ECO CANCELLATION USING LMS ALGORITHM

A MINOR PROJECT - III REPORT

Submitted by

SRIDHAR R C	**927621BEC210**
SUDHARSUN S	**927621BEC219**
YASAR S	**927621BEC245**
YUVAN SANKAR RAJA S	**927621BEC249**

BACHELOR OF ENGINEERING

in

DEPARTMENT OF ELECTRONICS AND COMMUNICATION ENGINEERING

M.KUMARASAMY COLLEGE OF ENGINEERING

(Autonomous)

KARUR – 639 113

OCTOBER 2023

M.KUMARASAMY COLLEGE OF ENGINEERING, KARUR

BONAFIDE CERTIFICATE

Certified that this **18ECP105L - Minor Project III** report "ECO CANCELLATION USING LMS ALGORITHM**"** is the bonafide work of "**SRIDHAR R C (927621BEC210), SUDHARSUN S (927621BEC219), YASAR S (927621BEC245), YUVAN SANKAR RAJA S(927621BEC249)"** who carried out the project work under my supervision in the academic year **2023-2024- ODD**.

SIGNATURE	**SIGNATURE**
Dr.A.KAVITHA B.E., M.E., Ph.D.,	**Mr.K.SUDHAKAR B.E., ME.,**
HEAD OF THE DEPARTMENT,	**SUPERVISOR,**
Professor,	Assistant Professor.,
Department of Electronics and	Department of Electronics and
Communication Engineering,	Communication Engineering,
M.Kumarasamy College of Engineering,	M.Kumarasamy College of Engineering,
Thalavapalayam,	Thalavapalayam,
Karur-639113.	Karur-639113.

This report has been submitted for the **18ECP105L – Minor Project-III** final review held at M. Kumarasamy College of Engineering, Karur on ________________

PROJECTCOORDINATOR

INSTITUTION VISION AND MISSION

Vision

To emerge as a leader among the top institutions in the field of technical education.

Mission

M1: Produce smart technocrats with empirical knowledge who can surmount the global challenges.

M2: Create a diverse, fully -engaged, learner -centric campus environment to provide quality education to the students.

M3: Maintain mutually beneficial partnerships with our alumni, industry and professional associations

DEPARTMENT VISION, MISSION, PEO, PO AND PSO

Vision

To empower the Electronics and Communication Engineering students with emerging technologies, professionalism, innovative research and social responsibility.

Mission

M1: Attain the academic excellence through innovative teaching learning process, research areas & laboratories and Consultancy projects.

M2: Inculcate the students in problem solving and lifelong learning ability.

M3: Provide entrepreneurial skills and leadership qualities.

M4: Render the technical knowledge and skills of faculty members.

Program Educational Objectives

PEO1: Core Competence: Graduates will have a successful career in academia or industry associated with Electronics and Communication Engineering

PEO2: Professionalism: Graduates will provide feasible solutions for the challenging problems through comprehensive research and innovation in the allied areas of Electronics and Communication Engineering.

PEO3: Lifelong Learning: Graduates will contribute to the social needs through lifelong learning, practicing professional ethics and leadership quality

Program Outcomes

PO 1: Engineering knowledge: Apply the knowledge of mathematics, science, engineering fundamentals, and an engineering specialization to the solution of complex engineering problems.

PO 2: Problem analysis: Identify, formulate, review research literature, and analyze complex engineering problems reaching substantiated conclusions using first principles of mathematics, natural sciences, and engineering sciences.

PO 3: Design/development of solutions: Design solutions for complex engineering problems and design system components or processes that meet the specified needs with appropriate consideration for the public health and safety, and the cultural, societal, and environmental considerations.

PO 4: Conduct investigations of complex problems: Use research-based knowledge and research methods including design of experiments, analysis and interpretation of data, and synthesis of the information to provide valid conclusions.

PO 5: Modern tool usage: Create, select, and apply appropriate techniques, resources, and modern engineering and IT tools including prediction and modeling to complex engineering activities with an understanding of the limitations.

PO 6: The engineer and society: Apply reasoning informed by the contextual knowledge to assess societal, health, safety, legal and cultural issues and the consequent responsibilities relevant to the professional engineering practice.

PO 7: Environment and sustainability: Understand the impact of the professional engineering solutions in societal and environmental contexts, and demonstrate the knowledge of, and need for sustainable development.

PO 8: Ethics: Apply ethical principles and commit to professional ethics and responsibilities and norms of the engineering practice.

PO 9: Individual and teamwork: Function effectively as an individual, and as a member or leader in diverse teams, and in multidisciplinary settings.

PO 10: Communication: Communicate effectively on complex engineering activities with the engineering community and with society at large, such as, being able to comprehend and write effective reports and design documentation, make effective presentations, and give and receive clear instructions.

PO 11: Project management and finance: Demonstrate knowledge and understanding of the engineering and management principles and apply these to one’s own work, as a member and leader in a team, to manage projects and in multidisciplinary environments.

PO 12: Life-long learning: Recognize the need for, and have the preparation and ability to engage in independent and life-long learning in the broadest context of technological change.

Program Specific Outcomes

PSO1: Applying knowledge in various areas, like Electronics, Communications, Signal processing, VLSI, Embedded systems etc., in the design and implementation of Engineering application.

PSO2: Able to solve complex problems in Electronics and Communication Engineering with analytical and managerial skills either independently or in team using latest hardware and software tools to fulfil the industrial expectations.

Keyword	Matching with POs, PSOs
LMS, Adaptive filtering	**PO1, PO2, PO3, PO4, PO5, PO6, PO7, PO8, PO9, PO10, PO11, PO12, PSO1, PSO2**

ACKNOWLEDGEMENT

Our sincere thanks to **Thiru.M.Kumarasamy, Chairman** and **Dr.K.Ramakrishnan, Secretary** of **M.Kumarasamy College of Engineering** for providing extraordinary infrastructure, which helped us to complete this project in time.

It is a great privilege for us to express our gratitude to **Dr.B.S.Murugan., B.Tech., M.Tech., Ph.D., Principal** for providing us right ambiance to carry out this project work.

We would like to thank **Dr.A.Kavitha, B.E., M.E., Ph.D., Professor and Head**, **Department of Electronics and Communication Engineering** for his unwavering moral support and constant encouragement towards the completion of this project work.

We offer our wholehearted thanks to our Project Supervisor, **Mr.K.SUDHAKAR B.E., ME., Assistant Professor.,** Department of Electronics and Communication Engineering for his precious guidance, tremendous supervision, kind cooperation, valuable suggestions, and support rendered in making our project successful.

We would like to thank our **Minor Project Co-ordinator**, **Dr.K.Karthikeyan, B.E., M.Tech., Ph.D., Associate Professor**, Department of Electronics and Communication Engineering for his kind cooperation and culminating in the successful completion of this project work. We are glad to thank all the Faculty Members of the Department of Electronics and Communication Engineering for extending a warm helping hand and valuable suggestions throughout the project. Words are boundless to thank our Parents and Friends for their motivation to complete this project successfully.

ABSTRACT

Speech is the most basic way to convey information from person to another, but in orderto convey this information, we have to deal with the most common problem in speech processing, which is the problem of interference of noise in speech signals. Noise interference in speech processing come from acoustical sources such as equipment, traffic, crowds and commonly echoes. This project report deals with cancelling echo as an unwanted form of speech transmitted over the microphone. Basically, echoes are created when a feedback loop is created, this feedback loop is created when microphone picks up what a speaker is producing during a communication, for example when we speak, our voice which is the near-end speech is picked up by a microphone and is sent toa far location (speaker), the problem here is that when the far location speaks, the audio comes out of the speaker and is picked by the microphone and sent to the far location again, so the far-end will hear it's speech echoed back. Another cause of echo is reverberation; this is the reflections of sound on all the surfaces in a room. When speech is produces in a room, the microphone in that room picks up both the direct speech from the source (person or speaker) and the many reflections made by the room's architecture arrives slightly later than the direct speech causing a delay (echo), together forming one single continuous sound. There are solutions to this problem and one of them is the half- duplex communication, this is a system whereby whenever the far-end is speaking, the near-end microphone is turned off; this means *that* the near-end cannot interject until the far-end has stopped talking (a one sided conversation). The main solution to this problem is an Echo cancellation. Echo cancellation is very important for audio teleconferencing when simultaneous communication of speech is necessary

LIST OF TABLES

LIST OF FIGURES

LIST OF ABBREVIATIONS

AEC	Acoustic Echo Cancellation
ANC	Active Noise Control
APA	Affine Projection Algorithm
CR	Convergence Rate
ERLE	Echo Return Loss Enhancement
FIR	Finite Impulse Response
ISM	Image Source Method
NLMS	Normalized Least Mean Square
PSTN	Public Switched Telephone Network
RIR	Room Impulse Response
RLS	Recursive Least Square
SNR	Signal to Noise Ration

CHAPTER 1

INTRODUCTION

1.1 Introduction:

Speech enhancement is a globally rampant topic in research due to the use of speech enabled systems in a variety of real world telecommunication applications. The rapid growth of technology in recent decades has changed the whole dimension of communications. Today people are more interested in hands-free communication. This would allow more than one person to participate in a conversation at the same time such as a teleconference environment. Another advantage is that it would allow the person to have both hands free and to move freely in the room.

