VISVESVARAYA TECHNOLOGICAL UNIVERSITY
Jnana Sangama, Belagavi - 590 018

INTERNSHIP PROJECT REPORT ON

NOISE CANCELLATION USING KALMAN FILTER

Thesis submitted in partial fulfillment for the Award of Degree of

Bachelor of Engineering

in

Electronics and Communication Engineering

Submitted by

Sushrutha S Athreya	1RN21EC147
Syed Raheel	1RN21EC148
T G Sathyanarayana Reddy	1RN21EC149
Tanmay S Badgeri	1RN21EC150
Tejas S Nayak	1RN21EC151

Under the Guidance of

Dr Prabhavathi C N

Associate Professor

DEPARTMENT OF ELECTRONICS AND COMMUNICATION ENGINEERING
(Accredited by NBA for the Academic years 2021-25)

RNS INSTITUTE OF TECHNOLOGY
(AICTE Approved, VTU Affiliated and NAAC 'A' Accredited) (UG Programs - CSE, ECE, ISE, EIE and EEE have been Accredited by NBA for the Academic years 2021-25)
Channasandra, Dr.Vishnuvardhan Road, Bengaluru-560098

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Certified that the Thesis work entitled **"NOISE CANCELLATION USING KALMAN FILTER"** is carried out by **Sushrutha S Athreya (1RN21EC147), Syed Raheel(1RN21EC148), T G Sathyanarayan Reddy(1RC21EC149), Tanmay S Badgeri(1RN21EC150) and Tejas S Nayak(1RN21EC151)** in partial fulfillment for the award of degree of Bachelor of Engineering in **Electronics and Communication Engineering** of Visvesvaraya Technological University, Belagavi, during the year 2021-2022. It is certified that all corrections / suggestions indicated during internal assessment have been incorporated in the report. The project report has been approved as it satisfies the academic requirements in aspect of the internship project work prescribed for the award of degree of **Bachelor of Engineering**.

....	....	....
Dr Prabhavathi C N	Dr. Vipula Singh	Dr. M K Venkatesha
Associate Professor	Head of the Department	Principal

External Viva

Name of the examiners	**Signature with date**
1	....
2	....

Acknowledgement

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Sushrutha S Athreya

Syed Raheel

T G Sathyanarayana Reddy

Tanmay S Badgeri

Tejas S nayak

Abstract

Speech is one of the basic signals used in our daily life. It is impossible, even in this era of advanced technology, to imagine life without speech signals or speech, in particular. This means that with every passing minute, enhancements in the field of speech and hearing are being made with new technologies being discovered or invented every day. The applications of speech technology are many. Some essential commercial applications are Hearing Aids, Noise-cancelling Headphones, audio systems. Improvements in the field of speech technology are dynamic and ongoing. As it has gained more and more importance over the past few decades, we can safely say that the last of the research in speech enhancement has not yet been seen, despite repeated successes in the said field suggesting so. With the advent of smaller and smaller devices and technologies even in the field of speech and hearing, such as small inner ear invisible hearing aids, research is on to constantly improve the experience of users and listeners, including those of the specially abled. Thus, it is with this idea that our project was taken up with enthusiasm and optimism.

In this internship project work, MATLAB has been used to create a noise suppression technique.

Table of Contents

Abstract **ii**

Table of Contents **iii**

List of Figures **1**

1 Introduction **2**
1.1 Motivation 2
1.2 Problem statement 2
1.3 Internship Objective 2
1.4 Internship 3
1.5 Organisation of Report 5

2 Noise and Noise Cancellation **6**
2.1 Speech Enhancement and Noise 6
2.2 Sound Level Meter 6
2.3 Environmental Noise 7
2.4 Noise Cancellation 8
2.5 Filter technique 8

3 Kalman Filter and algorithm **10**

4 Applications of Kalman Filter **14**
4.1 Autopilot: 14
4.2 Battery state of charge (SoC) estimation: 14
4.3 Brain-computer interface: 14
4.4 Dynamic positioning: 14
4.5 Satellite navigation systems: 15
4.6 Seismology: 15
4.7 Weather forecasting: 15
4.8 Navigation system: 16
4.9 3D modelling: 16

5 INTERNSHIP PROJECT **17**
5.1 OUR PROGRAM 17
5.2 Spectrogram of the signals 19

5.3 Periodogram of the signals 20
5.4 Plots of the signal 20

6 Conclusion **23**

References **24**

Appendices **25**

A MATLAB **25**

B Signal Processing Toolbox **26**

C Control System Toolbox **27**

List of Figures

2.1 Sound Level Meter 6

3.1 Algorithm of Kalman filter 13

5.1 Spectrogram of original signal 19
5.2 Spectrogram of filtered signal 19
5.3 Periodogram of original signal 20
5.4 Periodogram of filtered signal 20
5.5 Original signal 21
5.6 Random noisy signal 21
5.7 Combination of original and noisy signal 21
5.8 Estimated filtered signal 22

Chapter 1

Introduction

By utilising a variety of algorithms, voice enhancement seeks to enhance speech quality. The goal of enhancement is to use audio signal processing techniques to make the damaged speech signal more understandable and/or better overall. Three basic categories can be used to group voice improvement for noise reduction algorithms: filtering techniques, spectrum restoration, and model-based approaches.

