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Line echo cancellation using Adaptive Filter Algorithms

DIGITAL SIGNAL PROCESSING ENCS4310

I. INTRODUCTION

In communications over phone lines, a common problem that we face is hearing echo after we talk, which implies that the speaker at the far-end receives, in addition to the desired signal from the near-end speaker, an attenuated replica of his own signal. And this problem causes interference and decreases the quality of transmitted audio. In this project, we will represent a solution to this problem using adaptive line echo cancellers (LEC), and evaluate its algorithms.

II. PROBLEM SPECIFICATION

The problem involves addressing line echo in telecommunications systems, which occurs when a signal traveling from a far-end point to a near-end point is reflected at the near-end due to circuitry mismatches. The reflected signal returns to the far-end as an echo, degrading the quality of the received signal.

The objective of the project is to explore and evaluate techniques for line echo cancellation using adaptive algorithms, in order to do this, the following tasks will be done:

1. Analyze the impulse and frequency responses of the echo path.
2. Process a concatenated source signal that emulates speech properties.
3. Compute echo-return-loss (ERL) to estimate the attenuation introduced by echo path.
4. Train an adaptive line echo canceller using the source signal and echo data.
5. Evaluate the performance of the adaptive canceller by analyzing the far-end signal, echo, and error signal.
6. Compare the estimated FIR channel with the actual echo path.
7. Try a different adaptive algorithm and compare its results with the first algorithm.

III. DATA

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The project uses two main data sources. First, the "path.mat" file contains the impulse response sequence of a typical echo path. This data will be used to analyze the characteristics of the echo path, including its impulse and frequency responses. Second, the "css.mat" file contains 5600 samples of a composite source signal that emulates speech properties. This data will be used for training and evaluating the adaptive LEC.

IV. EVALUATION CRITERIA

In order to measure the performance of the adaptive LEC, several evaluation criteria will be employed. These include measuring the attenuation introduced by the echo path as the signal travels through it, quantified as the echo-return-loss (ERL), which is by measuring the difference in power between the input signal before it enters the echo path and after. Additionally, we will analyze the error signal, far-end signal, and estimated echo path and compare it with the actual echo path to evaluate the effectiveness of the adaptive algorithm in canceling the echo.

V. APPROACH

We started by loading and analyzing the "path.mat" file to obtain the impulse and frequency responses of the echo path, and loading the "css.mat" file, to plot the composite source signal samples and their Power Spectrum Density (PSD). Then we concatenated five blocks of the composite source signal and feed them into the echo path, plot the resulting echo signal. Estimate the input and output powers, and calculate the echo-return-loss (ERL).

After that, Train an adaptive LEC with 128 taps using the far-end signal as input data and the corresponding echo signal as reference data. Utilize the normalized least-mean-square (NLMS) algorithm with a step size of 6e-10 and $\mu = 0.25$. Plot the far-end signal, echo, and error signal provided by the adaptive filter. Also, visualize the estimated echo path by the adaptive filter at the end of the simulation.

Analyze and compare the amplitude and phase response of the estimated FIR channel with the given FIR system (Path).

At last, propose an alternative adaptive algorithm and compare its performance with the NLMS algorithm used.

VI. RESULTS AND ANALYSIS

Part A

Here is the Plot of impulse and frequency responses of the echo path:

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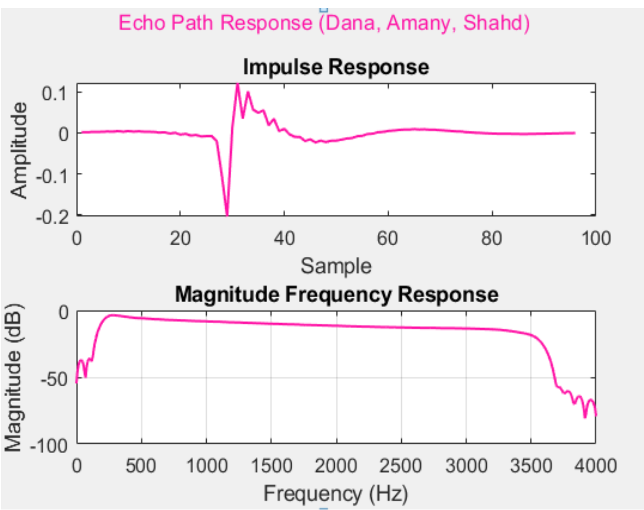
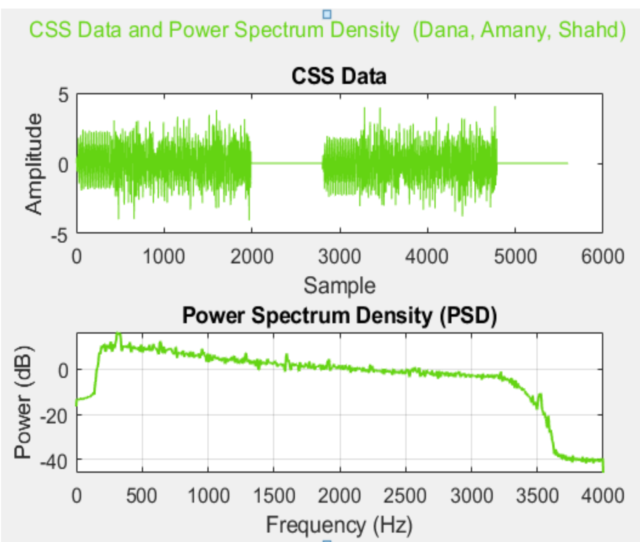


