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Development of an adaptive finite impulse response filter optimization algorithm using rough set theory

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ABSTRACT

Signal processing is crucial that as one sends information, there is a corresponding process to encode, decode, and clean the signal of unwanted noise and disruptions via use of filters. Due to the environment and how unpredictable it can be and how noise can come from almost anywhere, the typical filter to be used are adaptive filters. Adaptive filters are non-linear filters and have been used regularly regarding adaptive signal processing, this means that the filter changes accordingly and adapts to the environmental noise surrounding it. The world today has numerous applications for adaptive filters such as channel equalization and acoustic noise cancellation. This incentivizes the further development of this specific technology and the constant research that is ongoing. The main component when it comes to adaptive filtering is the algorithms used for the filter. This research compares the least mean square (LMS) and recursive least square (RLS) algorithms concerning their effectiveness in filtering out unwanted acoustic noises. The paper will cover the design and implementation of an optimized rough set based adaptive finite impulse response (FIR) filter for acoustic noise cancellation. The microstrip and bowtie antenna were used to relay the data. The software MATLAB was used for the simulation.

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1. INTRODUCTION

Digital signal processing has been a major component of any form of communications system all around the world [1], [2]. Moving with the times as the world progresses, so does the technology of signal processing. A non-linear filter known as the adaptive filter has had a major role in the field of digital signal processing [3]-[5]. This is due to the filter's capability to adapt in real time depending on the changing noisy environment that surrounds it [6]. This means that as the input signal changes, so does the process of noise attenuation [7], [8]. When talking about adaptive signal processing the component that dictates the changes are the algorithms it uses such as the least mean square (LMS) and recursive least square (RLS) algorithms, These filters have numerous applications, one of the most common is in the study of acoustic noise cancellation. There are plenty of advantages when making use of this specific filter, the main concern however is the complexity and length of said filter [9]. Due to its computational complexity, the system becomes less ideal in applications due to concerns in terms of the speed at which it can operate. With numerous applications of adaptive filters, studies are now primarily concerned with the improvement of the speed and efficiency of these forms of filters [10]. These are being done by manipulating the individual components of the software such as the algorithm, and the hardware itself.

Adaptive filters have two major steps in processing the input signal, The two major components are the adaptation of the filter and the filtering of the input signal itself [11]. The adaptation of the filter functions by changing the filter coefficients to limit the desired cost function. The filtering process on the other hand works by applying the changes and generating an output based on the given signal and noise [12]. There are many factors when designing a filter, and many components to consider [13], [14]. A factor is acoustic noise cancellation as many gadgets function such as headphones, earphones, and microphones. [15]-[17]. This will be done by comparing the two algorithms namely, the LMS and RLS algorithms [18]. The research will make use of a simulated version that is constructed and programmed via MATLAB Simulink [19]. The goal would be to compare and contrast the effectiveness and efficiency of both algorithms, and to be as accurate as possible when it comes to simulating the filter.

2. BACKGROUND OF THE STUDY

Linear and non-linear filters have been a key components in communication systems and signal processing ever since their development. Adaptive filters are one of these filters and have been one of the recent developments in its kind. Adaptive filters work by changing the transfer function of the filter and by optimizing the filter depending on the given input signal [20]. These algorithms work by computing and calculating the changes that are needed to be done to the transfer function and to the other components in the system. Adaptive filters function in a closed loop and make use of a cost function, the main point being to lessen the cost of adaptation per iteration by keeping the past iteration stored. This means that as time passes the system is more and more efficient at changing the transfer function. When taking a look at nonlinear filters such as the adaptive filter it is clear that it is more complex than other filters. It has more computational complexity and does not follow homogenic and additive properties.

The two algorithms that are being compared in this paper are the RLS and LMS algorithms. The recursive least square algorithm works by exactly minimizing the sum of the square of the designated signal estimation errors. This is done with the knowledge of the initial conditions and variables. These conditions vary as the filtering progresses. Since the simulation takes place in a dynamic environment this means it gives out new input signals and new initial conditions depending on the samples [21]-[23].

The least mean square algorithm on the other hand works by changing the filter taps of the system and matching them to the change in gradient of the error surface. The algorithm also lessens the mean square error by making sure past iterations are integrated to make future corrections and changes easier and more in the direction of the negative gradient vector. This is one of the more simple algorithms yet it is still effective as an algorithm to be used in an adaptive filter.