1.1 Motivation

One of the fundamental signals we utilise on a daily basis is speech. Even in this highly technological age, it is inconceivable to envision a world without speech signals or speech in general. This indicates that improvements in speech and hearing are being achieved every minute, and new technologies are being found or created daily. There are numerous uses for speech technology. Among the most important commercial uses are audio systems, noise-cancelling headphones, and hearing aids.

1.2 Problem statement

To enhance speech signal effectively using active noise cancellation technique and implementing it using Kalman filter. The input signals and output signals are then depicted using a spectrogram and periodogram.
The above technique is to improve audio signal efficiency for better application purposes.

1.3 Internship Objective

Most people assume that internships are only available to college students who want to obtain experience in a certain subject. But a wide range of individuals might gain from training internships in order to gain practical experience and hone their abilities. Technical students are exposed to an industrial setting that cannot be replicated in a classroom. Give people the opportunity to gain knowledge of, and hone the technical

and managerial abilities needed for the job. Expose yourself to recent technology advancements that are pertinent to the training topic. Use the knowledge you obtained from your industrial internship in your class discussions.
Create an environment that encourages knowledge acquisition and its use in the workplace. Enhance your report-writing skills for technical works and projects. Learn about numerous products, procedures, and applications, including Matlab Simulink and the Latex documentation tool.

1.4 Internship

In accordance with the VTU academic calendar for 2021, the internship program was introduced to us in October. Here, some of the key areas of electronics and communication were introduced to us. In a range of fields and domains, such as computer engineering, solid-state physics, robotics, signal processing, telecommunication, and radio engineering, it was successful at indicating electronic techniques and concepts. Some of the domains revised are:

- Signals and systems
- Photonics
- Communication
- VLSI

1.4.1 Signals and Systems

In signal processing, a signal is a function that sends data about a phenomenon. Any quantity that has the potential to change across time or space can be used as a signal to exchange information between observers. Signal processing includes audio, video, voice, picture, sonar, and radar as examples of signals. Even if it does not carry information, a signal can be described as any observable change in a quantity over time or place (a time series). Some are:

- Speech emotion recognition
- Audio signal compression
- Active noise cancellation
- Noise supression
- Image processing

1.4.2 Photonics

The study of light waves is called photonics. It discusses the science involved in the production, detection, and control of light. The wave-particle duality of light is a dual nature. In other words, light possesses properties of both a continuous electromagnetic wave and a particle (photon). Depending on the type of interaction being watched, a certain type of light may be used here we learned about:

- Simulation of an optical waveguide
- Fiber Bragg Grating sensing application

1.4.3 Communication

A communication system is a collection of individual telecommunications networks, transmission systems, relay stations, tributary stations, and terminal equipment usually capable of interconnection and interoperation to form an integrated whole. Some of the main domain we learnt in communication are:

- Standard amplitude modulator and demodulator
- DC motor speed control
- Parity encoder/Decoder
- Design of pulse code modulation/demodulation
- CDMA mux/demux

1.4.4 VLSI

Very large-scale integration (VLSI) is the process of creating an integrated circuit (IC) by combining millions or billions of MOS transistors onto a single chip. VLSI began in the 1970s when MOS integrated circuit (Metal Oxide Semiconductor) chips were developed and then widely adopted, enabling complex semiconductor and telecommunication technologies. The microprocessor and memory chips are VLSI devices.
Before the introduction of VLSI technology, most ICs had a limited set of functions they could perform. An electronic circuit might consist of a CPU, ROM, RAM and other glue logic. VLSI enables IC designers to add all of these into one chip.some of the main domain we learnt in VLSI are:

- Ring Counter -Johnson Counter
- 2:4 DECODER using NAND

- Detection of Even and Odd numbers
- Design of 4:1 MUX using basic gates / universal gates

1.5 Organisation of Report

Chapter 2 :Description of Noise and Noise Cancellation

Chapter 3 :Kalman Filter briefly explained with algorithm

Chapter 4 :Some applications of Kalman Filter

Chapter 5 :Internship Project carried out with code

Chapter 6 :Results achieved in the process

Chapter 7 :Conclusion of the internship project

Chapter 2

Noise and Noise Cancellation

2.1 Speech Enhancement and Noise

Speech enhancement[4] improves the performance of digital communications, speech pre-processing for hearing aids, and speech recognition, and it describes an algorithm to enhance perceived speech quality, reduce hearing fatigue, and improve speech intelligibility.Noise is any unwanted signal, random or deterministic, which interfere with the faithful reproduction of the desired signal in a system. This interfering signal is usually noticed as random fluctuations in voltage or current tending to obscure and mask the desired signals. Noise reduction from noisy speech is the most crucial step of monaural speech enhancement methods because the quality of the enhanced speech depends on the accurate estimation of the noise-power spectrum.
Noise is any unwanted signal, random or deterministic, which interfere with the faithful reproduction of the desired signal in a system. This interfering signal is usually noticed as random fluctuations in voltage or current tending to obscure and mask the desired signals.Natural noise comes from random thermal motion of electrons, atmospheric absorption and cosmic sources. Their interference can best be described statistically

2.2 Sound Level Meter

Figure 2.1: Sound Level Meter

Sound is measured based on the amplitude and frequency of a sound wave. Amplitude measures how forceful the wave is. The energy in a sound wave is measured in decibels (dB), the measure of loudness, or intensity of a sound; this measurement describes the amplitude of a sound wave. Decibels are expressed in a logarithmic scale. On the other hand, pitch describes the frequency of a sound and is measured in hertz (Hz). The main instrument to measure sounds in the air is the Sound Level Meter. There are many different varieties of instruments that are used to measure noise - Noise Dosimeters are often used in occupational environments, noise monitors are used to measure environmental noise and noise pollution, and recently smartphone-based sound level meter applications are being used to crowd source and map recreational and community noise. A-weighting is applied to a sound spectrum to represent the sound that humans are capable of hearing at each frequency. Sound pressure is thus expressed in terms of dBA. 0 dBA is the softest level that a person can hear. Normal speaking voices are around 65 dBA. A rock concert can be aboUT 120dBA.