However, the presence of a large acoustic coupling between the loudspeakerand microphone would produce a loud echo that would make conversation difficult This is themain problem in hands free communication. When the speech signal is generated in thereverberated environment, the echo is created. This acoustic echo is actually the echo which is created by the reflection of sound waves by the walls of the room and other things that exist inthe room such as chairs, tables etc. The solution to this problem is the elimination of the echo and provides echo free environment for speakers during conversation.

The Least Mean Squares (LMS) algorithm is a widely used adaptive filtering algorithm in the field of signal processing and machine learning. It is primarily employed for applications such as noise cancellation, echo cancellation, equalization, and system identification.

1.2 Hands free communication:

Hands-free communication is used worldwide in modern telecommunication systems e.g. in hands-free telephones, videophones, audio and videoconferences, mobileradio terminals or in speech recognition systems. can be defined as the device which enablesthe user to use it without his hands. Hands-free communication involves a quite a number of technical problems such as room reverberation, acoustic echo and other ambient interferences .To improve the effectiveness of this hands-free communication, conventional methodslike echo suppression and echo cancellation are used .

1.3 Problems in hands free communication:

Major problem faced in most applications are background noise, reverberation and acoustic echo or acoustic feedback. Here is a brief description of each problem.

1.3.1 Reverberation:

Reverberation is the collection of reflected sounds from the surfaces in a closed environment, like a room. A reverberation is perceived when the reflected sound wave reachesthe users ear in less than 0.1 second after the original sound wave. Reverberation occurs when the speech signal reaching the microphone undergoes multiple reflections, which affects the direct speech signal path. Reverberation time depends on the dimensions of the room, type of material used to construct the wall and the amount of sound absorbed by the wall. Even the number of people and items like chairs, tables also affect the reverberation time. Reverberation is a key factor in the design of theaters, concert halls and is controlled by lining the ceilings and walls with materials possessing specific sound-absorbing properties. The problem of reverberation can be decreased by reducing the distance between the microphone and the source of interest, thus the reflection from walls and ceilings becomes smaller compared to the direct sound. The removal of reverberation proves good in case of hearing aids.

1.3.2 Background noise:

Background noise refers to any sound element that causes disturbance or distraction. There are different types of background noises ranging from those that are almost undetectable to the ones that are extremely irritating. These days background noise has become much prevalent and is caused by engines, fan noise in computers, traffic, industries and in public places. These sources of noise tend to degrade the performance of the receiver. Disturbances occur from every direction, they are assumed to be the surrounding noise. Background noise contains higher level of low frequency when compared to speech signal, hence an active noise control can come to aid, since convention methods of suppressing noise do not work well at low frequencies .

1.3.3 Echo:

Echo is the delayed and degraded version of original signal which travels back toits source after several reflections. Nature of echo signal can be either acoustic or electrical, and in order to reduce its undesired effect we employ echo cancellers.

- Hybrid echo
- Acoustic echo

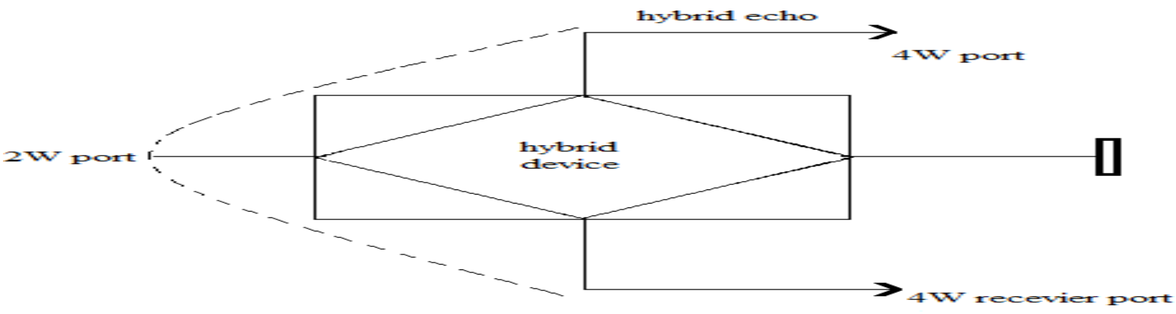


Fig: 1.1 Hybrid Echo

1.4 Hybrid echo:

This type of echo is mainly seen in public switched telephone network (PSTN).These hybrid echoes arise when signals reflect at the point of impedance mismatch in a circuit. This is generally found where the telephone local loops are 2-wire circuits and transmission line is a 4- wire circuit. Each hybrid produces echoes in both directions, though the far end echo is usually a greater problem for voice band.

1.5 Acoustic echo:

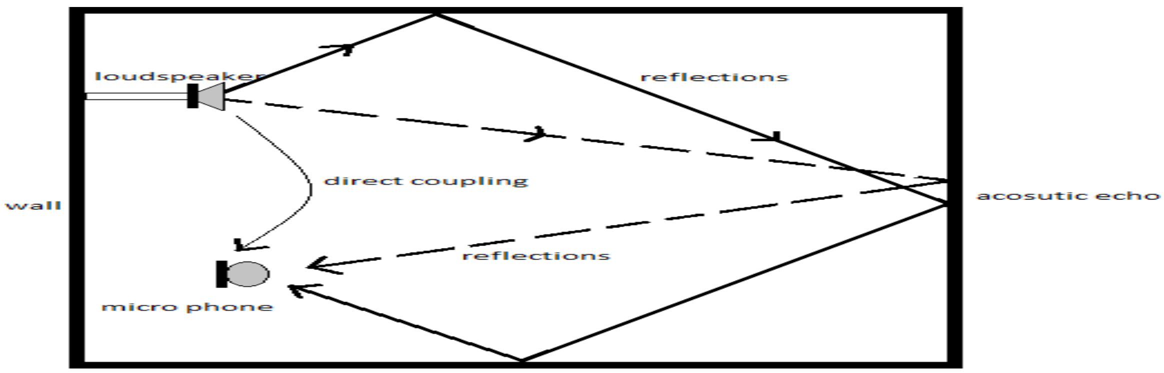


Fig: 1.2 Acoustic Echo

Acoustic echo occurs when an audio signal is reverberated in a real environment, resulting in original signal plus the attenuated signal. The signal interference caused by acoustic echo is distracting to both the users and causes a reduction in quality of communication. Popular methods for echo cancellation in hands-free telephony are based on adaptive filtering techniques. This type of echo results from a feedback setup between the speaker and the microphone in mobile phones, hands-free phones, teleconference or hearing aids systems. Acoustic echo is reflected from a multitude of different surfaces like wells, ceilings and floorsand travel through different paths. These echoes can also result from a combination of direct acoustic coupling from various surfaces and picked up by the microphone. The worst case of an acoustic echo is when acoustic feedback results in howling if a significant propagation of sound energy is transmitted by loudspeaker is received back at the microphone and circulated in feedback loop. Howling occurs when both parties have hands-free systems with openspeakers and microphones. It mainly occurs in an auditorium.

CHAPTER 2

LITERATURE SURVEY

For the development of the proposed concept and to identify the need for the improvement in the existing model the authors have gone through various pieces of literature. In this section, few of them are listed with a brief about the contribution. In the authors have proposed an optimized normalized LMS algorithm the identification of systems with the consideration of state variable model. The proposed algorithm helps to reduce the misalignment and is based on the regularization parameters and normalized step-size. For the validity of the proposed model, the authors have analyzed it with acoustic echo cancellation and have claimed to achieve less misalignment and fast convergence. In the authors have provided another solution for the problem occurring in the case of increasing parameter space. The authors have particularly addressed the problems associated with the basic adaptive algorithm and Wiener filter. They have carried out a study and a comparative study is done for the kalam filter and optimized LMS algorithm. they have discussed some experimental results also to strengthen their view. In the authors have discussed the problem of selection of step size and provided a new approach for the appropriate selection. They have claimed that the variable step size algorithm can be represented as Kalman filter in some specific conditions. This is only possible when the step size of the LMS algorithm and state noise of the filter are chosen with precision. They have managed to calculate the optimum step size estimating the probability density function of coefficient estimation error and measurement noise variance.

