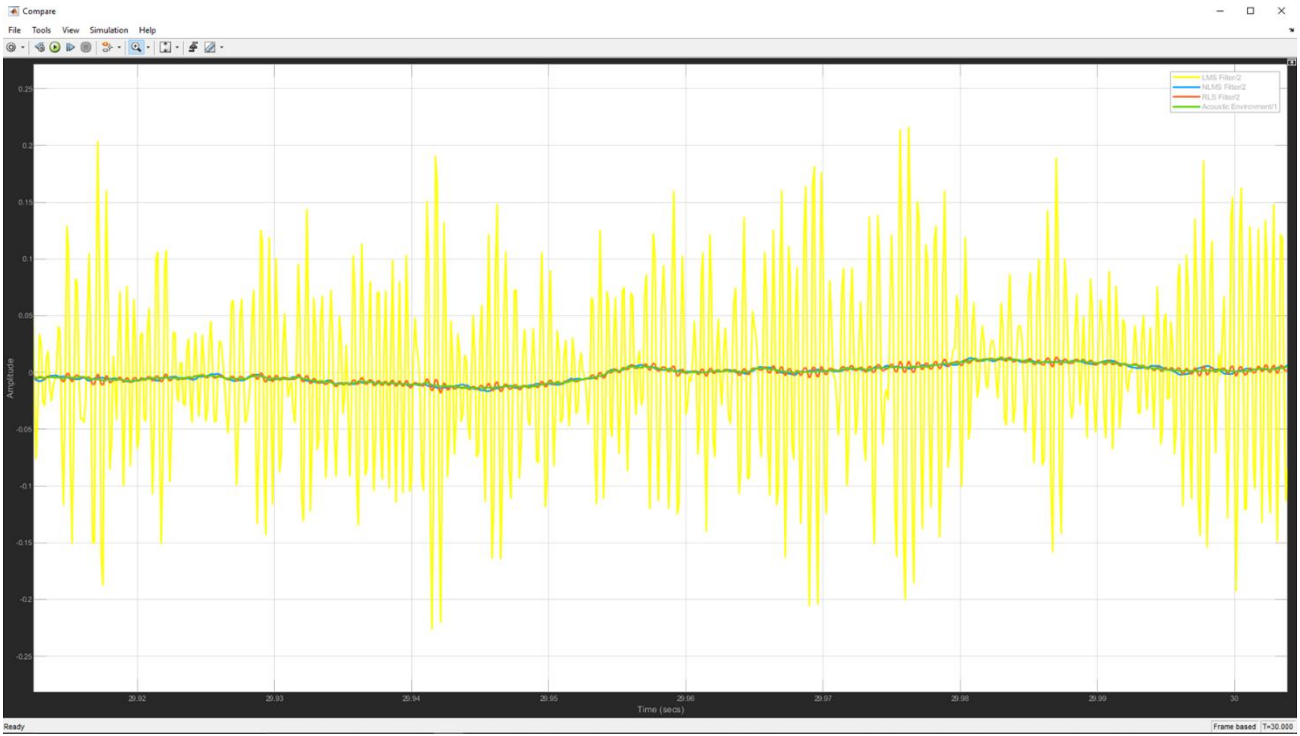

Figure 4.5 The result of LMS algorithm still diverges at 30 seconds mark

4.4 Signal-to-Noise Ratio (SNR)

SNR, or signal-to-noise ratio, is defined as the ratio between the desired information or power of a signal and the undesirable signal or power of background noise. The SNR unit of expression is typically in decibels (dB). The SNR comparison block diagram is added into the main system to calculate the output SNR value of the output filtered signal for each adaptive algorithm filters.

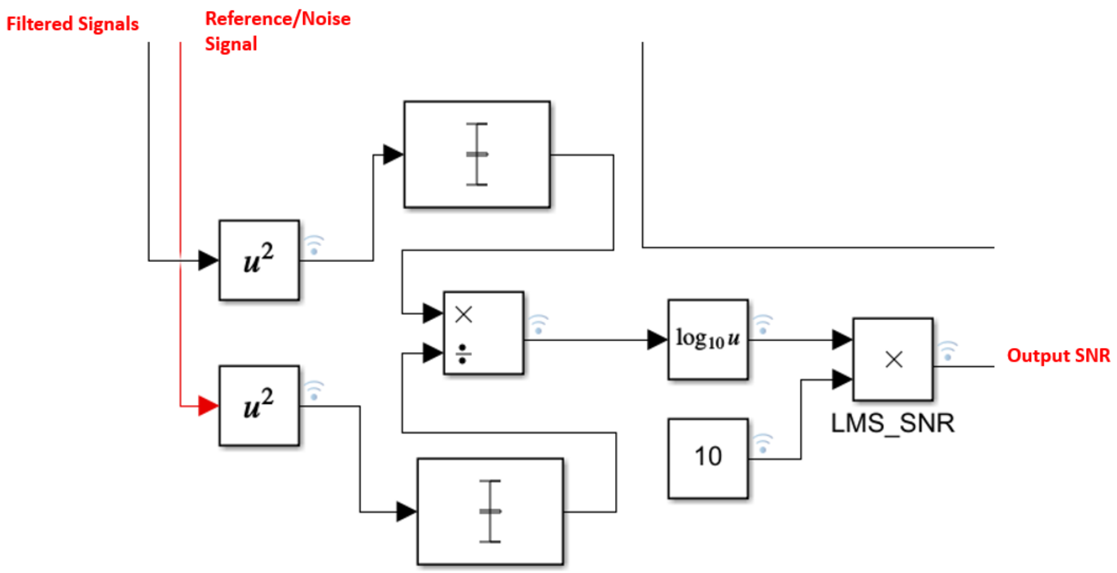

Figure 4.6 The SNR comparison block diagram

4.4.1 Output signal of SNR

The output signal of SNR is generated from the simulation by implementing the block diagram od SNR with the main system. From this, the output results of SNR was obtained.