2.3 Environmental Noise

Environmental noise is an accumulation of noise pollution that occurs outside. This noise can be caused by transport, industrial, and recreational activities. Noise is frequently described as 'unwanted sound'. Within this context, environmental noise is generally present in some form in all areas of human, animal, or environmental activity. The effects in humans of exposure to environmental noise may vary from emotional to physiological and psychological. Noise at low levels is not necessarily harmful. Environmental noise can also convey a sense of liveliness in an area, which can be desirable. The adverse effects of noise exposure (i.e. noise pollution) could include: interference with speech or other 'desired' sounds, annoyance, sleep disturbance, anxiety, hearing damage and stress-related cardiovascular health problems. As a result, environmental noise is studied, regulated, and monitored by many governments and institutions around the world. This creates a number of different occupations. The basis of all decisions is supported by the objective and accurate measurement of noise. Noise is measured in decibels (dB) using a pattern-approved sound level meter. The measurements are typically taken over a period of weeks, in all weather conditions.

2.4 Noise Cancellation

The elimination of unwanted signals[3] in an electronic circuit Noise Cancellation types and settings : Passive Noise Cancellation- Uses well designed ear cups to seal out unwanted noise. This is used for both over-ear headphones and in-ear earphones where the earbud itself will keep surrounding noise out. Active Noise Cancellation - Uses microphones and speakers to reduce background and surrounding noises. This is the most known type and has mostly been used in over-ear headphones. Technology has become so small and battery efficient now that it can be used in true wireless in-ear earphones. Adaptive Active Noise Cancellation -Uses microphones and speakers to automatically adjust to your surroundings. This is the more sophisticated type of ANC where the level of noise cancelling digitally adapts to the surroundings. Adjustable Active Noise Cancellation -Lets you change how much background noise you hear by manually adjusting noise cancellation levels. This is useful when you want to have full control.Transparency Mode lets you easily tune back into the world around you, without switching off your music or taking your earphones out.

2.5 Filter technique

2.5.1 Spectral subtraction method

Spectral subtraction method is a well-known noise reduction method based on the STSA estimation technique. In this proposed approach the output of PDE technique is applied as an input for the spectral subtraction method.

2.5.2 Signal subspace approach

The wavelet-based Wiener filtering which is used in suppressing additive noise requires the calculation of Wiener gains given as, Noise segments were detected using a voice activity detector (VAD).

2.5.3 Subspace approach (SSA)

In signal processing, signal subspace methods are empirical linear methods for dimensionality reduction and noise reduction. These approaches have attracted significant interest and investigation recently in the context of speech enhancement, speech modelling. and speech classification research.

2.5.4 Minimum mean square error

If the input SNR is known, the minimum mean square error technique[5] can be used. With the help of this technique, the estimated error caused by noise and clutter is reduced. It is a Weiner Filter implementation. The STSA estimation problem asks us to infer, given the noisy speech in that analysis frame, the modulus of each complex Fourier expansion coefficient of the speech signal. The Fourier Transform's Fourier Expansion Series are samples of it and are closely related. The relationship between the discrete Fourier transform and the Fourier series expansion allows for the efficient implementation of the technique using the FFT.

Chapter 3

Kalman Filter and algorithm

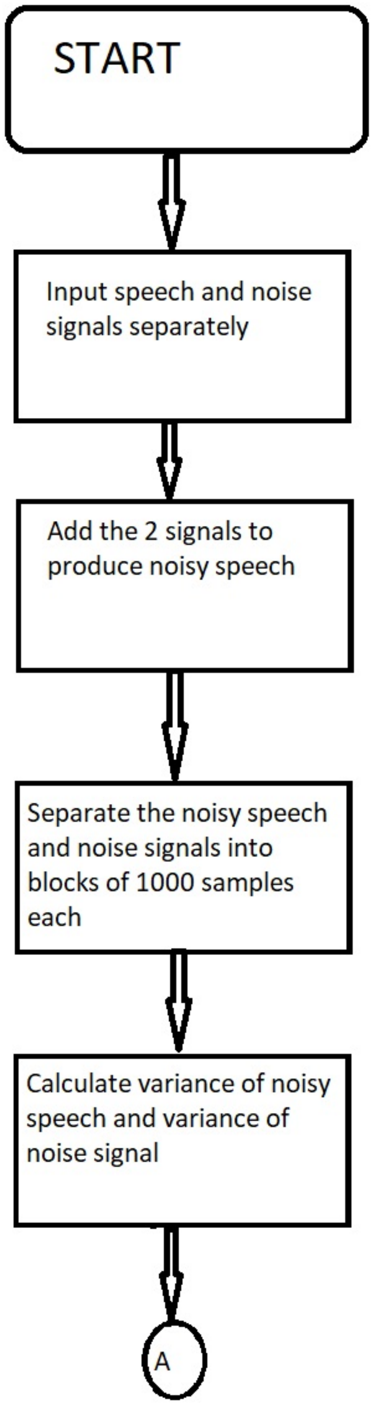
The Kalman filter[1], also known as linear quadratic estimation (LQE), is an algorithm that uses a of measurements observed over time, containing noise (random variations) and other inaccuracies, and produces estimates of unknown variables that tend to be more precise than those based on a single measurement alone. More formally, the Kalman filter operates recursively on streams of noisy input data to produce a statistically optimal estimate of the underlying system state. The filter is named for Rudolf (Rudy) E. Kalman, one of the primary developers of its theory.