3. STATEMENT OF THE PROBLEM

The goal of digital signal processing is to have a clean output signal that is clean of noise and unwanted signals. In acoustics, digital signal processing is done by canceling out the unwanted noise. This is made possible by the use of adaptive filters. Every day the demand for better quality acoustics is getting higher and higher. The environment in real-life applications is dynamic. This means that noise and disruptions can come from multiple sources with varying degrees of intensity. Industrial machinery, automobiles, echo, and so much more can contribute to the disruption of sound such as speech. Adaptive filters are the only type of filter that can accommodate these types of problems since the filter changes its filter on the go. There are several problems however to consider when designing an adaptive filter, these include slow convergence times, high computational complexity, presence of residual echo, and near-end distortion for some of the more advanced algorithms. All of these factors will be considered plus the addition of the mean square error.

4. SIGNIFICANCE OF THE STUDY

This study aims to know and simulate the difference in the quality of outputs in adaptive filters when using two different algorithms, which is significant today as adaptive filters are being used in numerous applications. Applications such as the ones for headphones, earphones, and speakers. All of this research is to determine which algorithm would perform better in practical application. The factors to be considered in determining whether it did better in reducing noise are the MSE or the mean square error.

To do this, however, before designing the filter also means considering the different properties of a filter such as convergence times, computational complexity, residual echoes, near-end distortions, and many more properties that can affect the filter's capacity and effectiveness in filtering out noise. These factors need to be considered as they would also dictate the practicality of such a system outside of simulations and in actual use. Simulations and trials will be done in MATLAB Simulink. This way the utilization of MATLAB can lead to easier simulations and familiarity with the programming language of MATLAB which will be useful for further research in the future. Different Simulink blocks will be used to determine and design an adaptive filter that is near to life. Several parameters will be added and set to aid in getting a more accurate result and representation of the filter if it is the actual practice or use.

5. THEORETICAL CONSIDERATIONS

To define and categorize inadequate or insufficient information, Pawlak [24] suggested a set theory extension known as rough set theory. Additionally, it is a mathematical instrument that dispels uncertainty and skepticism. Additionally, it confirms logic, permits the discovery of incomplete implications, and permits conflicting data and uncertainty. By organizing, it creates knowledge that is insufficient, ambiguous, and incomplete. The appropriate data are organized in a rough set for analysis. Knowledge finding and processing are crucial for decision systems because real-world applications may incorporate some uncertain and partial qualities in knowledge representation systems in dynamic situations. A useful tool for finding patterns with upper and lower approximations, it is supported as a framework for understanding and analyzing some specific and ambiguous sorts of data [25]. Figure 1 shows the rough set theory analysis.

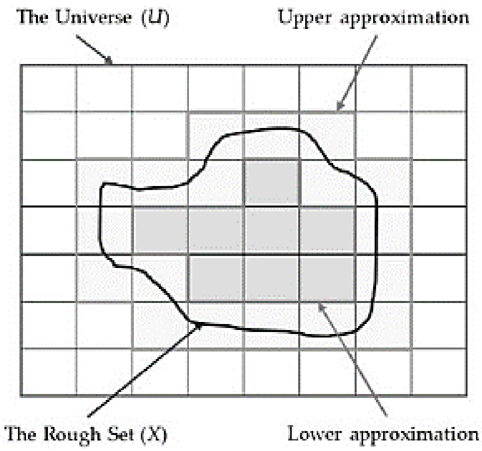

Figure 1. Rough set theory analysis [25]

Rough set theory can be used to optimize the three main components which are the adaptive algorithm, source, and digital filter. The adaptive filter has two types of sources which are the noise and the desired acoustic signal. A sine wave converted into a discrete signal will be the form of the desired acoustic signal. The noise signal will be random in form and is generated from a random source. The digital filter will have a shape factor of 0.5 and a Kaiser-Bessel window. The filter will have a form that is direct to the filter structure and a length of 32. The normalized specification of the type II filter is in a range of 0 to 1. The algorithms used are RLS and LMS algorithms. The output is composed of four types which are filter output, weight output, noise signal, and the signal error produced.

6. METHOD

The methods and simulation used in this paper are utilized to compare and contrast the performance and efficiency of the system when using the RLS and the LMS algorithms. The simulations utilized in the paper for filter realization made use of MATLAB’s filter design and analysis (FDA) tool and a simulated environment that was modeled in Simulink. In the simulation’s design, there was a need for setting specific parameters in the filter design. These parameters are composed of the filter’s structure, length, and type which was set to direct form, 32, and type 2 respectively. rough set theory will be used to determine the efficient parameters for the testing. Figure 2 shows the simulation flow chart.

