المؤتمر العلمي الدولي الثالث للعلوم و الهندسة
The 3RD Scientific International Conference in Science & Engineering

http://bwu.edu.ly/icse2024
Received 25/07/2024 Revised 20/08/2024 Published 10/09/2024

VOIP Acoustic Echo Cancellation Performance Improvement Using a Novel Time-Varying Step Size of NLMS Algorithm

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Abstract: When multiple people participate in an online meeting, because the sound is in the space Factors such as medium transmission or reflection make the sound emitted by the speaker It is taken back by the microphone again, causing echo. if there is When a user's device generates an echo, the echo signal will Spread among participants, affecting all users ring. The echo delay time is difficult to predict, and because network transmission and Factors such as sound distortion cause the traditional echo cancellation mechanism to frequently fails in multi-person network chats. In VoIP phones, acoustic echo cancellers are generally used to cancel Echo generated during calls to improve voice quality. adaptive filter and double-ended call detection filter. Adaptive filters eliminate echoes by adaptively modeling the echo path. sound, and its performance determines the effect of echo cancellation. In practical applications, adaptive. The most commonly used filter is the NLMS algorithm and its various improvements. algorithm, the reason is that the NLMS algorithm is simple and has low complexity. However, NLMS The algorithm has a fatal weakness, that is, when the input signal is like speech, etc. When there are signals with strong correlation, the convergence speed of the algorithm will be significantly reduced. And affect the quality of echo cancellation, effectively improving the convergence speed of the NLMS algorithm it is based on the instantaneous energy of the input signal the calculation step size will inevitably bring about random fluctuations in the steady-state offset error, thus Will affect the performance of the algorithm.

This paper proposed a modified of NLMS method to the decorrelation significantly reduces the updating the decorrelation filter coefficients and through the step size determination machine proposed in this study control, which can effectively distinguish the difference between normal speech and echo. Effectively eliminates echo and exerts silence suppression (Silence Suppression) effect, blocking content that does not contain voice content packets. It is shown that our method can eliminate more than 90% of echoes When utilizing the suggested technique in place of the conventional NLMS algorithm, the ERLE boost is around 11 dB. Means that more robust variant of the algorithms and it exhibits a better balance between simplicity and performance than the conventional NLMS algorithm.

Keywords: Acoustic Echo Cancellation (AEC), Least Mean Square (LMS), Normalized Least Mean Square (NLMS),Voice Over IP (VoIP),Echo Return Loss Enhancement (ERLE).

1. Introduction.

Several customers opt to utilize their computers' built-in speakers and microphones for online VoIP conversations rather than investing in additional equipment. In this manner, once the speaker plays a sound, it travels through the air, reflects off objects in the room, and then is picked up by the microphone once more, creating echo in the process. This phenomena can have an impact on the call's quality and, in extreme situations, even the meeting's progress. For instance, in "Fig. 1", once the speech of speaker A is sent to the speaker of listener B, the microphone retrieves the voice from the wall and plays it back to speaker A, resulting in echo. The "acoustic path" is the route taken by sound as it travels from the speaker to the microphone [1].

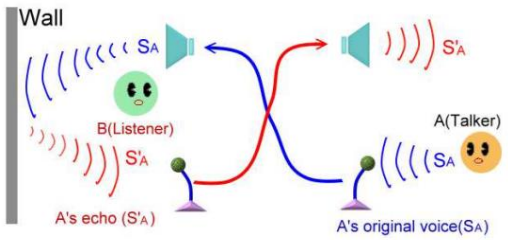

Fig. 1: Echo caused by transmission or reflection of sound.

VoIP lacks conventional sound playback and capture hardware, which causes a wide variation in the transmission delay time of echo on the acoustic route. Furthermore, the circumstances and features of echo might become unpredictable when numerous persons are involved in a discussion, therefore the VoIP system Solving the echo problem in its entirety is difficult.

Every device involved in a VoIP conversation between two or more people typically a personal computer usually has an echo cancellation mechanism. This allows the echo to be effectively eliminated without interfering with the conversation, even when the microphone picks up the speaker's sound. move forward. Because participant B's speaker is too close to the microphone, the microphone picks up the signal the speaker is sending out. On device B, this signal is successfully removed without interfering with the online chat when the echo cancellation mechanism is turned on regularly as shown "Fig. 2". The elimination method might not function as intended, though, because computer hardware and software can malfunction. In a multi person meeting, the entire thing might be harmed if one person's echo cancellation system malfunctions and they start producing echo instead of canceling it. impact, causing all users' calls to be interfered [2].