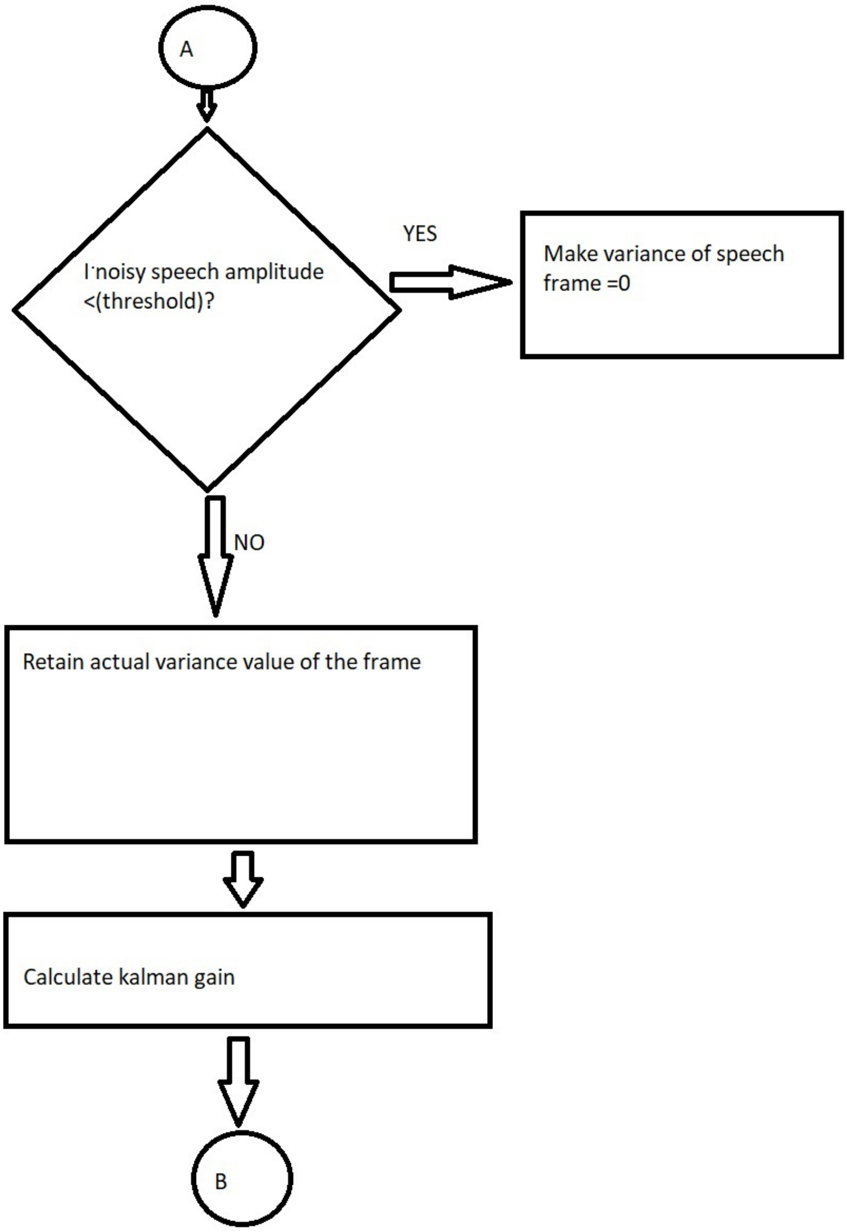
The Kalman has numerous applications.The common application is for guidance, navigation and control of vehicles, particularly aircraft and spacecraft. Furthermore, the Kalman filter is a widely applied concept in time series analysis used in fields such as signal processing and econometrics.

The algorithm works in a two-step process. In the prediction step, the Kalman filter produces estimates of the current state variables, along with uncertainties. Once the outcome of the next measurement (necessarily corrupted with some amount of error, including random noise) is observed, these estimates are updated using a weighted average, with more weight being given to estimates with higher certainty. Because of the algorithm's recursive nature, it can run in real time using only the present input measurements and the previously calculated state and its uncertainty matrix; no additional past information is required.

It is a common misconception that the Kalman filter assumes that all error terms and measurements are Gaussian distributed. Kalman's original paper derived the filter using orthogonal projection theory to show that the covariance is minimized, and this result does not require any assumption, e.g., that the errors are Gaussian. He then showed that the filter yields the exact conditional probability estimate in the special case that all errors are Gaussian-distributed.