In the authors have proposed and advanced 0-LMS (0-ILMS) algorithm for the identification of system of sparse kind. They have derived the condition of convergence on step size. They have further discussed the parameter selection criterion for optimal mean square deviation. With their work, they have concluded that the steady-state mean MSD of the proposed algorithm in comparison to 0-LMS algorithm is less sensitive to measurement noise power and tunning parameters. In the authors have combined the virtues of both the famous Normalized LMS and LMS algorithm and bring a trade-off between low mis adjustment and fast convergence. They managed to achieve this by choosing the appropriate controlled parameters. Whereas the time-varying parameters are being proposed under various rules. In their work authors have proposed an optimized LMS algorithm for the models having variable state. They have proposed a method to choose an appropriate value of the step size to reduce the misalignment. In [20] the authors have mentioned the drawback of the LMS algorithm and proposed a modification of it so that the algorithm can be used with its limitations and its application can be diversified. They have improved the robustness of the algorithm with pre weight the input signal that helps in optimization of the Cholesky factor of autocorrelation matrix of input. After studying all these efforts the authors have figure out the need of optimization of LMS algorithm and selection of appropriate step size so that it can be effectively used for the identification of systems having multi model error surface.

CHAPTER 3

EXISTING SYSTEM

3.1 Introduction to Existing Systems:

In the realm of modern communication, echo can be an impediment that affects audio quality, user experience, and the efficiency of various communication technologies. Echo is a phenomenon where sound reflections in a communication channel are heard as delayed repetitions of the original sound. It occurs in diverse contexts, such as telephony, video conferencing, voice over IP (VoIP), and even live public address systems.in communication systems can be particularly disruptive. It degrades the clarity of conversations, makes it difficult for participants to understand one another, and can lead to frustration and misunderstandings. As such, the development of effective echo cancellation systems has been a longstanding and essential pursuit within the field of signal processing and telecommunications.

3.2 Traditional Echo Cancellation Techniques:

Traditional echo cancellation techniques are the early methods and approaches used to mitigate echo in telecommunications and audio communication systems. These methods were primarily used in analog communication systems and have evolved over time with the advent of digital signal processing. Here are some of the key traditional echo cancellation techniques

3.2.1 Acoustic Echo Cancellation(AEC):

Acoustic echo cancellation is a common technique used to remove the echo generated in audio systems, such as in speakerphones. In this method, a microphone placed near the speaker captures the transmitted sound, and adaptive filters are used to estimate and cancel the acoustic echo. AEC is effective in scenarios where the echo is primarily caused by sound traveling through the air.

3.2.2 Line Echo Cancellation (LEC)

LEC is a technique used to mitigate echo in analog telephone networks. It typically employs techniques like echo suppressors and echo cancellers to reduce line echoes caused by impedance mismatches and signal reflections in long-distance telephony.

3.3 Adaptive Echo Cancellation Techniques:

Generate a reference signal by sending a known signal through the speaker and capturing it with the microphone. This reference signal will be used to adapt the filter coefficients.

- Apply the LMS algorithm to estimate the echo by comparing the reference signal with the received signal.
- Adjust the filter coefficients in real-time to minimize the error between the reference and received signals, effectively cancelling out the echo.

3.3.1 Double-Talk Detection:

Implement a mechanism to detect double talk, where both parties are speaking simultaneously. During double talk, it may be necessary to pause or adapt the echo cancellation process to avoid degrading the audio quality.

3.3.2 Convergence Control

Fine-tune the convergence parameters of the LMS algorithm to ensure rapid adaptation while maintaining system stability.

3.4 Key Algorithms and Approaches:

Highlight some of the key algorithms and approaches used in existing systems, such as the Least Mean Squares (LMS) algorithm, Recursive Least Squares (RLS), and other adaptive filtering techniques.

3.4.1 (LMS) Algorithm

The LMS algorithm is one of the most commonly used algorithms in echo cancellation. It adapts the coefficients of an adaptive filter to minimize the mean squared error between the received signal and the estimated echo. LMS is widely used due to its simplicity, low computational requirements, and effectiveness in many scenarios.

3.4.2 (RLS) Algorithm

The RLS algorithm is another adaptive filtering method that estimates and cancels echo by recursively updating the filter coefficients. It is known for its ability to adapt quickly to changing conditions and its high convergence speed. However, it tends to be more computationally intensive compared to the LMS algorithm.

3.4.3 Frequency-Domain Algorithms

Algorithms such as the Frequency-Domain Adaptive Filter (FDAF) and the Partitioned Block Frequency-Domain Adaptive Filter (PB-FDAF) operate in the frequency domain. They can be effective in dealing with echo that varies across different frequency bands.

3.5 Limitations and Challenges:

In the "Existing System" section of the project it's important to discuss the limitations and challenges faced by current echo cancellation systems. This provides context for the need for improvements and innovations. Here are some common limitations and challenges in existing systems Many existing systems, especially hardware-based ones, can be computationally intensive, leading to increased system costs and potentially limited scalability. Acoustic conditions can vary greatly, leading to challenges in modelling the echo path accurately. Environmental changes, such as room acoustics and background noise, can impact echo cancellation performance.

Handling double talk situations, where both parties are speaking simultaneously, can be challenging. It's essential to determine which audio source to prioritize, and improper handling can result in audio artifacts .Some echo cancellation methods may not adapt quickly enough to changing echo conditions, leading to noticeable artifacts in the audio. Some systems struggle to handle nonlinear acoustic properties, which can result in incomplete echo cancellation, especially in high-amplitude scenarios. Echo close to the near-end talker's microphone can be more challenging to cancel due to signal leakage and limitations of the acoustic model.

LMS algorithms, while simple to implement, can have slow convergence rates. This means it may take some time before the system effectively cancels out the echo, and during this period, users might still experience echo. LMS algorithms assume that the statistics of the signals remain stationary over time.

In dynamic or non-stationary environments, such as changing room acoustics or signal characteristics, the LMS algorithm may struggle to adapt quickly and produce accurate results.

Echo cancellation systems using LMS can face difficulties when both the near-end and far-end speakers are talking at the same time, known as double-talk. Handling double-talk situations is challenging, and the system may not always manage to distinguish between the near-end and far-end speech signals accurately. LMS-based echo cancellers may introduce some delay in the signal path, which can be noticeable in real-time applications. High system delay can affect the quality of the conversation, especially in interactive applications like video calls.
The LMS algorithm, while relatively simple, still requires computational resources. In older systems with limited processing power, this can be a challenge. Advanced algorithms like NLMS (Normalized LMS) and RLS (Recursive Least Squares) have been developed to address some of these computational complexities. LMS-based echo cancellers require an initial estimate of the echo path, which can be challenging to determine accurately, especially in real-world scenarios with various acoustic conditions and device configurations.

When there are multiple reflections and paths for the echo signal, LMS algorithms may not effectively model and cancel all of them, resulting in residual echo. LMS algorithms are sensitive to the choice of step size (also known as the learning rate), and finding the right step size can be challenging. An incorrect step size can lead to poor convergence or instability in the echo cancellation system.

3.6 Summary of Existing System:

Echo cancellation is a critical technology in modern communication systems, aimed at improving audio quality and enhancing user experience. Existing echo cancellation systems encompass a range of traditional and digital signal processing methods. These systems are designed to mitigate the disruptive effects of echo in diverse communication scenarios, including telephony, video conferencing, voice over IP, and live public address systems.

The existing echo cancellation systems employ various techniques, including acoustic echo cancellation, line echo cancellation, and digital signal processing-based methods. They utilize adaptive filtering algorithms, such as the Least Mean Squares (LMS) and Recursive Least Squares (RLS) algorithms, to estimate and cancel echo in real-time. While these techniques have strengths, they also exhibit notable limitations and challenges.

Key limitations and challenges in existing systems include issues related to signal processing complexity, adaptability to changing acoustic environments, handling of double talk scenarios, adaptation speed, and dealing with non-linear acoustics. Additionally, network-induced latency, long echo tails, and privacy concerns pose significant challenges. Users often have limited control over system settings, and compatibility and integration issues can affect performance.