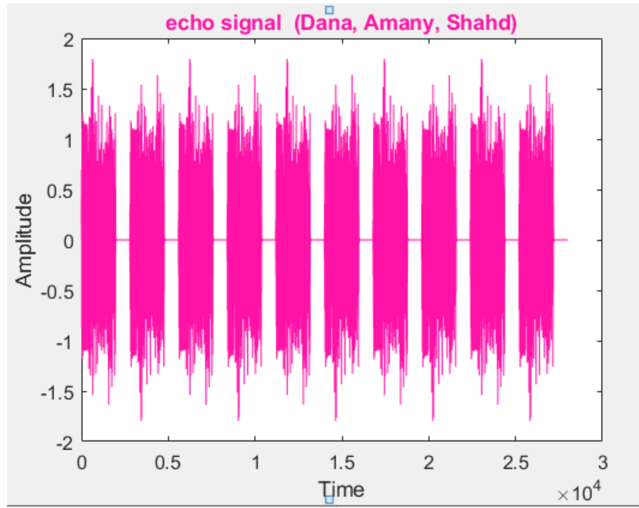
Part B

Here is the Plot of the CSS data samples, as well as their Power Spectrum Density PSD



Part C

Concatenate five such blocks and feed them into the echo path. Plot the resulting echo signal:



Note that the echo signal is an attenuated replica of the original input signal (css date) that is showed in part A.

Then we calculated the power of the input signal and the power of the echo signal to calculate echo-return-loss (ERL) Which represents the attenuation in dB that is introduced by the echo path as the signal travels through it.
Power equation:

$$\hat{P} = 10\log_{10}\left(\frac{1}{N}\sum_{i=1}^{N}\left|signal(i)\right|^2\right)$$

Result:

```
>> projectC
power of input signal Xc: 1.7358e-14
power of output echo signal: -6.3313
echo-return-loss (ERL):  6.3313
>>
```

The positive ERL indicates that the echo signal is attenuated or reduced by approximately 6.3313 decibels (dB) as it passes through the echo path.

Part D

Here we will work with the line echo canceller.
Plot the far-end signal, the echo, and the error signal provided by the adaptive filter. Plot also the echo path and its estimate by the adaptive filter at the end of the simulation.

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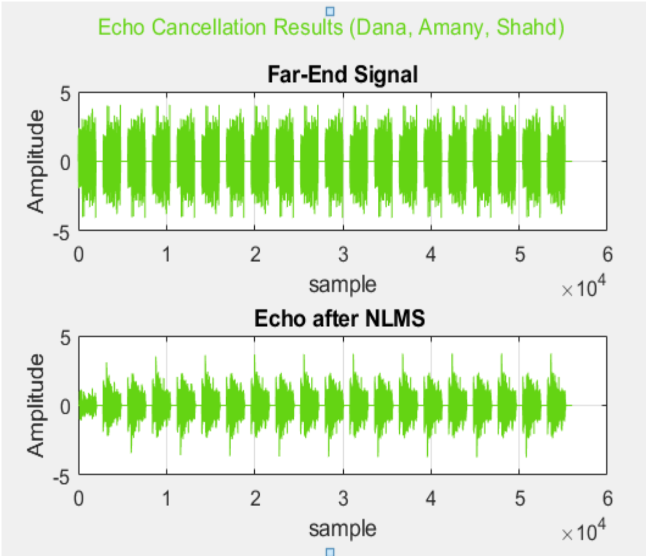
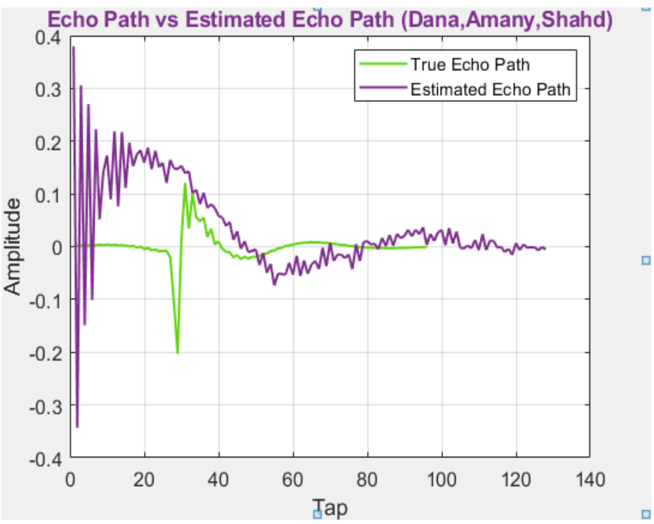
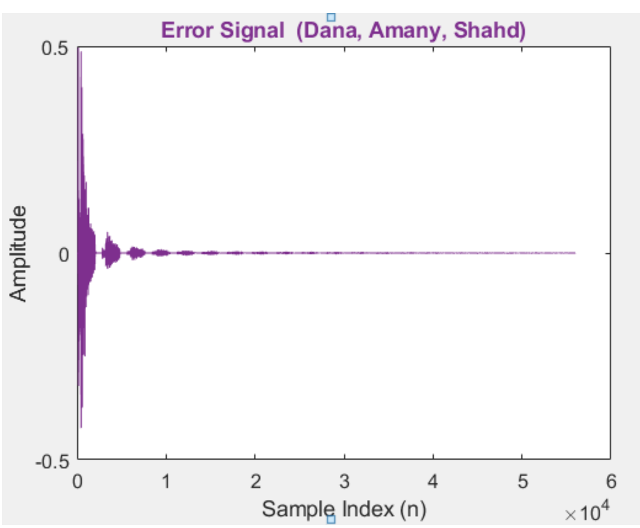