FLOWCHART


START
Input speech and noise signals separately
Add the 2 signals to produce noisy speech
Separate the noisy speech and noise signals into blocks of 1000 samples each
Calculate variance of noisy speech and variance of noise signal
A


A
I·noisy speech amplitude
<(threshold)?
YES
Make variance of speech
frame =0
NO
Retain actual variance value of the frame
Calculate kalman gain
B

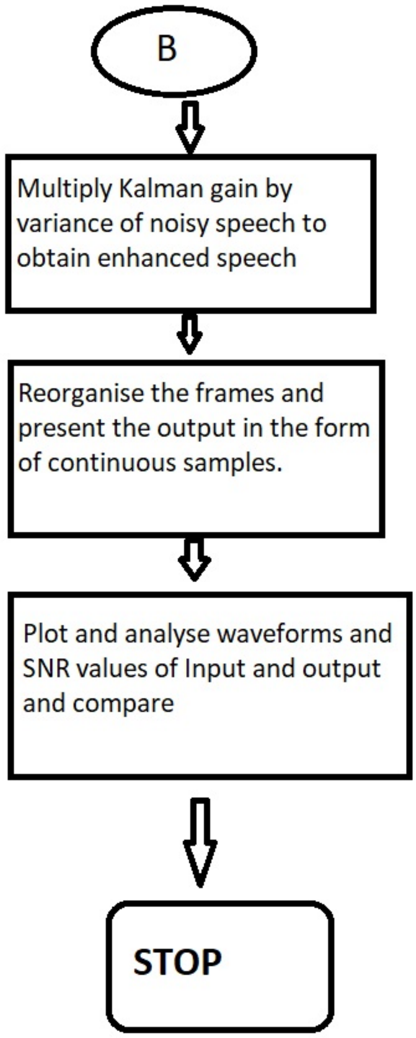


Figure 3.1: Algorithm of Kalman filter

Chapter 4

Applications of Kalman Filter

4.1 Autopilot:

An autopilot[5] is a system used to control the trajectory of a vehicle without constant 'hands-on' control by a human operator being required. Autopilots do not replace a human operator, but assist them in controlling the vehicle, allowing them to focus on broader aspects of operation, such as monitoring the trajectory, weather and systems. Autopilots are used in aircraft, boats (known as self-steering gear), spacecraft, missiles, and others. Autopilots have evolved significantly over time, from early autopilots that merely held an attitude to modern autopilots capable of performing automated landings under the supervision of a pilot.

4.2 Battery state of charge (SoC) estimation:

State of charge (SOC) is the equivalent of a fuel gauge for the battery pack in a battery electric vehicle (BEV), hybrid vehicle (HEV), or plug-in hybrid electric vehicle (PHEV).

4.3 Brain-computer interface:

A brain-computer interface (BCI), often called a mind- machine interface (MMI), or sometimes called a direct neural interface (DNI), synthetic telepathy interface (STI) or a brain-machine interface (BMI), is a direct communication pathway between the brain and an external device. BCIS are often directed at assisting, augmenting, or repairing human cognitive or sensory-motor functions.

4.4 Dynamic positioning:

Dynamic positioning (DP) is a computer-controlled system to automatically maintain a vessel's position and heading by using its own propellers and thrusters. Position reference sensors, combined with wind sensors, motion sensors and gyro compasses, provide information to the computer pertaining to the vessel's position and the magnitude and direction of environmental forces affecting its position. Examples of vessel types that employ DP include, but are not limited to, ships and semi-submersible mobile offshore drilling units (MODU), oceanographicresearch vessels and cruise ships. The computer program contains a mathematical model of the vessel that includes information pertaining to the wind and current drag of the vessel and the location of the thrusters. This knowledge, combined with the sensor information, allows the computer to calculate the required steering angle and thruster output for each thruster. This allows operations at sea where mooring or anchoring is not feasible due to deep water, congestion on the sea bottom (pipelines, templates) or other problems.

4.5 Satellite navigation systems:

A satellite navigation[2] or sat nav system is a system of satellites that provide autonomous geo-spatial positioning with global coverage. It allows small electronic receivers to determine their location (longitude, latitude, and altitude) to high precision (within a few metres) using time signals transmitted along a line of sight by radio from satellites. The signals also allow the electronic receivers to calculate the current local time to high precision, which allows time synchronisation. A satellite navigation system with global coverage may be termed a global navigation satellite system or GNSS.

4.6 Seismology:

Seismology is the scientific study of earthquakes and the propagation of elastic waves through the Earth or through other planet-like bodies. The field also includes studies of earthquake effects, such as tsunamis as well as diverse seismic sources such as volcanic, tectonic, oceanic, atmospheric, and artificial processes (such as explosions).

4.7 Weather forecasting:

Weather forecasting is the application of science and technology to predict the state of the atmosphere for a given location. Human beings have attempted to predict the weather informally for millennia, and formally since the nineteenth century. Weather forecasts are made by collecting quantitative data about the current state of the atmosphere on a given place and using scientific understanding of atmospheric processes to project how the atmosphere will evolve on that place.

4.8 Navigation system:

A navigation system is a (usually electronic) system that aids in navigation. Navigation systems may be entirely on board a vehicle or vessel, or they may be located elsewhere and communicate via radio or other signals with a vehicle or vessel, or they may use a combination of these methods.

4.9 3D modelling:

In 3D computer graphics, 3D modeling is the process of developing a mathematical representation of any three-dimensional surface of object (either inanimate or living) via specialized software. The product is called a 3D model. It can be displayed as a two-dimensional image through a process called 3D rendering or used in a computer simulation of physical phenomena.