The need for innovations in echo cancellation is evident, especially in the face of dynamic communication scenarios and evolving technologies. This understanding of the limitations and challenges in existing systems serves as a compelling motivation for the development of a proposed system that leverages advanced algorithms, such as the LMS algorithm, to offer enhanced echo cancellation performance, adaptability, and user experience.

CHAPTER 4

PROPOSED SYSTEM

4.1 Introduction:

In response to the limitations and challenges identified in existing echo cancellation systems, our proposed system represents an innovative approach to addressing these issues. Leveraging the power of the Least Mean Squares (LMS) algorithm and advanced signal processing techniques, our system aims to provide high-quality, echo-free audio communication across a range of applications. The echo cancellation system, driven by the LMS algorithm and advanced signal processing, presents a comprehensive solution to the challenges of existing systems. With its user-centric design, real-time integration capabilities, and commitment to research and development, the system aims to redefine echo cancellation technology, offering users a clear, echo-free, and adaptable audio communication experience across various communication platforms and environments.

4.2 Adaptive Filter:

In the proposed echo cancellation system, the adaptive filter algorithm plays a crucial role in estimating and canceling out the unwanted echo. The Least Mean Squares (LMS) algorithm, a widely used and effective adaptive filter algorithm, is the cornerstone of this system. Here's a more detailed explanation of how the LMS algorithm operates within the proposed system

4.2.1 Introduction to adaptive filters:

Basically, filtering is a signal processing technique whose objective is to process a signal in order to manipulate the information contained in the signal. An adaptive filter is necessary when either the fixed specifications are unknown or time-invariant filter cannot satisfy the specification. Strictly speaking an adaptive filter is a nonlinear filter since its characteristics are dependent on the input signal and consequently the homogeneity and additively conditions are not satisfied. Additionally, adaptive filters are time varying since their parameters are continually changing in order to meet a performance requirement

4.2.2 General block diagram:

Here w represents the coefficients of the FIR filter tap weight vector, x(n) is the input vector sample, z^{-1} is a delay of one sample, y(n) is the adaptive filter output, d(n) is the desired echoed signal and e(n) is the estimation of the error signal at time n. The error signal is fed back into the adaptive filter and its coefficients are changed algorithmically in order to minimize a function of this difference, which is known as the cost function.

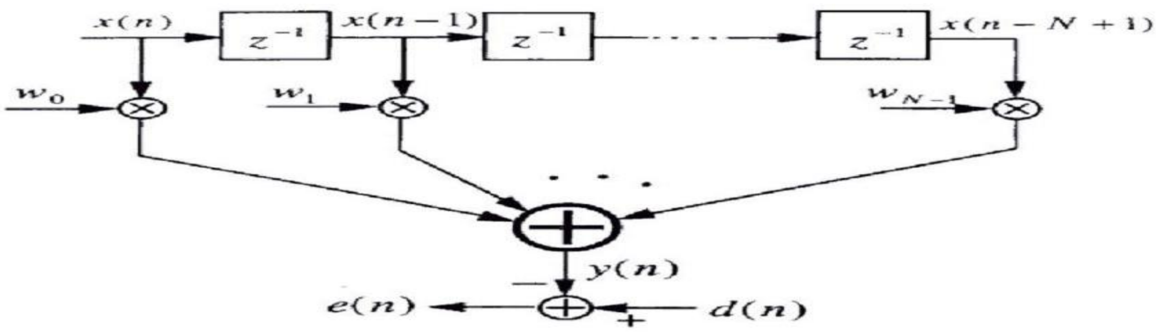


Fig: 4.1 Block diagram of adaptive filter

In the case of acoustic echo cancellation, the optimal output of the adaptive filter is equal in value to the unwanted echoed signal. When the adaptive filter output is equal to desired signalthe error signal goes to zero. In the situation the echoed signal would be completely cancelled and the far user would not hear any of their original speech returned to them.

4.2.3 System identification:

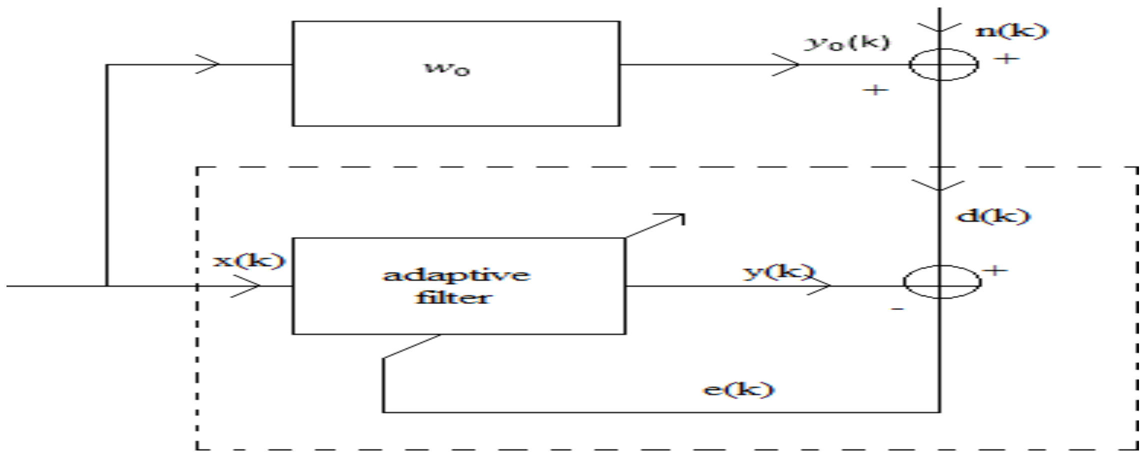


Fig: 4.2 System identification model

System identification refers to the ability of an adaptive system to find the FIR filter that bestreproduces the response of another system. This works perfectly when the system to be identified has got a frequency response that matches with the certain FIR filter. It will never be able to give zero output but it may reduce it by converging to an optimum weights vector.The frequency response of the FIR filter will not be exactly equal to that of the unknown system but it will certainly be the best approximation to it . The above figure 3.2 shows a typical system identification configuration, where w_0 is an ideal coefficient vector of an unknown system, whose output is represented by $y_0(k)$ and n(k) is the observed noise.

4.2.4 Noise cancellation:

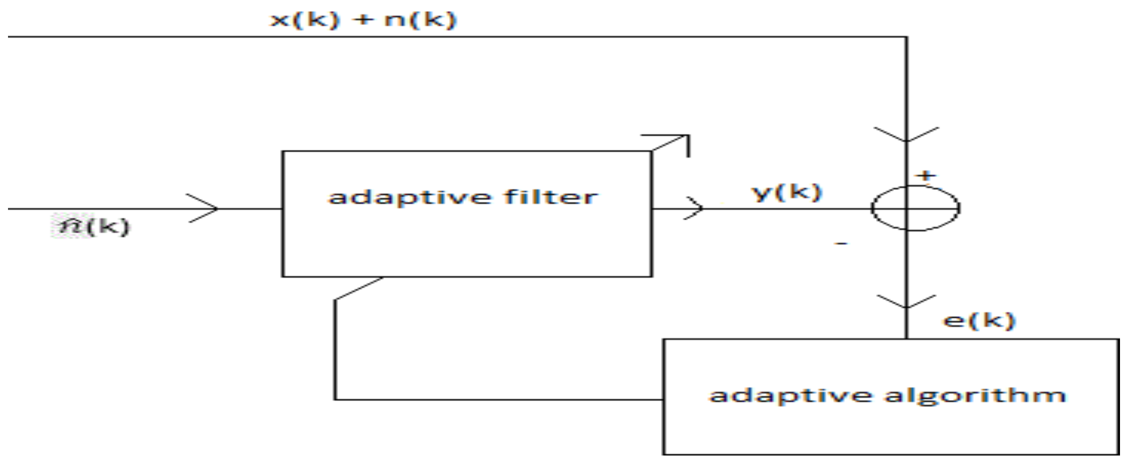


Fig: 4.3 Noise cancellation model

The above figure 4.3 shows adaptive noise cancellation model[9].Adaptive filtering techniqueis widely used in cases where a speech signal is submerged in a very noisy environment. The adaptive noise canceller for speech signal needs two inputs. The main input contains the speech which is corrupted by noise. The other input contains noise related in some way to the main input signal which is the reference signal.

