Salwa Fayyad 1200430 , Katya Kobari 1201478 ,Sondos Farrah 1200905

Adaptive filter algorithms for system identification

Abstract — The project focuses on solving the problem of echo in phone line communication. Echo occurs when a signal reflects back from the near-end to the far-end, degrading the communication quality. The project aims to develop a system that cancels out the echo using an adaptive filter. By analyzing the incoming signal and the corresponding echo, the system learns to remove the unwanted echo and improve communication clarity. The performance of the system is evaluated by comparing the input signal, echo signal, and the error signal. The goal is to enhance the quality of phone line communication by effectively eliminating echo.

I. INTRODUCTION

Objective: The objective is to explain the occurrence of line echo in phone line communication, where signals reflecting at the near-end result in an echo received by the far-end speaker. The objective is to highlight the need for a line echo cancellation system to address this issue and improve communication quality.

II. PROBLEM SPECIFICATION

The problem is the presence of echo in phone line communication, where the signal from the far-end is reflected at the near-end and results in an attenuated replica reaching the far-end speaker. This echo interferes with the desired signal and degrades the quality of communication. The objective is to develop an adaptive line echo canceller (LEC) system that utilizes the far-end signal as input and its reflected version as the reference. The goal is to effectively eliminate the echo and enhance voice quality in phone line communication.

III. EVALUATION CRITERIA

The evaluation criteria for the line echo cancellation system include assessing its echo cancellation performance, voice quality improvement, robustness to circuit mismatches, adaptability to varying echo conditions, and computational efficiency. These criteria are used to determine the system's effectiveness, its ability to enhance voice quality, handle different scenarios, adapt to changes, and operate in real-time.

IV. APPROACH

We are using Matlab to tackle the issue of line echo in telecommunications. When a signal is sent from one end to another over phone lines, it can get reflected at the receiving end due to circuit mismatches. This reflection results in an echo that travels back to the sender. The echo causes interference, as the far-end speaker hears a faint replica of their own signal along with the desired near-end signal. Our project, titled "Line Echo Cancellation," aims to develop Matlab-based techniques to reduce or eliminate this echo, improving the clarity of communication. Figure 1 illustrates this echo phenomenon. domain or by using the Z-transform to convert both signals to the Z-domain and then performing the multiplication using Z-transform properties. The Matlab filter () function can be used for this. A signal of the same length as the input signal is generated. The learning factor must first be chosen within its valid range, followed by defining the filter length, which is crucial for the estimation process. To estimate the coefficients, a loop is used to iteratively apply a new window size to each value of the input signal. The loop repeats as many times as the length of the input or desired signal. The transposed windowed data from the input signal is multiplied with the estimated matrix, which has M columns and one row, each time. Alternatively, an estimated values matrix with one column and M rows can be used. In this case, the estimated values vector is transposed (w). The result of the multiplication is used to calculate the error, which is then used to update the estimated values for the next iteration.

V. ALGORITHMS USED

1. The Normalized Least Mean Squares (NLMS) algorithm is an adaptive filtering technique used to minimize the difference between a desired output and the output of an adaptive filter. It operates by iteratively adjusting the filter coefficients based on the error signal and the input data. The algorithm starts by initializing the filter coefficients and then proceeds to iterate over the input data. In each iteration, the output of the adaptive filter is computed by multiplying the filter coefficients with the input data. The error signal is then calculated as the difference between the desired output and the actual output. The filter coefficients are updated by taking into account the error signal, the input data, and a step-size factor. The step-size factor controls the rate of adaptation, while a normalization term ensures stability and prevents large updates when the input data has high energy. The iterations continue until convergence is achieved or a specified number of iterations is reached. The NLMS algorithm is widely used in applications such as noise cancellation, echo cancellation, and system identification, as it enables the filter to adapt and track changes in the system or environment, leading to improved performance and accuracy.

M: filter length,
μ:step-size factor,
x(n): input data to the adaptive filter of length N (Vector),
w(0) :initialization filter (vector) =zeros of length M,
Outputs at each iteration (n) y(n) = w^T(n) x(n),
calculate error : $e(n) = d(n) - y(n)$,
the error at n sample $w(n+1) = w(n) + \frac{1}{\epsilon+||x||^2}$ e(n) x(n) the updated filter coefficients , || x || is the norm.

Figure 1

Inputs: M: filter length
μ:step-size factor
x(n): input data to the adaptive filter of length N (Vector)
w(0):initialization filter (vector) =zeros of length M
Outputs at each iteration (n) $y(n)=w^T(n)x(n)$
$e(n)=d(n)-y(n)$
$w(n+1)=w(n)+\frac{1}{\epsilon+\|x\|^2}e(n)x(n)$, the updated filter coefficients , $\|x\|$ is the norm.