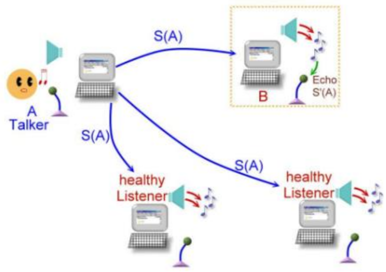

Fig.2: Multi-person network conversation when the echo cancellation mechanism is normal.

The occurrence rate of echo will grow proportionally when a large number of users join in the conference, even when the failure rate of the echo cancellation mechanism of each participating device is quite low. When n people engage in a discussion and the echo cancellation mechanism of a single device sound card fails with probability x, the likelihood that there will be no echo is (1-x)n.

Among the most common uses is Acoustic Echo Cancellation (AEC). The adaptive algorithms aim to determine the acoustic echo route (i.e., the room acoustic impulse response) between a loudspeaker and a microphone placed in an acoustic enclosure and to produce an estimate of the acoustic echo that is subtracted from the microphone's output. The majority of adaptive echo cancellation methods that have been published in the literature operate under the presumption that the echo route is linear. Acoustic echo is the consequence of an audio signal reverberating in an actual setting, producing images of the original sound that are attenuated and delayed in time.

The majority of modern telecommunications networks use full duplex mode. This involves linked acoustic input and output devices that operate simultaneously in full duplex mode. In full duplex mode, the equipment thus functions as both a transmitter and a receiver. In this case, the audio signal is emitted from Far End Speech (FES) through the telephone loudspeaker (audio source). It then reverberates through the surrounding environment and is caught up by the system microphone (audio sink) at Near End Speech (NES). The signal is reflected back to the system through the microphone input after reverberating across the surrounding area. The original signal's time-delayed pictures are contained in these reverberated signals and are subsequently sent back to the original sender.

Acoustic echo during speech transmission results in signal interference and decreased communication quality. An acoustic echo canceller (AEC) is used in contemporary full-duplex communication systems to stop the echo from being sent back to the channel. In essence, the AEC calculates the echo and deducts it from the signal coming from the microphone. Adaptive filtering is the technique used to eliminate the echo.

Adaptive filters are dynamic filters that modify their properties repeatedly to reach the best possible result. In order to minimize a function of the difference between the desired and actual outputs, an adaptive filter modifies its parameters algorithmically. The adaptive algorithm's cost function is the name given to this function.

2. Acoustic echo canceller structure

“Fig. 3”Structure of the new acoustic echo canceller designed for this article. Decoder Accept the bit stream from the network and decode it into the remote reference signal x (n). x (n) produces actual echo y (n) through the echo path H, which is the same as the near-end speech v(n) is superimposed to become the near-end desired signal d(n). Adaptive filter Continuously adjust the coefficient w(n) to produce the estimated echo path h(n). Will The estimated echo y(n) is subtracted from the near-end signal d (n) to obtain the error signal e(n) and sent to the remote end. At the same time, it can be directly The short-term excitation signal is then extracted and used to remove the error signal. The associated linear prediction coefficien. The short-term excitation signal u(n) is as decorrelated signal of x(n), $e_f(n)$ as the decorrelated signal of e(n), divided into Special inputs to the NLMS control and step control blocks of the adaptive filter And in the double-ended call detector. The step control module adaptively controlsStep size factor, the NLMS controller controls the filtering process of the adaptive filter Cheng. Double talk detector determines the existence of Double talk situation, Controls the updating of adaptive filter coefficients and step size [3].

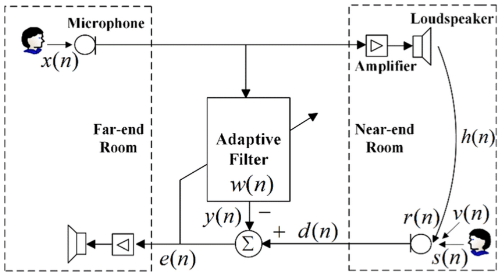

Fig.3: Acoustic echo canceller structure**.**

3. Adaptive echo cancellation Filtering

The adaptive filter in Acoustic Echo Cancellation (AEC) is primarily responsible for adjusting the filter tap weight to overcome the echo issue. For this, a variety of algorithms may be applied, including the following: Least Mean. Affine Projection Algorithm (APA), Recursive Least Square (RLS), Normalized Least Mean Square (NLMS), and Least Square (LMS) [4]. A popular method for adaptive applications like echo cancellation and channel equalization is the LMS. In comparison to the NLMS and RLS algorithms, this one is the easiest to understand. It has been noted that the NLMS algorithm is the preferred option for the real-time acoustic echo cancellation system when taking into account the attenuation values and the processing complexity for each approach. Moreover, stability is ensured without requiring previous knowledge of the signal levels [4].