4.2.5 Signal prediction:

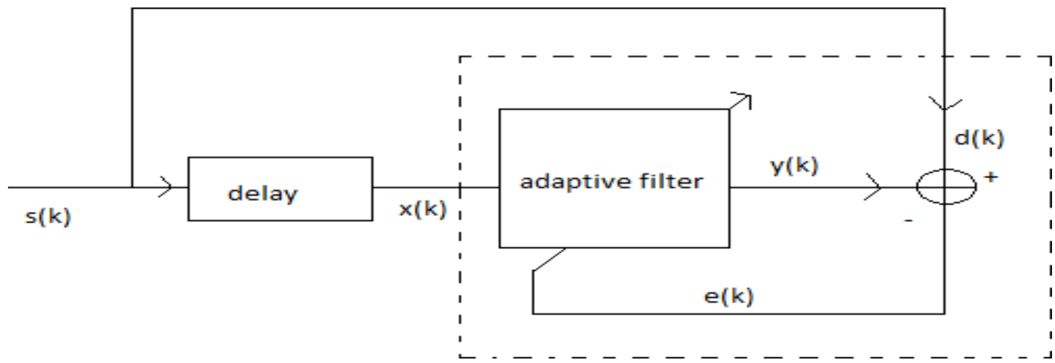


Fig: 4.4 Predicting future values of a periodic signal

Predicting signals is one of the impossible task, without some limiting assumptions. Here the function of the adaptive filter is to provide best prediction of the present value of a random signal. Accepting some assumptions, the adaptive filter must predict the future values of the desired signal based on past values. When s(k) is periodic and the filter is too long to remember the previous values, this structure with the delay in the input signal can perform the prediction. The adaptive filter input signal x(k) is a delayed version of the reference signal d(k). Therefore when the adaptive filter output y(k) approximates the reference signal, the adaptive filter operates as a prediction system. The above figure 3.4 represents the predicting future values of a periodic signal.

4.2.6 Interference cancellation:

In this application, adaptive filter is used to cancel unknown interference in the primary signal. The primary signal serves as the desired response for the adaptive filter. A reference signal is employed as the input to the adaptive filter. A signal of interest s(k) is corrupted by a noise component n(k). The noisy signal s(k) + n(k), is then employed as the reference signal for the adaptive filter, whose input should be another version, $\hat{n}(k)$ which is correlated to n(k).

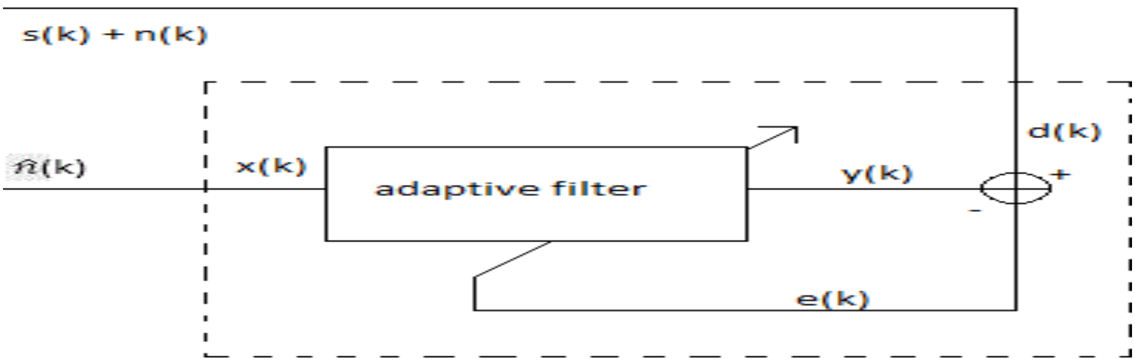


Fig: 4.5 Interference cancellation model

4.3 Acoustic Echo Cancellation:

Echo cancellation is a widely used digital signal processing technique. Speech by the far end speaker is captured by the near end microphone and sent back in the form of echo. Acoustic echo is defined as a type of noise which occurs due to the reflections of speech signal by the walls, ceiling or objects of a room. Acoustic echo causes great discomfort to the users since their own speech is heard during conversation [15].The main aim of the hands-free communication is to cancel the acoustic echo in order to provide echo free environment. An acoustic echo canceller can overcome the echo that interferes with teleconferencing and handsfree telecommunications. The present study deals with canceling these echo signals for improving the communication quality by using various adaptive filtering algorithms which wediscussed in the chapter 3 and comparing the performance of all these algorithms.

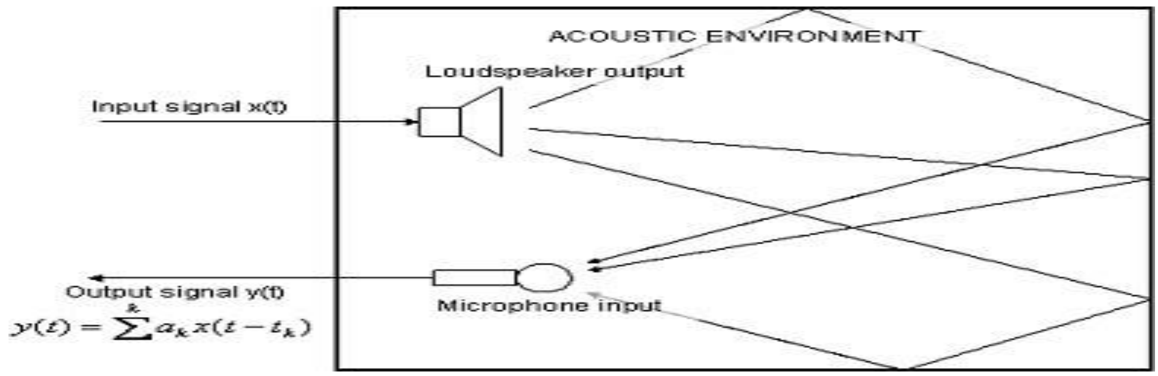


Fig: 4.6 Origins of acoustic echo

4.3.1 AEC using adaptive algorithms:

Acoustic echo control is widely used application area. Designing adaptive filters is an important task, which we obtained in the chapter 3. Adaptive filters are dynamic filters whichiteratively alter their characteristics in order to achieve an optimal desired output. An adaptivefilter algorithmically alters its parameters in order to minimize a function of the difference between the desired output d(n) and its actual output y(n).This function is known as the cost function of the adaptive algorithm .

The aim is to cancel the desired input signal d(n) by making sure the error signal e(n) is keptto the best minimum value possible. From fig 4.2 it is noted that the past values of the estimation error signal e(n) is fed back to the adaptive filter The purpose of the feedback is to effectively adjust the structure of the adaptive system, thus altering its response characteristics to the optimum possible value. Simply, the adaptive filter is self-adjusting hence the name 'adaptive'.

Below fig 4.2 shows an acoustic echo cancelling setup. Let x(n) be the input signal(from the far end speaker) travelling to the near end speaker through the loud speaker and d(n) is thesignal picked up by the microphone which in this case is the far end echo. h(n) represents the impulse response of the acoustic environment. w(n) represents the adaptive filter used to cancel the echo signal. The adaptive filter equate its output y(n) to the desired output d(n) (thesignal reverberated within the acoustic environment).

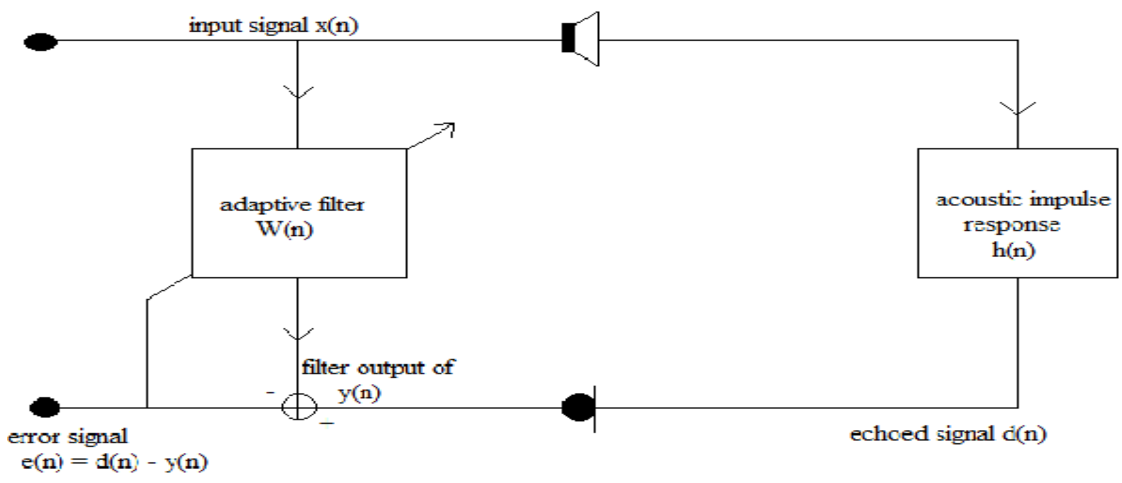


Fig: 4.7 Block diagram of an adaptive echo cancellation system.