2- The Least Mean Squares (LMS) algorithm is a method used to adjust the coefficients of a linear filter in order to minimize the difference between the desired output and the actual output of the filter. It operates by iteratively updating the filter coefficients based on the error between the desired and actual outputs. The algorithm begins with initial coefficient estimates and calculates the output of the filter using the current input data. The error is then computed by comparing the desired output with the actual output. The coefficients are updated by adding a factor determined by the error and the current input data. This factor helps to gradually adjust the coefficients and reduce the overall error. The process is repeated for each sample, allowing the coefficients to converge towards the optimal values. By iteratively refining the coefficients, the LMS algorithm improves the performance and accuracy of the linear filter.

$(n) = [x(n), x(n-1), \ldots, x(n-p-1P]\ T$:
The window of size M
$(n) = [w0 , w1, \ldots wxM{-}1\ ]\ T$,
estimated values. $y(n) = wT\ .\ x(n)$,
the factor to calculated the error $e(n) = d(n) - y(n)$,
the error at n sample $w(n+1) = w(n) + 2 \times \mu \times$e(n) $\times\ x(n)$ the new estimated values So the above equations is in general the work if the LMS algorithm

Figure 2

LMS coefficients update criteria

Parameters: $p =$ filter order
$\mu =$ step size
Initialisation: $\hat{\mathbf{h}}(0) = \text{zeros}(p)$
Computation: For $n = 0, 1, 2, \ldots$

$$\mathbf{x}(n) = [x(n), x(n-1), \ldots, x(n-p+1)]^T$$
$$e(n) = d(n) - \hat{\mathbf{h}}^H(n)\mathbf{x}(n)$$
$$\hat{\mathbf{h}}(n+1) = \hat{\mathbf{h}}(n) + \mu\, e^*(n)\mathbf{x}(n)$$

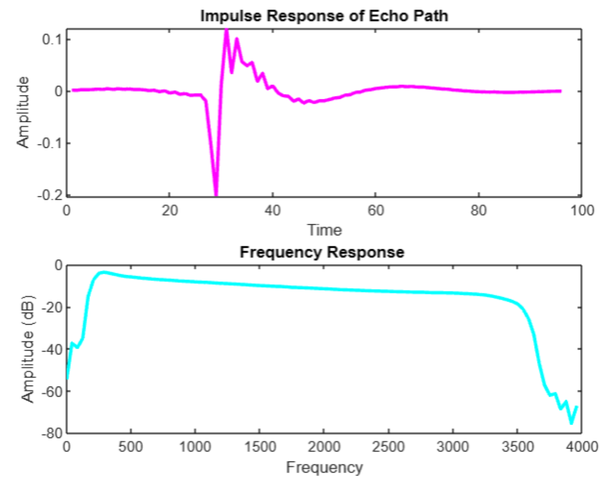
Where, $p = M, h = w$

VI. RESULTS AND ANALYSIS

A) After loading the file path.mat, which contains the impulse response sequence of a typical echo path. The impulse and frequency responses of the echo path were plotted.

Figure 3

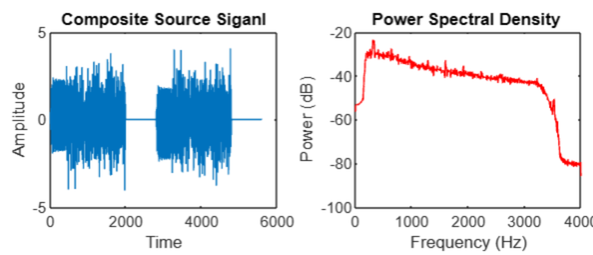
impulse and frequency responses of echo path in time domain



B) css.mat file loaded, which contains 5600 samples of a composite source signal, then plot the samples of the CSS data and its Power Spectrum Density PSD.

Figure 5

Composite source signal and its PSD



C) In this part we concatenate five such blocks and feed them into the echo path. In figure 6 shows the resulting echo signal . It calculates the input power and output power of the convolved signal, and then computes the Effective Radiated Power Loss (ERL) by subtracting the input power from the output power.