Chapter 5

INTERNSHIP PROJECT

5.1 OUR PROGRAM

The main objective of this Internship project is to understand the implementation of kalman filter in the field of noise cancellation. The program is:

5.1.1 Program for plotting Periodogram

```
[y,fs] = audioread('Path of the audio file');
y = y(:,1);
dt = 1/fs;
t = 0:dt:(length(y)*dt)-dt;
plot(t,y); xlabel('Seconds'); ylabel('Amplitude');
figure

    plot(psd(spectrum.periodogram,y,'Fs',fs,'NFFT',length(y)));
figure
spectrogram(y);
```

5.1.2 Kalman filter program

```
clc;
clear all;
close all;
v=rand (size(x));
orig=x+v;
no=orig;
N=length(x);
F = zeros (5, N);
I = eye (5);
H = zeros (5, N);
sig = zeros (5, 5*N);
```

```
K = zeros (5, N);
XX = zeros (5, N);
y = zeros (1, N);
vv = zeros (1, N);
yy = zeros (1, N);
Q = 0.0001*eye (5, 5)
; R = 0.1;
y=x (1: N);
sig (1:5, 1:5) = 0.1*I;
for k=6: N
F(1:5,k)=-[y(k-1);y(k-2);y(k-3);y(k-4);y(k-5)];
H(1:5,k)=-[yy(k-1);yy(k-2);yy(k-3);yy(k-4);yy(k-5)];
K(1:5,k)=sig(1:5,5*k-29:5*k-25)*F(1:5,k)*inv(F(1:5,k)'*sig(1:5,5*k-29:5*k-25)*F(1:5,k)+R);

sig(1:5,5*k-24:5*k-20)=sig(1:5,5*k-29:5*k-25)-sig(1:5,5*k-29:5*k-25)*F(1:5,k)*inv(F(1:5,k)'*sig(
29:5*k-25)*F(1:5,k) +R)*(F(1:5,k)'*sig(1:5,5*k-29:5*k-25))+Q;

XX(1:5,k) =(I - K(1:5,k)*F(1:5,k)')*XX(1:5,k-1) + (K(1:5,k)*y(k));
orig (k) =y (k)-(F (1:3, k)'*XX (1:3, k));
yy (k) = (H (1:5, k)'*XX (1:5, k)) + orig (k);
end;
for it = 1:1: length(x)
MSE (it) = orig (it) - x (it);
end;
tt = 1:1: length(x);
figure (1);
subplot (311);
plot(x);
title ('ORIGINAL SIGNAL');
subplot (312);
plot (no);
title ('Noisy Speech Signal');
subplot (313); plot (orig); title ('ESTIMATED SIGNAL');
figure (2);
plot (tt, x, tt, orig);
title ('Combined plot');
legend ('original',' estimated');
figure (3);
```

plot (MSE.^{2});
$title('Meansquareerror');$
$soundsc(orig);$

5.2 Spectrogram of the signals

A spectrogram is a visual representation of the spectrum of frequencies of a signal as it varies with time. Spectrograms are used extensively in the fields of music, linguistics, sonar, radar, speech processing,seismology, and others.When applied to an audio signal, spectrograms are sometimes called sonographs, voiceprints, or voicegrams.
The figures, 5.1 and 5.2 depict the spectrograms of the original signal and the filtered signal respectively.

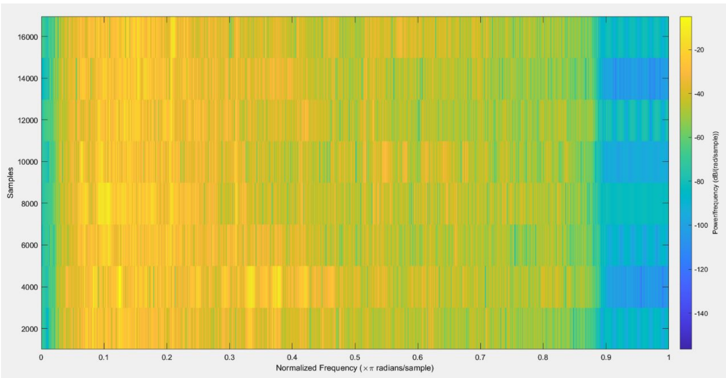


Figure 5.1: Spectrogram of original signal

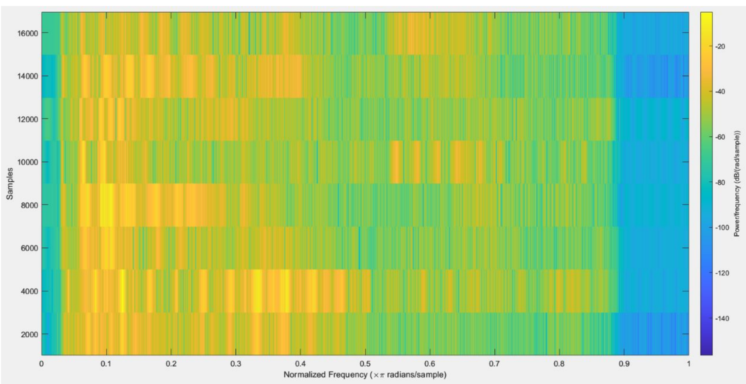


Figure 5.2: Spectrogram of filtered signal

5.3 Periodogram of the signals

In signal processing, a periodogram is an estimate of the spectral density of a signal. It is the most common tool for examining the amplitude vs frequency characteristics of FIR filters and window functions. FFT (fast fourier transform) spectrum analyzers are also implemented as a time-sequence of periodograms.
The figures, 5.3 and 5.4 depict the periodograms of the original signal and the filtered signal respectively.