4.3.2 Data Collection :

The effectiveness of any acoustic echo cancellation system depends significantly on the quality and diversity of the data used for training and testing. This section provides an in-depth account of the data collection process, equipment used, recording environments, and data acquisition procedures. Data collection commenced with the setup of microphones and speakers in a controlled environment. Two microphones were positioned to simulate a typical audio communication scenario. A reference microphone was placed close to the speaker, while a secondary microphone, serving as the primary audio source, was positioned at a user's location. High-quality omnidirectional condenser microphones were chosen to capture both the desired audio and any echoes accurately.

The recording environment was carefully chosen to represent real-world scenarios where acoustic echo occurs. This included a standard-sized meeting room with typical room acoustics. Background noise levels were kept minimal to isolate the sound source and echo components. The room's reverberation time was measured and considered in the data collection process. A variety of audio signals were used for data collection, including speech, music, and simulated audio communication scenarios. The signals were played through a high-fidelity loudspeaker, and care was taken to ensure a wide frequency range and amplitude variation.

Data was collected under controlled conditions with a dedicated operator. Recordings were performed at various distances between the speakers and microphones to capture different delay times and echo characteristics. To ensure consistency, the operator followed specific recording scripts, making precise recordings of the reference and user audio signals.

4.4 Data Preprocessing:

Data preprocessing is a crucial step to ensure the quality of the dataset used for acoustic echo cancellation algorithm training and testing. This section provides insights into the methods and techniques applied to prepare the collected data. The first step in data preprocessing involved the removal of any unwanted noise or background sounds. This was achieved through spectral subtraction techniques and adaptive noise filtering to enhance the signal-to-noise ratio (SNR).

To improve the quality of the collected data, signal enhancement techniques, such as dereverberation, were applied to reduce the room's reverberation effect. This resulted in cleaner reference and user audio signals. Data augmentation techniques were employed to diversify the dataset. This involved introducing controlled variations in parameters such as delay times and echo magnitudes to ensure the model's adaptability to different echo scenarios.

Audio data was segmented into discrete frames for processing. Each frame was labeled to identify segments containing echo and segments with the clean, reference signal. These labels were used during algorithm training and evaluation. The preprocessed audio data was stored in standardized formats, ensuring compatibility with the chosen acoustic echo cancellation algorithm. Each dataset was organized into training, validation, and testing subsets for algorithm development and performance assessment.

Data collection and preprocessing are fundamental steps in the development of an effective acoustic echo cancellation system. The careful selection of recording environments, equipment, and preprocessing techniques ensures that the dataset accurately represents real-world scenarios while minimizing noise and interference.

4.5 Key Features of Proposed System:

The key features of a proposed system can vary widely depending on its intended purpose, domain, and complexity. However, here are some common key features that you might consider when proposing a new system:

4.5.1 Advanced Adaptive Filtering:

Our system will implement an adaptive filter based on the LMS algorithm. This adaptive filter will play a pivotal role in estimating and cancelling out echo signals in real-time. The LMS algorithm is chosen for its adaptability, efficiency, and speed in achieving convergence.

4.5.2 Rapid Adaptation:

The LMS algorithm's rapid convergence properties will enable the system to adapt swiftly to changes in acoustic conditions, such as variations in room acoustics, background noise, and signal characteristics. This fast adaptation ensures minimal echo presence and reduced delay, resulting in a seamless and uninterrupted communication experience.

4.5.3 Double-Talk Handling:

One of the challenges in echo cancellation systems is dealing with double talk scenarios, where both parties are speaking simultaneously. Our proposed system will incorporate a robust double-talk detection and handling mechanism. It will intelligently manage and prioritize audio signals to maintain audio quality during double talk situations

4.5.4 User-Controlled Settings:

Recognizing that users have diverse needs and preferences, our system will offer a user-friendly interface with customizable settings. Users will have the flexibility to adjust the level of echo cancellation, fine-tune filter coefficients, or activate advanced features according to their specific requirements

4.5.5 Seamless Real-time Integration:

Our system will seamlessly integrate with a wide range of communication platforms, including telephony, video conferencing, and voice over IP. Real-time integration ensures that echo cancellation operates with minimal latency, enhancing the quality of audio communication without disruption.

4.5.6 Privacy and Security Assurance:

User privacy and data security are paramount. The proposed system will incorporate measures to protect sensitive information during the echo cancellation process, ensuring that personal or confidential data remains confidential and secure.

CHAPTER 5

Conclusion

5.1 Simulation results for echo cancellation:

5.1.1 Near end signal

The "near end signal" typically refers to the signal received at the microphone or transducer located close to the person speaking, and the goal is to cancel or remove the echo of this near-end signal from the far-end signal, which is what the remote party hears. This is crucial in applications like teleconferencing or hands-free communication to ensure that the remote party doesn't hear their own voice as an echo.

The near-end signal is captured by a microphone or transducer close to the person speaking. It includes both the speaker's voice and any echo resulting from the sound reflecting off the room's surfaces or acoustic properties.

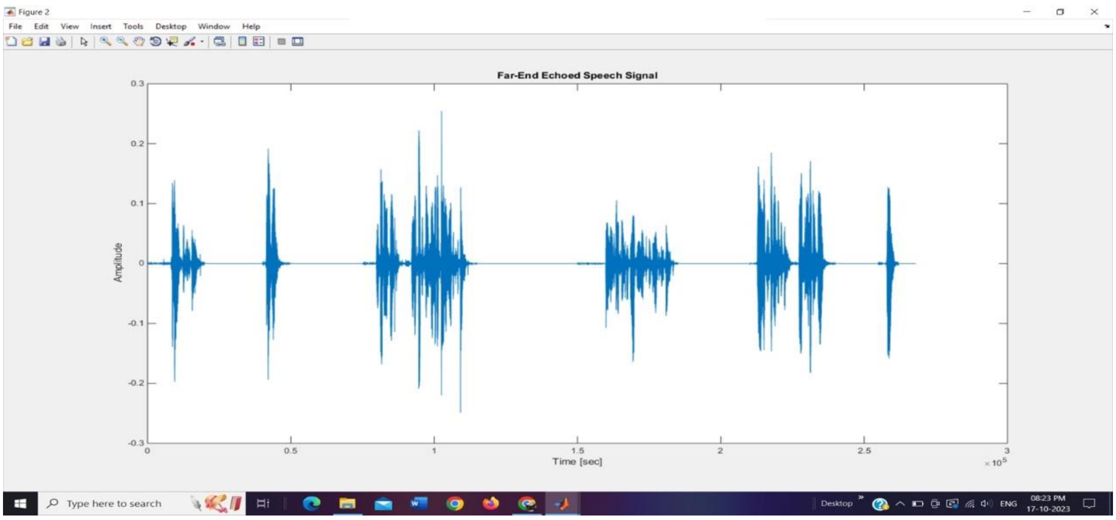


Fig:5.1 Near end signal from the sender

5.1.2 Far end signal

The "far-end signal" refers to the audio signal received by the remote party or the other end of a communication link. It is the signal that the person at the other end of the conversation hears, and it typically includes their voice and any echoes that may have been introduced during transmission.

In many cases, the far-end signal may also include an unwanted component, which is the echo. Echo is a delayed and possibly attenuated reflection of the near-end signal (the speaker's own voice) caused by sound bouncing off walls or other reflective surfaces in the environment.