Figure 6

Far-end signal ,resulting echo signal

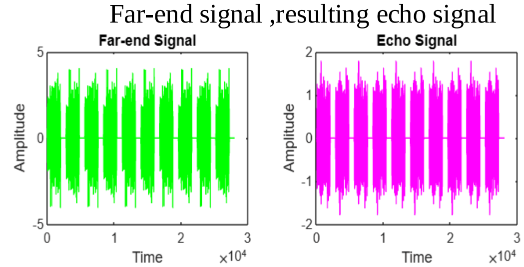


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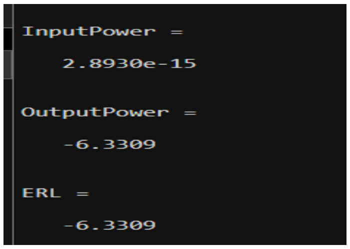
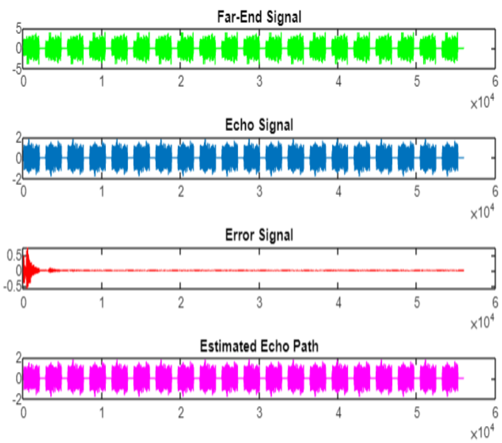
Figure 7
Power Computes



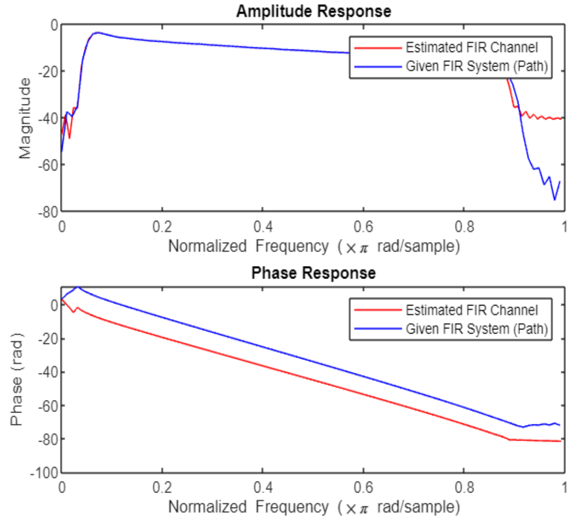
D) Using 10 blocks of CSS data as far-end signal, Implements an echo cancellation system using an adaptive filter. loads the necessary data, including the echo path and composite source signal (CSS), and generates the far-end signal by repeating the CSS. The code then estimates and cancels the echo using the adaptive filter, and plots the resulting signals and estimated echo path. This helps visualize the performance of the echo cancellation system.

Figure 8
echo cancellation system



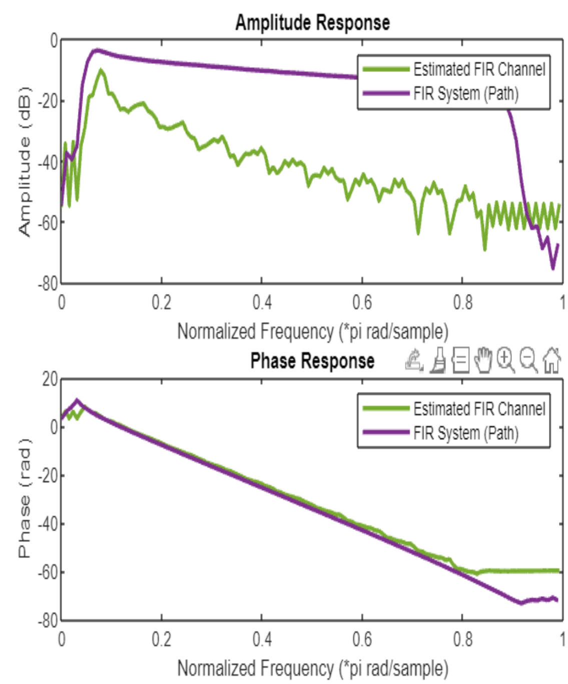
e) Estimates and compares the frequency response of an adaptive FIR channel with a given FIR system. And calculates the response by processing the far-end signal through the adaptive filter and echo path. The resulting amplitude and phase responses are plotted to visualize and compare the frequency characteristics of the two systems.

Figure 9
frequency response of an adaptive FIR channel with a given FIR system



F) In this part we repeat part E but with different algorithm Least mean square (LMS)

Figure 10
Frequency Response Comparison



As shown in figure 10 ,there was a distortion there was a noise in the Estimated FIR channel and the signal is different from FIR system

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VII. . CONCLUSION

In conclusion, Line echo cancellation is a vital technology used in telecommunications and audio communication systems to eliminate or minimize echo caused by signal reflections. By analyzing incoming and outgoing signals, estimating the echo path, and subtracting the estimated echo from the received signal, line echo cancellation improves audio quality. It finds extensive application in VoIP, video conferencing, and telephony networks, enhancing user experience and enabling effective communication. Ongoing research focuses on enhancing the effectiveness of line echo cancellation algorithms for even better results.

VIII. REFERENCES

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[2] An overview on optimized NLMS algorithms for acoustic echo cancellation | EURASIP Journal on Advances in Signal Processing | Full Text (springeropen.com)

[3] Normalized LMS Filters (keil.com)