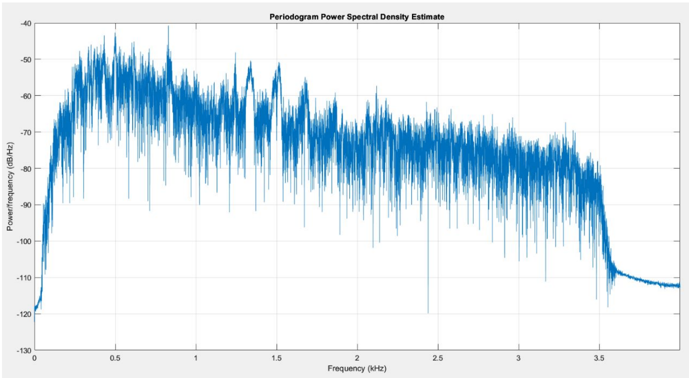


Figure 5.3: Periodogram of original signal

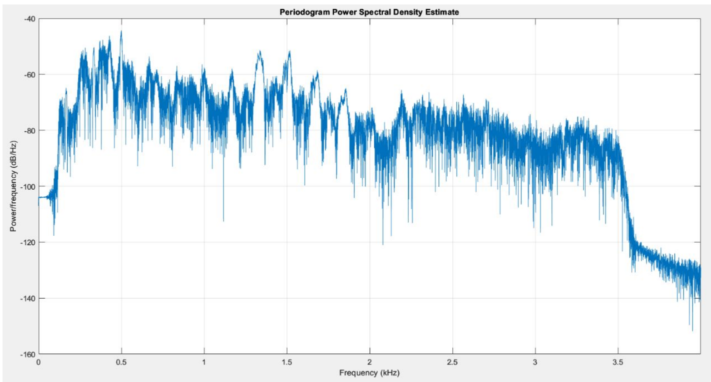


Figure 5.4: Periodogram of filtered signal

5.4 Plots of the signal

To reduce the noise in a signal, we are initially giving a random noise to the system and combining it with the original signal. From the figures 5.5 we know how the original signal is and from figure 5.6, we know the noisy signal added. The combination of both signals would turn out to be like figure 5.7. This combined signal is then separated into blocks of 1000 samples each. This is done to calculate the variance of the noisy speech and noisy signal which is separated.

It is later checked if the noisy speech amplitude is lesser than the threshold and if the condition is satisfied, then the variance is set to 0 and if it does not satisfy, then the original variance value will be retained. Then, the kalman gain is calculated.

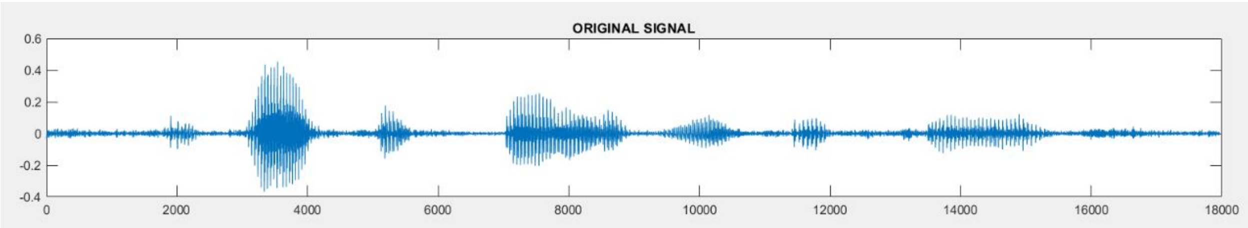


Figure 5.5: Original signal

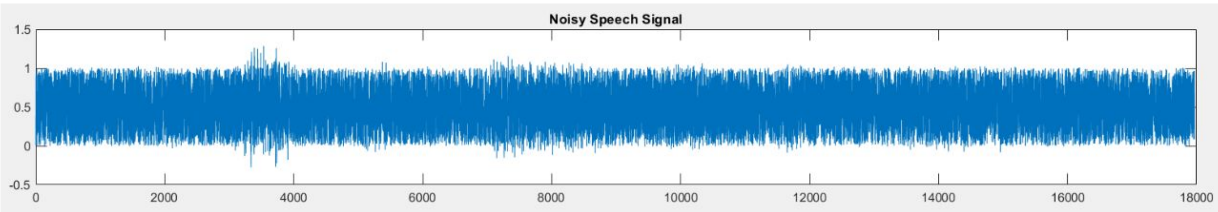


Figure 5.6: Random noisy signal

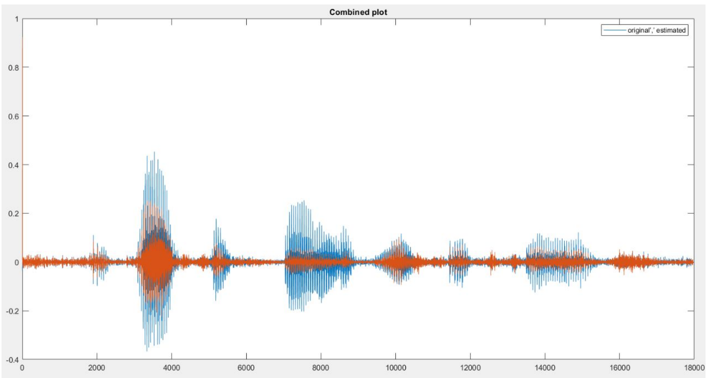


Figure 5.7: Combination of original and noisy signal

The kalman gain is multiplied by the variance of the noisy speech to obtain the enhanced speech. The frames that are sampled are then re-organised in the form of continuous samples. Then the estimated signal is plotted as shown in figure 5.8.