The simulation made use of the two algorithms (RLS and LMS). The simulation time was set to 30, 50, and 80 per iteration. The design method was the Kaiser Window with β=0.5. The frequency was normalized at 0-1. This was done to obtain optimized results.

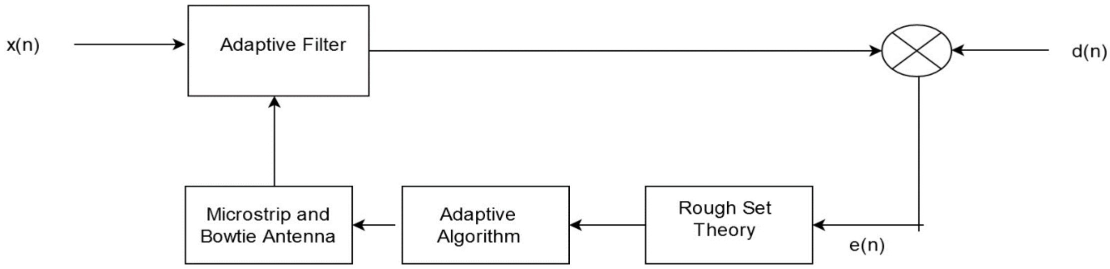

Figure 2. Simulation flowchart

7. REVIEW OF THE RELATED LITERATURE

Dewasthale and Kharadkar [26] states that various adaptive filtering techniques are applicable in the use of acoustic noise cancellation. The quality of the signal being transmitted is tied to the noise from the surrounding environments and can reduce the transmitted signal's efficiency. The research includes the possible improvements in the transmission that tackles noise cancellation where algorithms, principles, and various filter structures are implemented to take care of the problem. The researchers then saw that testing different adaptive filter algorithms to determine if these methods are a possible aid in helping the transmissions' performance be improved with noise canceling properties. Testing the different methods and algorithms, the researchers determined that various adaptive filter structures are possible to be formulated as a system alongside performance adaptive algorithms for noise cancellation. The algorithms, in the end, were found to be limited to specific operational limitations which can be further improved in progressive research regarding this topic [26]. According to Niranjan and Ashwini [27] music signals use adaptive filtering algorithms for noise cancellation. Their paper focused on the cancellation of noise signals by making an adaptive filtering method. The paper suggested that with certain fixed parameters the filtering technique can help enhance the signal, but with active fluctuations, the algorithm does seem ineffective. The advantages of the use of an adaptive filter over those that don't are further elaborated to better understand the filter characteristic requirements to aid more in noise cancellation. The researchers then suggested another method such as the LMS algorithm and averaging algorithm to tackle the problems presented by rapid changes occurring to signals. With these conditions set, the researchers compared the results of using each respective adaptive filter algorithm by an experiment. The testing included testing different frequencies where the step size of each output is observed to have discrepancies. The conclusion the researchers have reached is that the LMS algorithm and averaging algorithm are efficient for use of the adaptive filters since these two methods are versatile when it comes to the change of frequencies. Another limitation however is that the time and area of computation in the LMS is a less efficient method than averaging algorithm [27].

In the research paper of [28], it was expressed that in computerized signal handling if two finite impulse response filter and an infinite impulse response filter cannot achieve ideal separating an adaptive filter must be utilized to follow the progressions of clamor in the sign. The specialists gave a has suggested that the algorithm is pertinent to adaptive filters which are the least mean square algorithm and the recursive least squares algorithm. Taking the conversation into account, the analysts directed tests by utilizing the personal computer programming language MATLAB to mimic, analyze, and demonstrate that utilizing the two recently referenced calculations could improve execution when contrasted with other ordinary channel plans. Through the experiment, it was discovered that the RLS calculation had a substantially more good outcome and had a considerably more predominant presentation as its combination time was a lot quicker when being contrasted with utilizing the LMS calculation. Moreover, it was noticed that regardless of whether the LMS did not create a similarly acceptable outcome a greatly improved yield can be accomplished by changing certain variables and the length of the channel strategy. Considering this, the scientists inferred that the two calculations can be utilized to finish an adaptive filter system when being utilized for a versatile channel framework.