Note that the estimated echo path converges towards the true echo path which means that NLMS algorithm is working effectively Because the algorithm aims to adaptively estimate and cancel the echo by adjusting the filter coefficients (estimated echo path) based on the input signal and the error signal.

And we see that the echo signal after NLMS is the same as echo before, which is what we need for the error (e[n]) to be almost zero, and the signal reached to the other end will be clean.

The pseudo for $\in$-NLMS adaptive algorithm:

$$d[n] = \sum_{k=0}^{M-1} h[k]x[n-k]$$

$$y[n] = \sum_{k=0}^{M-1} w[k]x[n-k]$$

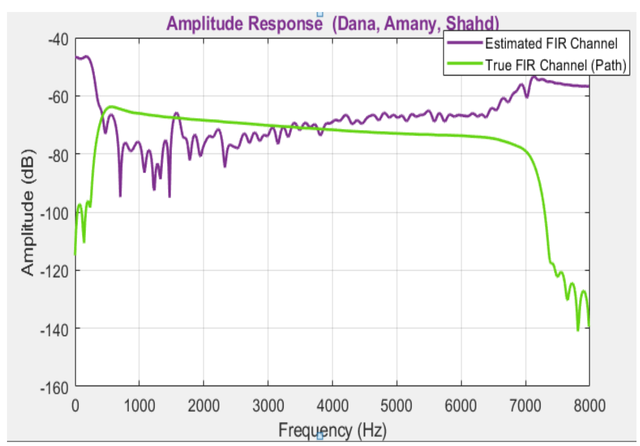
e[n]=d[n]-y[n] must be equal to zeros.

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) $y(n)=w^T(n)x(n)$
$e(n)=d(n)-y(n)$
$w(n+1)=w(n)+\frac{1}{\in+\|x\|^2}e(n)x(n)$, the updated filter coefficients , $\|x\|$ is the norm.

Part E

Plot the amplitude and phase response for the estimated FIR channel at the end of the iterations.



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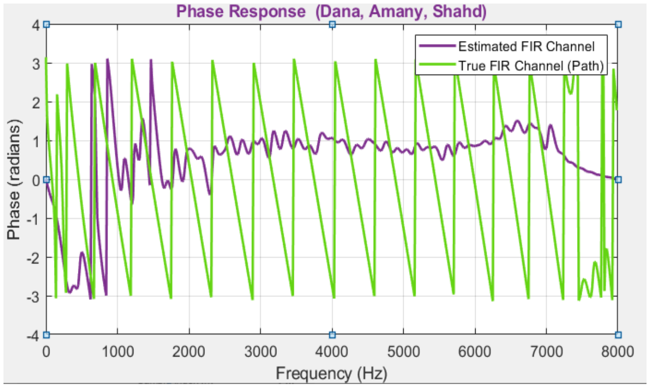


They are almost the same, but the estimated is smaller in phase.