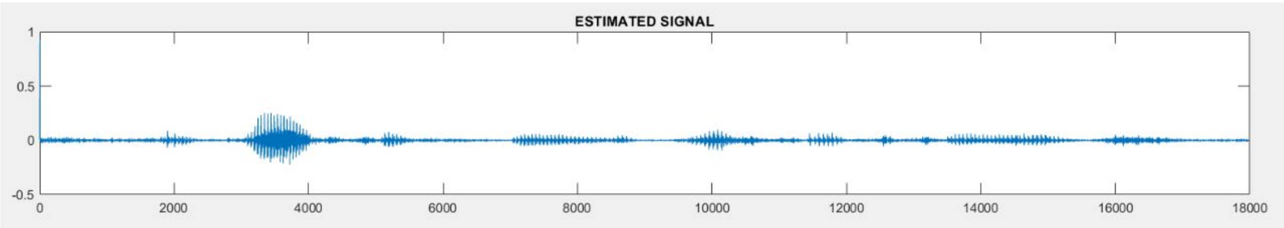


Figure 5.8: Estimated filtered signal

Chapter 6

Conclusion

In a nutshell, this internship has been excellent and rewarding experince. I can say with certainty that working on our project has taught us a a great deal. It goes without saying that the technical aspects of the work we've done could be improved given enough time. Since we had zero prior knowledge on noise cancellation and filters , we feel that the time we invested in learning about it was well worth it and helped me find a workable solution to create a project in the domain of signals and systems. Two main things that we learnt are the importance of soft skills and literature skills required for handling a projects which plays a vital role.

Concluding on the project,we can tell that active noise cancellation is very much necessary and plays a very vital role in future market. The increasing number of smartphones users, smart wearables and tech-savvy consumers and the adoption of new audio technology with increased control over ambient sound are some of the prominent factors that are expected to drive the ANC headphones market.

As India is a densely populated country which throws a diverse assortment of unwanted sound which the consumer wants to block out, thus cementing the positive outlook and potential for this category in India.

There are plenty of features that define a high-end headphone, ANC is one of the most important one since it helps to block out all unwanted sound and lets you hear nothing but your music, whether it's a constant humming of a jet engine while travelling, a busy cafe house or a commuting noisy public transport. ANC is a premium feature which earlier was very expensive. However technological advancements have bought down cost substantially making it much more appealing to a wider audience base. Lots of personalization and fine-tuning can also be achieved through the Headphones Connect app, including adjustments for ambient sound reduction, and the ability to optimize audio based on atmospheric pressure.

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Appendix A

MATLAB

MATLAB (an abbreviation of "MATrix LABoratory") is a proprietary multi-paradigm programming language and numeric computing environment developed by MathWorks. MATLAB allows matrix manipulations, plotting of functions and data, implementation of algorithms, creation of user interfaces, and interfacing with programs written in other languages. Although MATLAB is intended primarily for numeric computing, an optional toolbox uses the MuPAD symbolic engine allowing access to symbolic computing abilities. An additional package, Simulink, adds graphical multi-domain simulation and model-based design for dynamic and embedded systems. As of 2020, MATLAB has more than 4 million users worldwide. They come from various backgrounds of engineering, science, and economics. Few applications of MATLAB –

- Statistics and machine learning(ML)
- Curve fitting
- Control systems
- Signal Processing
- Mapping
- Deep learning
- Financial analysis
- Image processing
- Text analysis
- Aerospace
- Audio toolbox

Appendix B

Signal Processing Toolbox

Signal Processing Toolbox™ provides functions and apps to manage, analyze, pre-process, and extract features from uniformly and non-uniformly sampled signals. The toolbox includes tools for filter design and analysis, re-sampling, smoothing, de-trending, and power spectrum estimation. You can use the Signal Analyzer app for visualizing and processing signals simultaneously in time, frequency, and time-frequency domains. With the Filter Designer app you can design and analyze FIR and IIR digital filters. Both apps generate MATLAB® scripts to reproduce or automate your work.

Using toolbox functions, you can prepare signal datasets for AI model training by engineering features that reduce dimensionality and improve the quality of signals. You can access and process collections of files and large datasets using signal datastores. With the Signal Labeler app, you can annotate signal attributes, regions, and points of interest to create labeled signal sets. The toolbox supports GPU acceleration in addition to C/C++ and CUDA® code generation for desktop prototyping and embedded system deployment.

Appendix C

Control System Toolbox

Control System Toolbox™ provides algorithms and apps for systematically analyzing, designing, and tuning linear control systems. You can specify your system as a transfer function, state-space, zero-pole-gain, or frequency-response model. Apps and functions, such as step response plot and Bode plot, let you analyze and visualize system behavior in the time and frequency domains.

You can tune compensator parameters using interactive techniques such as Bode loop shaping and the root locus method. The toolbox automatically tunes both SISO and MIMO compensators, including PID controllers. Compensators can include multiple tunable blocks spanning several feedback loops. You can tune gain-scheduled controllers and specify multiple tuning objectives, such as reference tracking, disturbance rejection, and stability margins. You can validate your design by verifying rise time, overshoot, settling time, gain and phase margins, and other requirements.