This article on integrated acoustic echo and noise cancellation systems by Jayakumar and Sathidevi [29] centers around making a solitary framework that can attenuate noise echo while simultaneously improving the coherence of discourse in a hands-free portable device because of numerous frameworks that manage reverberation and commotion independently, there is a colossal absence of exploration concerning a single system that could manage both simultaneously. The impacts of acoustic echoes are most clear when the reverberation is postponed by many milliseconds Jayakumar and Sathidevi [29] however, like the ones expressed above, utilizes adaptive filtering for their integrated system. The framework clarifies estimating the room impulse response (RIR) to then use in figuring the acoustic reverberation of that condition. The reverberation is lessened by taking away the assessed acoustic echo from the mouthpiece input signal. Even though the most widely recognized techniques for versatile sifting are LMS and RLS, the paper utilized an improved adaptation that cut down on computational multifaceted nature by utilizing a channel shortening channel. Proceeding onward to the clarity of discourse, notwithstanding, Jayakumar and Sathidevi [29] specifies various manners by which it tends to be cultivated with regard to managing cell phones. These strategies extended from linear and non-linear spectral subtraction wherein the spotless discourse range is gained and handled to deliver high caliber and clear discourse to multiband spectral subtraction wherein ghostly deduction is done per discourse band. The paper utilized discrete wavelet transform (DWT) and perpetual wavelet packet transform (PWPT) as the fundamental segments when it went to their acoustic noise cancellers (ANC). Taking everything into account, the paper utilized a framework that utilized DWT or PWPT to decrease the reverberation of a given framework, while utilizing a low parity multifaceted range assessor to improve discourse quality [30]. The framework was successful in uproarious situations and significantly under low signal-to-noise ratio (SNR) conditions [31], [32].

8. DATA AND RESULTS

This section discusses the data and results. Figures 3 and 4 shows the finite impulse response (FIR) filter using the LMS and RMS algorithm respectively. Figures 5 to 16 shows the scope output using the LMS algorithm for 30 seconds, scope output using the LMS algorithm for 50 seconds, scope output using the LMS algorithm for 80 seconds, coefficients adjustment using the LMS algorithm for 30 seconds, coefficients variation using the LMS algorithm for a 50 seconds, coefficients variation using the LMS algorithm for 80 seconds, scope output using the RLS algorithm for 30 seconds, scope output using the RLS algorithm for 50 seconds, RLS algorithm for a simulation time of 80 seconds, coefficients variation using the RLS algorithm for 30 seconds, coefficients variation using the RLS algorithm for 50 seconds and coefficients adjustment using the RLS algorithm for 80 seconds respectively.