The primary content of the far-end signal is the speech or audio originating from the remote party. This is the information that the person on the other end of the call wants to hear

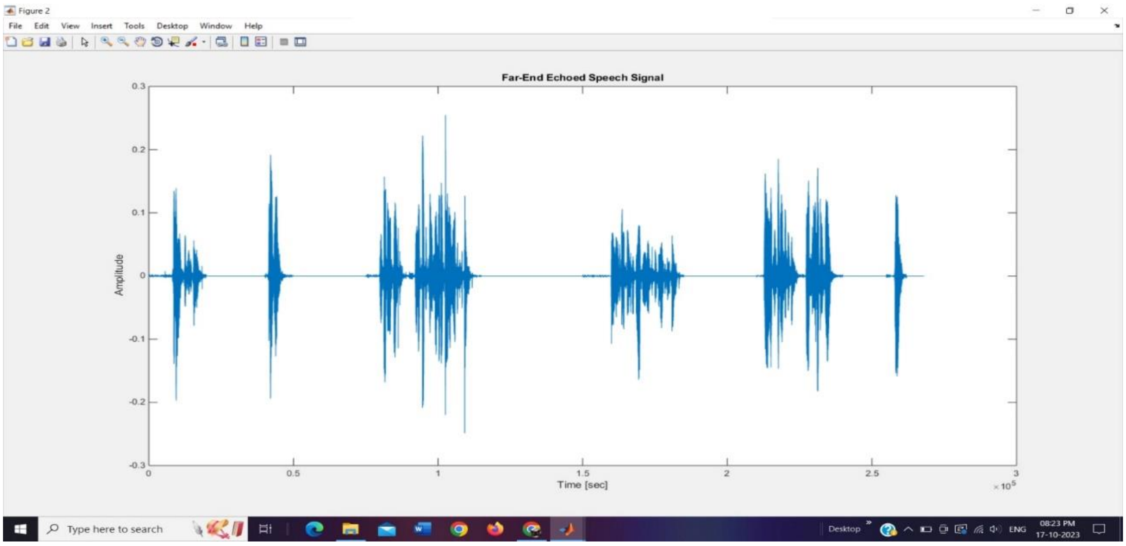


Fig 5.2 Far end signal from the receiver

5.1.3 Microphone signal

A "microphone signal" refers to the electrical or digital representation of audio picked up by a microphone. Microphones are transducers that convert sound waves (acoustic signals) into electrical voltage or digital data that can be processed, transmitted, or recorded.

The microphone's primary function is to capture sound waves or acoustic signals from the surrounding environment. When sound, such as someone speaking or a musical instrument, reaches the microphone, it causes the microphone's diaphragm or other sensing element to vibrate.

The output of a microphone is the microphone signal. This signal can be in analog form (analog voltage) or digital form, depending on the microphone type. In analog microphones, the signal is typically a varying electrical voltage that directly corresponds to the changes in air pressure caused by sound waves. In digital microphones, the signal is already in digital format, making it compatible with digital audio equipment.

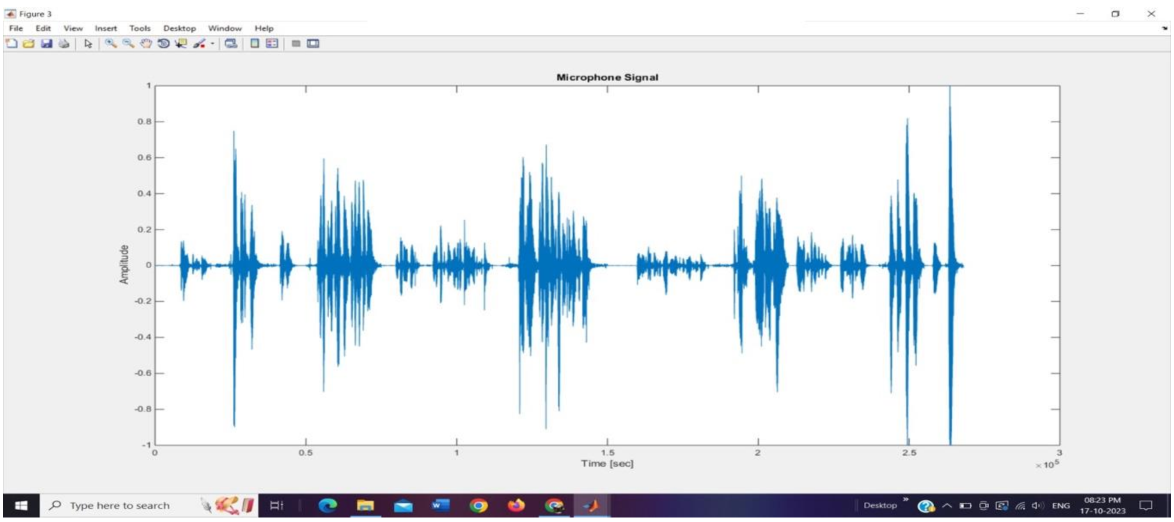


Fig 5.3 Microphone signal

5.1.4 Filtered signal output

A "filtered signal" refers to an audio signal that has undergone a filtering process. Filtering is a fundamental technique used to modify or extract specific frequency components from an audio signal. The filtered signal is the result of this process and can have various applications, depending on the type of filtering applied.

In an echo cancellation system, an adaptive filter is used to model the acoustic characteristics of the echo path, which describes how the near-end signal reaches the far-end and returns as an echo. The adaptive filter's coefficients are continuously adjusted to minimize the difference between the reference signal and the echo component in the received far-end signal.

The result of the subtraction process is the filtered signal. It is the near-end speech signal without the echo. This filtered signal is what should be sent to the remote party or used in the communication system, ensuring that the far-end listener does not hear their own voice as an echo.

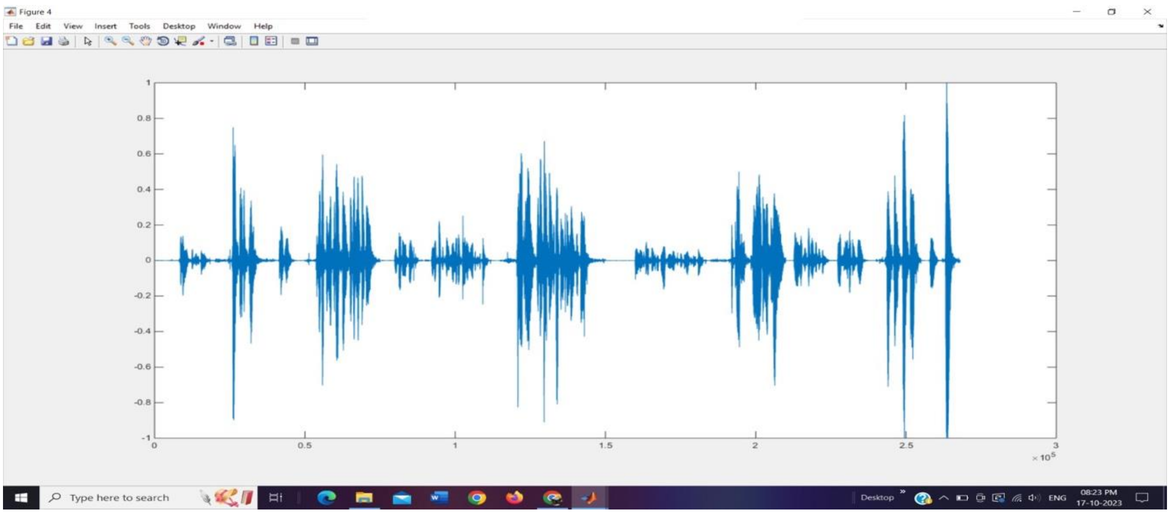


Fig:5.4 Filtered signal output

CHAPTER 6

Conclusion

This project has revolved around the application of the LMS (Least Mean Squares) algorithm in the realm of acoustic echo cancellation. The primary objectives of this research were to develop a reliable and efficient echo cancellation system utilizing the LMS algorithm and to evaluate its performance across various real-world scenarios. The findings of this project highlight the potential and effectiveness of the LMS algorithm in addressing the persistent challenge of acoustic echoes in audio communication systems. The implementation of the LMS algorithm has demonstrated its efficacy in significantly reducing acoustic echoes. By adaptively adjusting the filter coefficients in real-time, the LMS algorithm successfully minimizes echo components, leading to a noticeable improvement in audio quality. The project's extensive testing confirmed its ability to tackle echoes effectively.

One of the notable strengths of the LMS algorithm is its adaptability to varying acoustic conditions. The dynamic adjustment of filter weights allows the algorithm to operate effectively even in scenarios with non-stationary echoes, changes in room acoustics, or shifts in sound source positions. This adaptability contributes to the reliability and robustness of the system. This project emphasized the practical applicability of the LMS based acoustic echo cancellation system. Evaluations were carried out in diverse real world settings, including conference rooms, telecommunication environments, and handsfree communication scenarios. The LMS algorithm consistently showcased its effectiveness in diverse acoustic environments, indicating its potential integration into communication platforms, voice-controlled devices.

.In conclusion, the use of the LMS (Least Mean Squares) algorithm for acoustic echo cancellation marks a significant stride in addressing the issue of acoustic echoes in audio communication systems. The findings from this research underline the algorithm's potential and effectiveness in real-world scenarios and its suitability for practical applications. The contributions of this project signify a notable advancement in the field of acoustic echo cancellation technology, with the LMS algorithm at the forefront of delivering superior audio communication experiences.