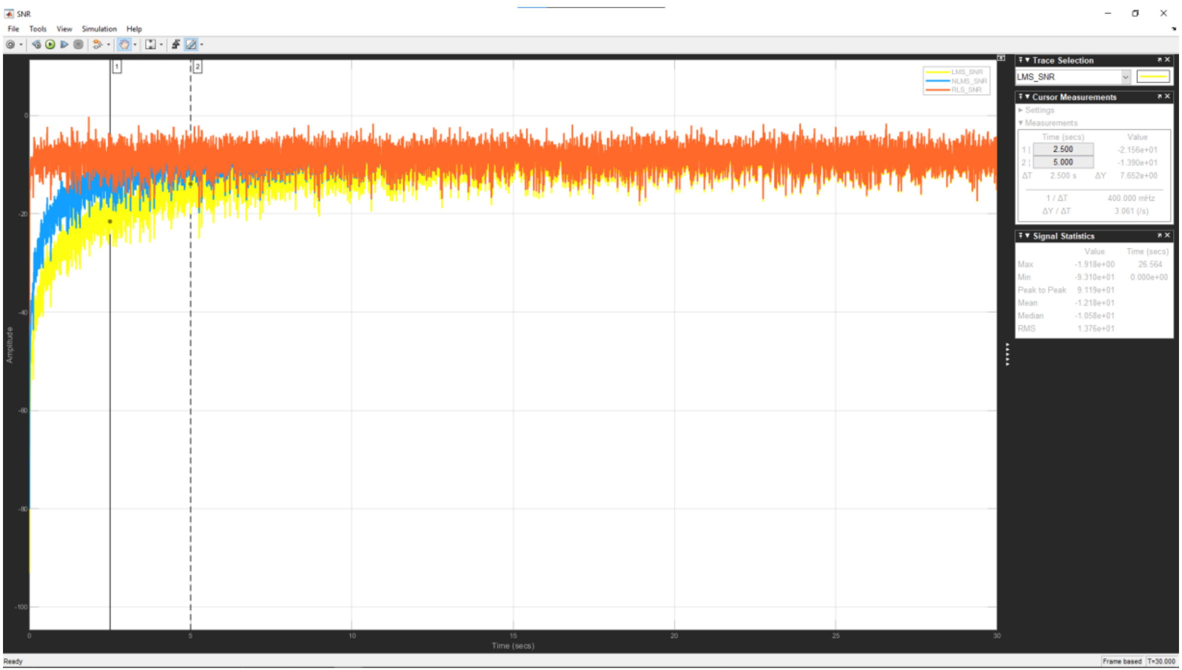

Figure 4.7 The output SNR at 2.5, 5, 10 and 30 seconds

4.4.2 Comparison of SNR values between 3 adaptive algorithm filters

In the 2.5, 5 and 10 second mark, RLS has the highest SNR as compared to LMS and NLMS showing that the noise can be filter out more efficiently and faster in comparison. NLMS and RLS has reach highest SNR of -5.65dB at 30s mark whereas LMS have lower SNR value in comparison.

Table 4.1 SNR values at each specific time interval

Time (s)/Type	**2.5 s**	**5 s**	**10 s**	**30 s**
LMS	-21.56 dB	-13.90 dB	-10.59 dB	-5.84 dB
NLMS	-14.17 dB	-8.61 dB	-7.88 dB	-5.65 dB
RLS	-10.13 dB	-7.36 dB	-7.73 dB	-5.65 dB

CHAPTER 5

CONCLUSION

5.1 Introduction

This chapter will discuss about the overall brief summary of conducting this project and potential suggestions for future recommendations for this project. It also discusses the comparison of the overall performances of each adaptive algorithm filters in the noise cancellation system. It also discusses whether the objectives are achieved and the method implemented on this final year project. By observing the overall performances, I can conclude that my final year project 2 this semester is more effective and improved compare to the final year project 1 in terms of data analysis and simulation process.

5.2 Conclusion

In this project, the main objective is to develop an adaptive noise cancellation system by implementing the digital filtering technique from taking inspiration of assistive listening system. The literature review that I researched from various sites of website and media in this semester and previous semester has helped me a lot to gain some inputs on handling this project. As a conclusion, the objectives of this project have been achieved by following each of the procedures I set for this project. In the adaptive noise cancellation system, I can able to eliminate high-noise up to 80dB. The MATLAB Simulink software has assisted me a lot in developing this system to get a proper output results. From the results we get, we can conclude that RLS shows superior performance in terms of convergence speed and RMSE metrics followed by NLMS and LMS. RLS approximately converges with the original audio input signal after 7.5 second mark and reaches almost zero RMSE around the 3 second mark. The overall performance and output results of RLS is greater than the other two. Noise plays a major part in humans life disrupting and interrupting in various ways in which one of them in speech signals. By implementing this method humans can be able to overcome this problem for better listening environment.

5.3 Recommendation for Future Work

There are some suggestions and improvement that can be possible for this project. The filter can be fabricated and used for real-time audio input filtering to approximate actual environment conditions. It is recommended that the final year project design can be implemented in hardware system to test the functionality of the project design. This method can be implemented in various types of assistive listening system devices such as hearing aids, earmuff, and other assistive listening related devices. This will be very useful especially for the hearing impaired persons and people who work in company production areas and noisy environmental areas.

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