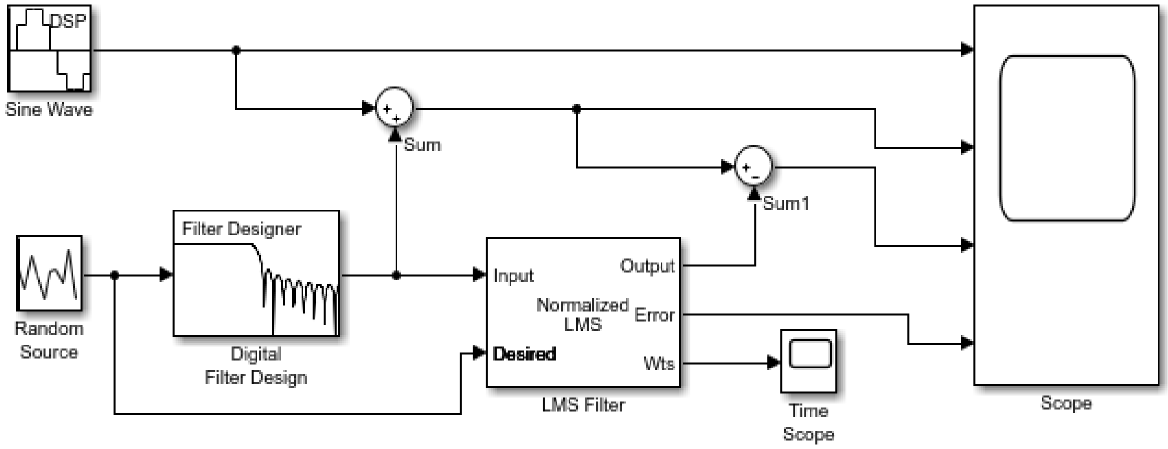

Figure 3. Adaptive FIR filter using the LMS algorithm

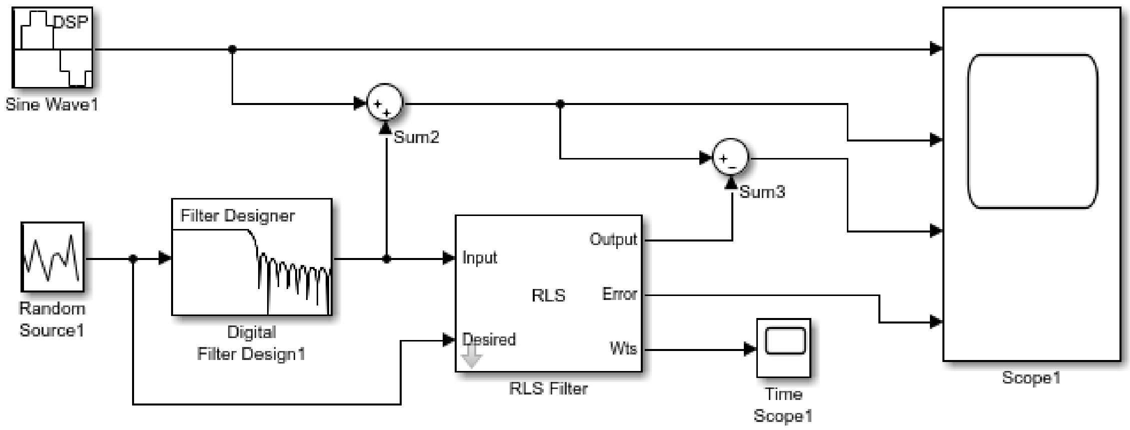

Figure 4. Adaptive FIR filter using the RLS algorithm

9. ANALYSIS OF DATA

From the information obtained the LMS and RLS adaptive FIR channels are built and simulated utilizing the Simulink blocks. As observed from the figures, both channels can play down the noise from an arbitrary source. Also, seen in Figures 5 to 16, the behavior of the noise is seen to be drawing nearer zero, for the coefficients of the versatile channels are being balanced over time. Firstly, as seen in Figures 5-7, the recreation time is specifically corresponding to the viability of noise cancellation. As the simulation time raises, the adequacy of the noise removal moreover increments. As seen within the information for the LMS calculation, the simulation time of 80 seconds in Figure 7 appears to have the most elevated noise cancellation, and the simulation time of 30 seconds in Figure 4 shows to have the most reduced noise, but ineffectively. As seen within the information in Figures 8-10, an increment in reenactment time moreover increments the adequacy of the clamor expulsion where the clamor begins to approach zero. Moreover, seen in Figures 8-10 that the LMS algorithm with an initial value of 2 has a slow convergence time and slowly approaches zero.