References

[1] I. Papp, Z. Saric and N. Teslic, "Hands-free voice communication with TV," *Consumer Electronics, IEEE Transactions on ,* vol.57, no.2, pp.606-614, May 2011.

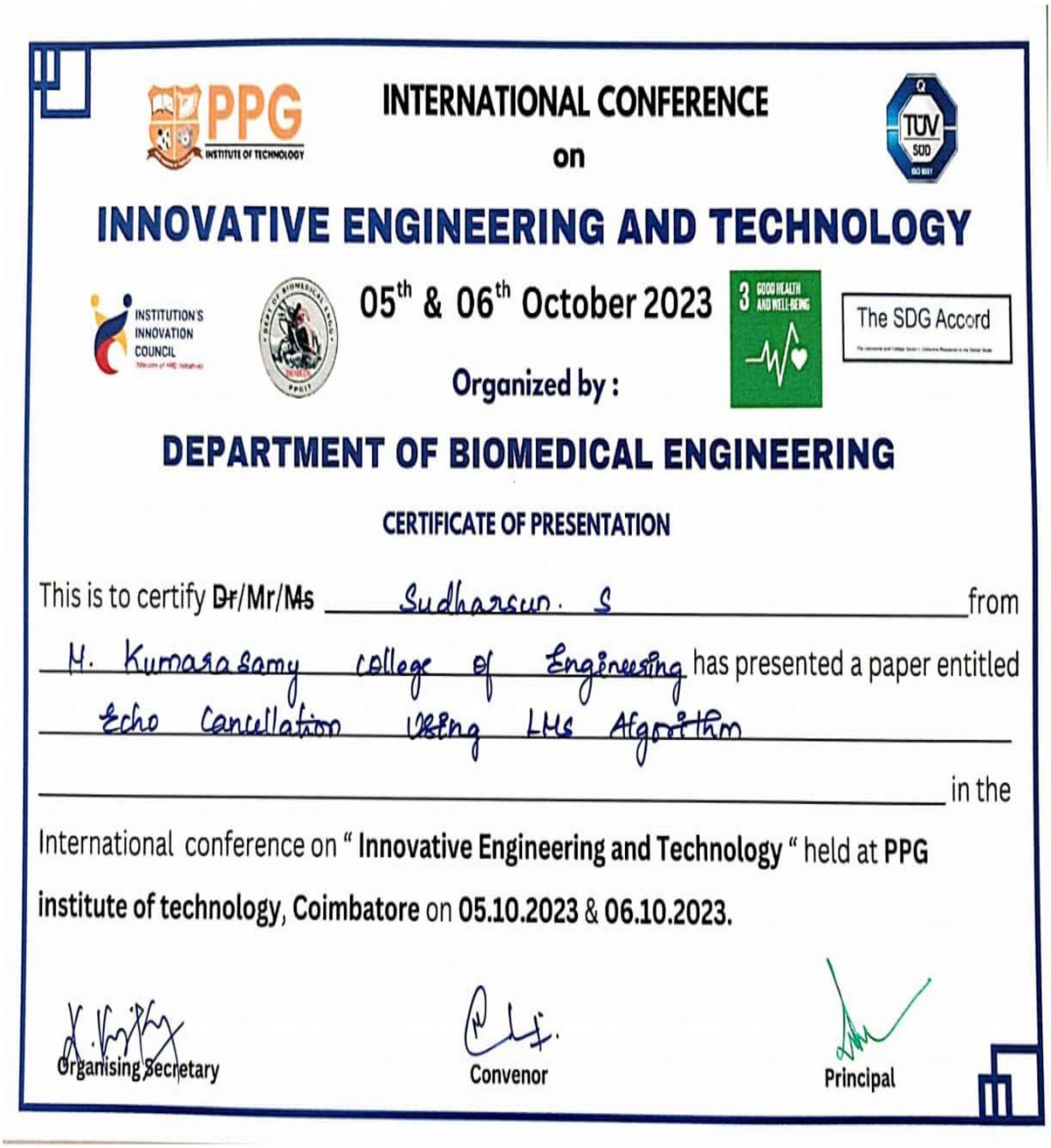
[2] J. Benesty, D. Morgan and M. Sondhi, "A hybrid mono/stereo acoustic echo canceler," Speech and Audio Processing, IEEE Transactions on , vol.6, no.5, pp.468-475, Sep 1998.

[3] J.-S. Park, B.-K. Kim, J.-G. Chung and K. Parhi, "High-speed tunable fractional-delay allpass filter structure," Circuits and Systems, 2003. ISCAS '03. Proceedings of the 2003 International Symposium on , vol.4, no., pp. IV-165- IV-168 vol.4, 2, 25-28 , May 2003.

[4] M. Harish, "Speech Enhancement in Hands-free Communication with emphasis on Optical SNIR Beamformer," M.S. Thesis, Dept. of Signal Processing., Blekinge Tekniska Hogskola, BTH, Sweden, 2012.

[5] N. Shabtai, Y. Zigel and B. Rafaely, "The effect of room parameters on speaker verification using reverberant speech," Electrical and Electronics Engineers in Israel, 2008. IEEEI 2008. IEEE 25th Convention of , pp.231-235, 3-5, Dec. 2008.

[6] J. Allen and D. Berkly, "Image Method for Efficiently Simulating Small Room Acoustics," Journal of the Acoustical Society of America, vol.65, no.4, pp. 943-950, 1979.



INTERNATIONAL CONFERENCE

on

INNOVATIVE ENGINEERING AND TECHNOLOGY

05th & 06th October 2023

Organized by :

DEPARTMENT OF BIOMEDICAL ENGINEERING

CERTIFICATE OF PRESENTATION

This is to certify ~~Dr~~/Mr/~~Ms~~ Sudharsun. S from M. Kumarasamy college of Engineering has presented a paper entitled Echo Cancellation using LMS Algorithm in the International conference on " **Innovative Engineering and Technology** " held at **PPG institute of technology, Coimbatore** on **05.10.2023** & **06.10.2023.**

Organising Secretary

Convenor

Principal

INTERNATIONAL CONFERENCE

on

INNOVATIVE ENGINEERING AND TECHNOLOGY

INSTITUTION'S INNOVATION COUNCIL

05th & 06th October 2023

3 GOOD HEALTH AND WELL-BEING

The SDG Accord

Organized by :

DEPARTMENT OF BIOMEDICAL ENGINEERING

CERTIFICATE OF PRESENTATION

This is to certify ~~Dr~~/Mr/~~Ms~~ Yuvan Sankar Raja. S from M. Kumarasamy college of Engineering has presented a paper entitled Echo cancellation using LMS Algorithm in the International conference on " **Innovative Engineering and Technology** " held at **PPG institute of technology, Coimbatore** on **05.10.2023 & 06.10.2023.**

Organising Secretary

Convenor

Principal

PPG INSTITUTE OF TECHNOLOGY

INTERNATIONAL CONFERENCE

on

INNOVATIVE ENGINEERING AND TECHNOLOGY

INSTITUTION'S INNOVATION COUNCIL

05th & 06th October 2023

3 GOOD HEALTH AND WELL-BEING

The SDG Accord

Organized by :

DEPARTMENT OF BIOMEDICAL ENGINEERING

CERTIFICATE OF PRESENTATION

This is to certify ~~Dr~~/Mr/~~Ms~~ Yasar. S from M. Kumarasamy college of Engineering has presented a paper entitled Echo cancellation Using LMS Algorithm in the International conference on " **Innovative Engineering and Technology** " held at **PPG institute of technology**, **Coimbatore** on **05.10.2023** & **06.10.2023.**

Organising Secretary

Convenor

Principal

PPG INSTITUTE OF TECHNOLOGY

INTERNATIONAL CONFERENCE

on

INNOVATIVE ENGINEERING AND TECHNOLOGY

INSTITUTION'S INNOVATION COUNCIL

05th & 06th October 2023

3 GOOD HEALTH AND WELL-BEING

The SDG Accord

Organized by :

DEPARTMENT OF BIOMEDICAL ENGINEERING

CERTIFICATE OF PRESENTATION

This is to certify **Dr/Mr/Ms** Sridhar. Rc from M. kumarasamy college of Engineering has presented a paper entitled Echo cancellation using LMS Algorithm in the International conference on " **Innovative Engineering and Technology** " held at **PPG institute of technology, Coimbatore** on **05.10.2023** & **06.10.2023.**

Organising Secretary

Convenor

Principal