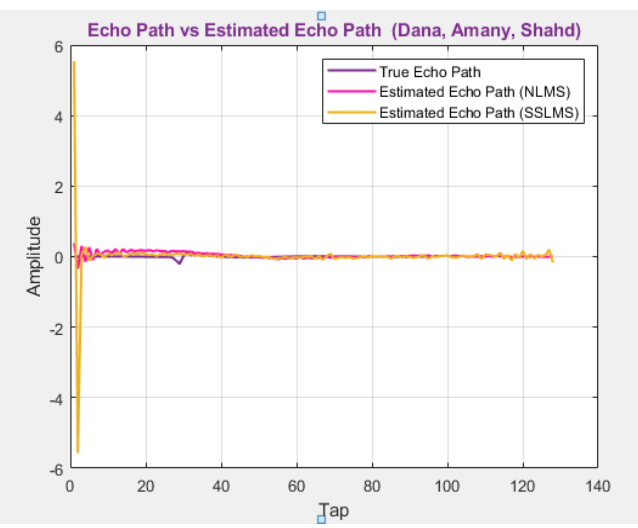
Part F

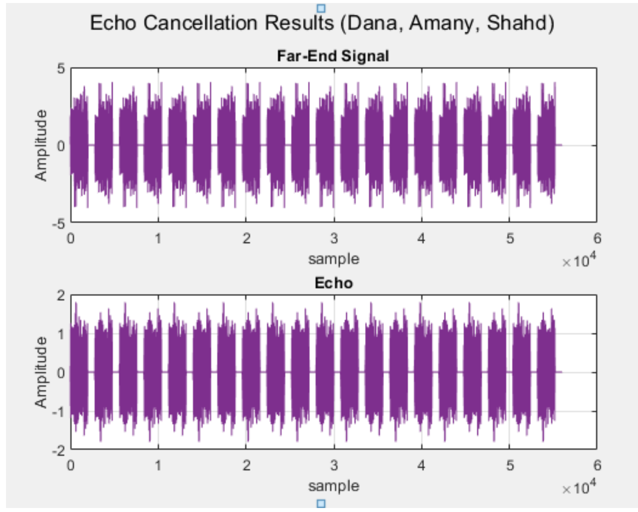
Propose a different appropriate Adaptive algorithm and compare it to the ∈-NLMS
We will use **SSLMS** algorithm.

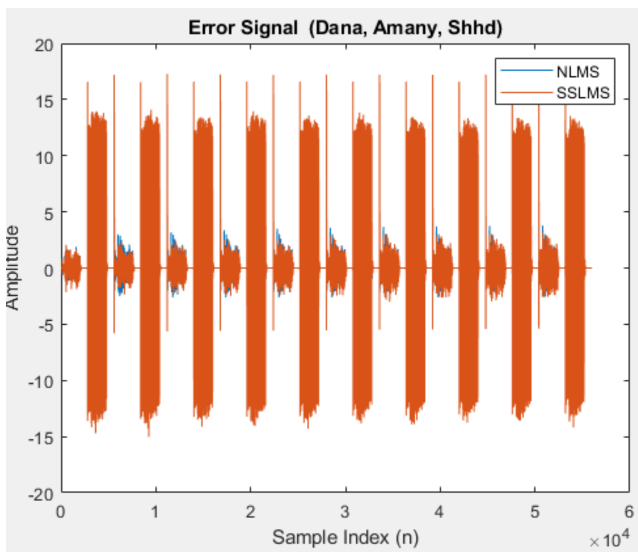
```
# SSLMS algorithm
Initialize:
  Set filter_coeffs_sslms = [0, 0, ..., 0]  // Initial filter coefficients
  Set step_size = 0.25  // Step size parameter
  Set leakage = 1e-6  // Leakage parameter
  Set Xcc = [X, X, ..., X]  // Concatenated input signal

For n = num_taps to length(Xcc):
  // Adaptation
  x = Xcc(n:-1:n-num_taps+1)  // Input vector
  y_sslms = filter_coeffs_sslms * sign(x).'  // Estimate of the echo signal
  e_sslms = echoSignal(n) - y_sslms  // Error signal
  filter_coeffs_sslms = filter_coeffs_sslms + (step_size / (norm(x)^2 + leakage)) * sign(e_sslms) * x  // Update filter coefficients

End for
```







Note that the three paths are the same.
But the error signal in SSLMS is bigger because SSLMS tends to have slower convergence and may exhibit more fluctuation in the error signal during the adaptation process. This can result in larger error values compared to NLMS.

VII CONCLUSION

In conclusion, the project provided a practical understanding of line echo cancellation and the use of adaptive algorithms to solve the problem of echo in communication systems.
We learned about the NLMS and SSLMS algorithms that worked effectively to cancel the echo and improve voice quality. The project also highlighted the importance of accurately estimating the echo path and analyzing the amplitude and phase response. Overall, the project enhanced knowledge and skills in signal processing and offered real-world applications for solving the problem of echo interference.

VIII REFERENCES

https://www.mathworks.com/products/matlab.html