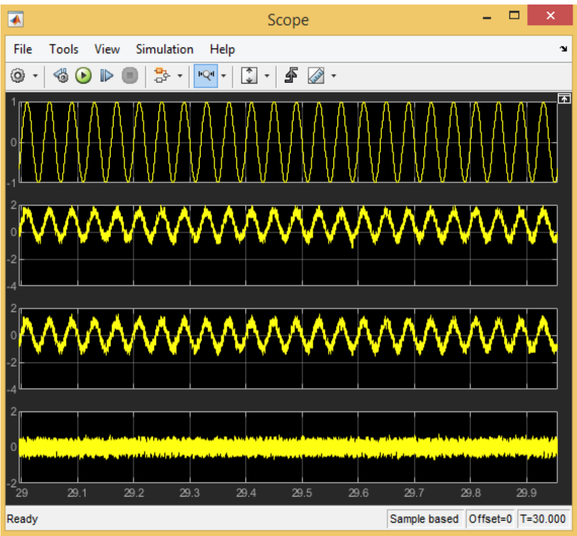

Figure 5. Scope output using the LMS algorithm for 30 seconds

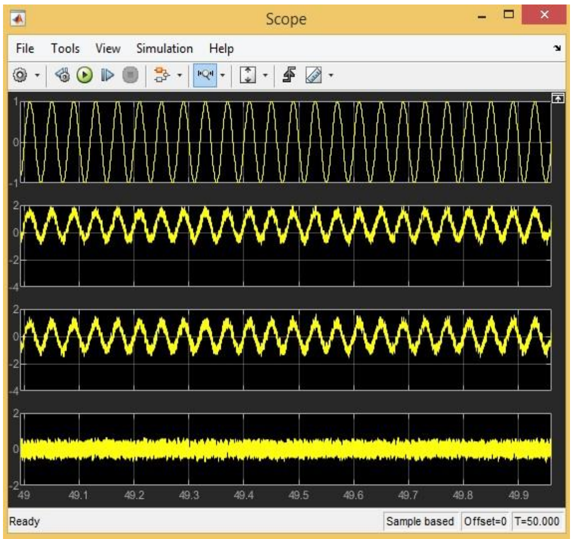

Figure 6. Scope output using the LMS algorithm for 50 seconds

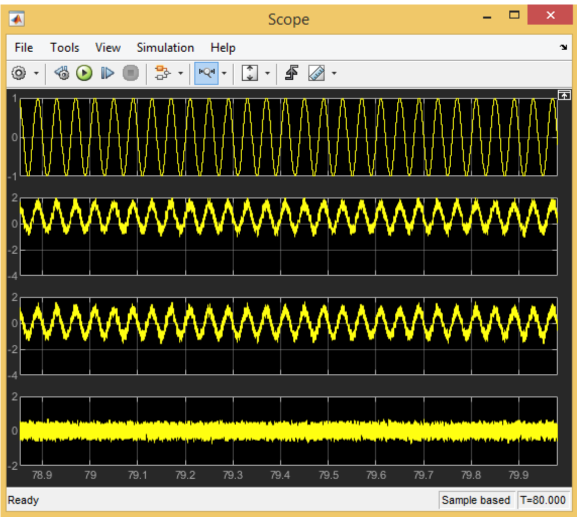

Figure 7. Scope output using the LMS algorithm for 80 seconds

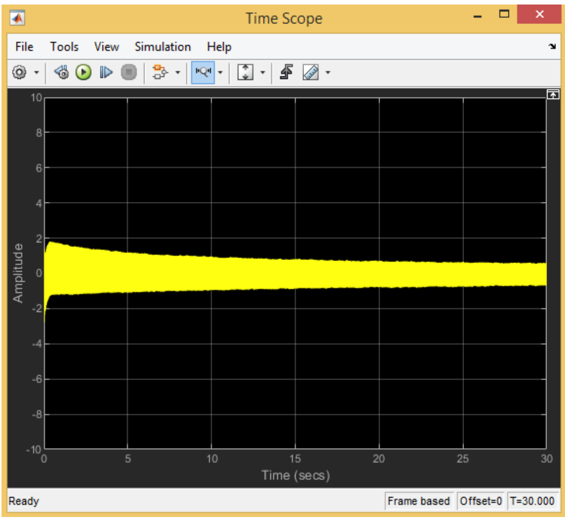

Figure 8. Coefficients adjustment using the LMS algorithm for 30 seconds

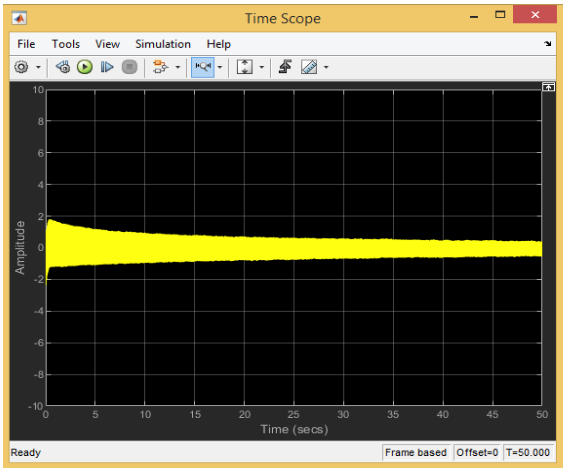

Figure 9. Coefficients variation using the LMS algorithm for 50 seconds

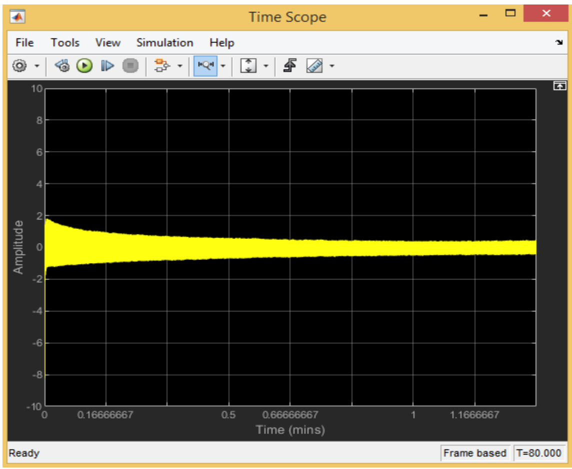

Figure 10. Coefficients variation using the LMS algorithm for 80 seconds

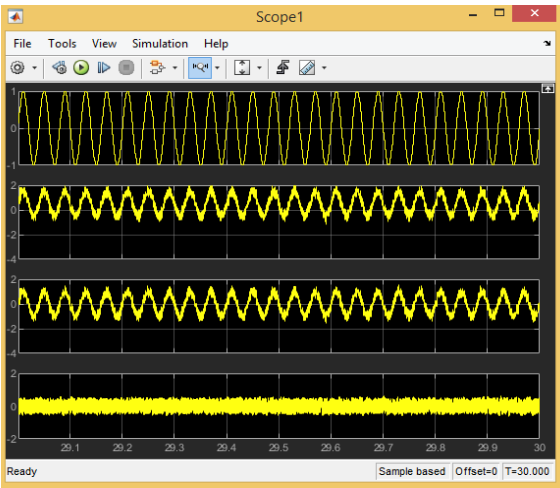

Figure 11. Scope output using the RLS algorithm for 30 seconds

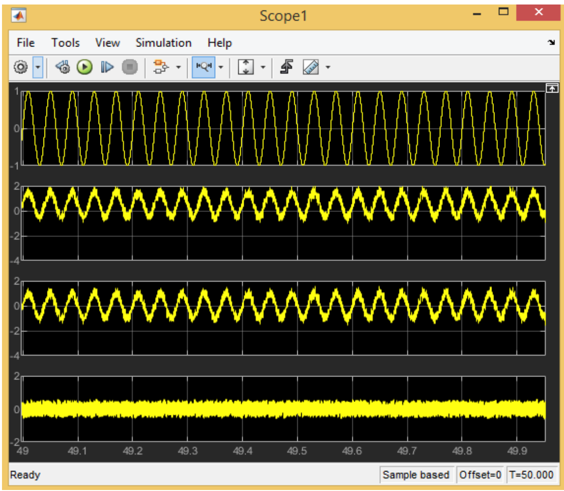

Figure 12. Scope output using the RLS algorithm for 50 second

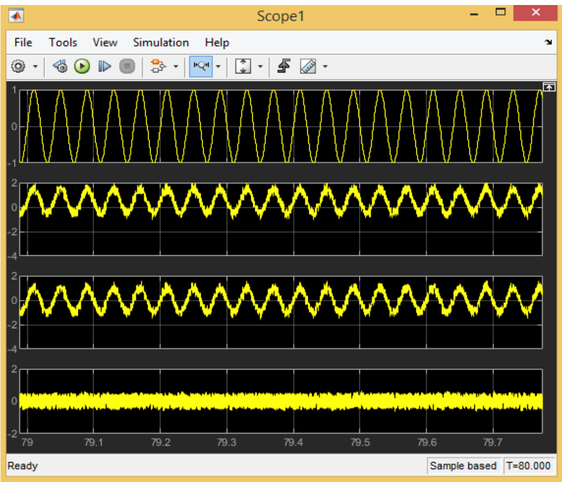

Figure 13. RLS algorithm for a simulation Time of 80 seconds

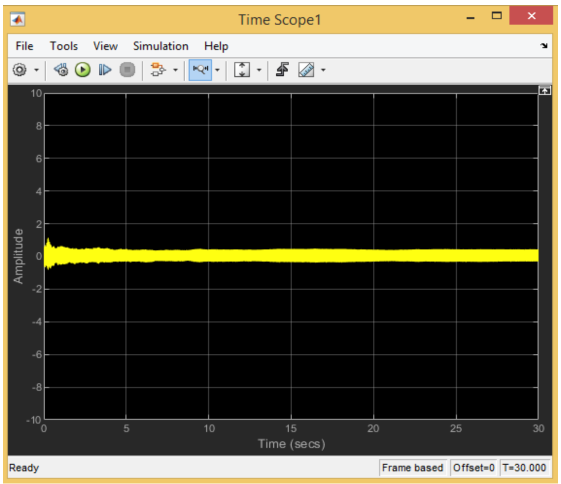

Figure 14. Coefficients variation using the RLS algorithm for 30 seconds

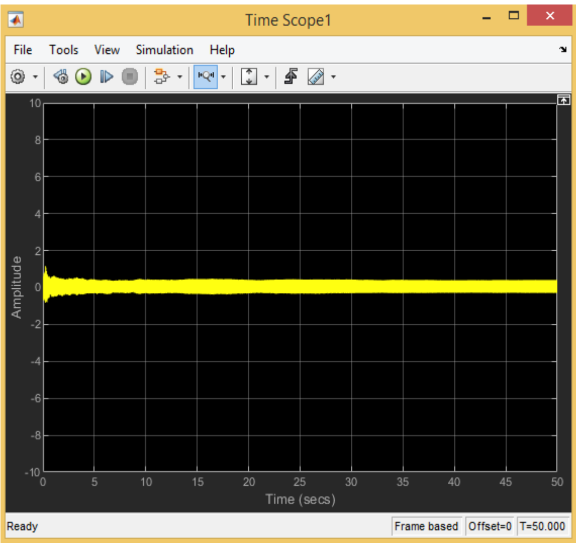

Figure 15. Coefficients variation using the RLS algorithm for 50 seconds

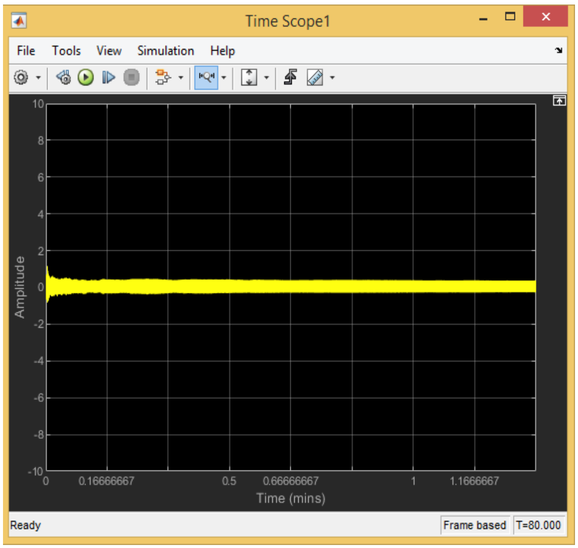

Figure 16. Coefficients adjustment using the RLS algorithm for 80 Seconds

10. CONCLUSION

MATLAB Simulink provided a good platform to test and simulate the two algorithms in an adaptive filter. However, after numerous testing on different simulation times and iterations, it was clear that among the two algorithms the least mean square algorithm was superior when compared to the recursive least squares. This statement holds when looking at the aspects of filtering, meaning attenuating the unwanted signals and noise. It is seen in the simulations that the LMS algorithm filtered the signal with better quality as compared to the RLS algorithm. The LMS adaptive filter provided higher quality signal outputs. The downside however to the LMS algorithm is the slow convergence time that it had. This is seen in the figures as the LMS algorithm takes longer for it to converge to a stable signal while for the RLS it was considerably faster. Overall however it can be concluded that the LMS algorithm fared better when compared to the RLS algorithm not only because it filtered the unwanted signals and noise better, but also due to its simplicity. One of the main problems and restrictive aspects of an adaptive is its length and computational complexity, these make it difficult for it to be viable in a practical application. With more components comes more cost and more hardware making it less practical, not only economically but also technically. It was stated above that not only the MSE was being considered but also factors such as convergence times, computational complexity, residual echoes, and near-end distortions. This means that when comparing the two filters it is clear that the LMS algorithm makes for a much simpler filter, with even better results. Although it may be slower in the convergence time, the adaptive filter which uses the LMS algorithm is still preferred especially in practical applications due to it taking fewer components and due to it having a much simpler design. The microstrip and bowtie antenna were used to transmit the data.

11. RECOMMENDATIONS

Due to the limited number of trial conditions performed, specifically the varied simulation time durations, it is recommended for future works or experiments to attempt to obtain results from even more simulation time durations. It is suggested to maintain a substantial interval between simulation times such as having 20-second intervals or more between each set. Doing so may allow for further observation of the performance of both the LMS and RMS algorithms for prolonged durations such as those far greater than 80 seconds, as performed in the simulations done by the researchers of this paper.

Furthermore, it is recommended for future works to improve the convergence rate of the LMS algorithm to lead toward developments or improvements in Adaptive FIR filters. A more difficult but rewarding task would also be to develop a new algorithm that provides the same desirable performance of an LMS algorithm while having a convergence rate closer to that of the RLS.

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