4. Adaptive echo cancellation filtering

The purpose of a filter is to enhance or extract the desired information from a signal. The most crucial part of an acoustic echo canceller is the adaptive filter, which is also the main component that cancels out sound waves. In order to get a duplicate of the echo signal, it completes the task of determining the room's echo route [5].

An adaptive filter is one that has a corresponding adaptive algorithm for filter coefficient updates, enabling the filter to function in an unidentified and dynamic environment. The Wiener Filter—which is referred to as optimal in the mean square sense—is the resultant filtering issue solution if the filter's inputs are stationary. However, it needs knowledge about the statistics of the data to be processed beforehand. Another effective technique, if the environment is unknown, is to employ a recursive algorithm with an adaptive filter.By modifying filter coefficients, also known as tap weights, in accordance with the signal circumstances and performance standards, also known as quality evaluation, the adaptive algorithm establishes the filter features.

An error signal, or the difference between the filter output signal and a specified reference (or desirable) signal, is the basis for a standard performance criteria. An adaptive algorithm determines and updates the coefficients of a digital filter, which is known as an adaptive filter. One of the fundamental techniques of digital signal processing is adaptive filtering, which has many uses in both business and science. Many different applications, including as adaptive beamforming, adaptive equalization, adaptive noise cancellation, and echo cancellation, employ adaptive filtering techniques[5].

"Fig. 4", shows the block diagram for the adaptive filter method utilized in this paper.

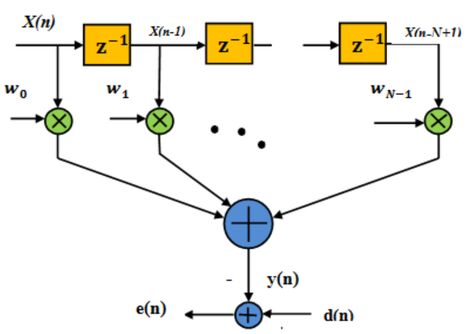

Fig. 4: Adaptive filter block diagram [5].

The coefficients of the FIR filter tap weight vector are denoted by w, the input vector samples are represented by x(n), the delay is denoted by Z-1, the adaptive filter output is denoted by y(n), the intended echoed signal is represented by d(n), and the estimation error at time n is represented by e(n).

Finding the difference between the intended signal and the adaptive filter output, e(n), is the goal of an adaptive filter. The adaptive filter receives this erroneous signal back and uses an algorithm to modify its coefficients in order to minimize the cost function, which is a function of this difference. The undesirable echoed signal is equivalent to the optimum output of the adaptive filter in the case of acoustic echo cancellation. The error signal decreases to zero when the output of the adaptive filter equals the target signal. Under these circumstances, the distant user would not hear any of their original voice being back to them, and the echoed signal would be fully canceled [6].

5. Implementation of the NLMS algorithm

The Normalized Least Mean Square (NLMS) method is a frequently utilized subclass of least mean square algorithms. These algorithms are improvements on the conventional LMS algorithms, with the goal of accelerating convergence and enhancing algorithm stability. Their emergence is due to the LMS adaptive filters' reliance on the step size µ. The applications may be significantly impacted by the step size trade-offs. Raising µ results in a loss in stability since it raises both the speed and the asymptotic error. Because of this, a scaling factor for µ is added to the equation to scale the step size continuously, improving the stability and performance of the algorithm in the process.

A modified NLMS algorithm is employed in this AEC system. The NLMS algorithm's practical implementation is somewhat similar to the LMS algorithm's as it is an extension of the latter. These actions must be taken in the following order for every iteration of the NLMS algorithm.

1. The output of the adaptive filter is calculated.

$$y(n)=\sum_{i=0}^{N-1} w(n)x(n-1)=w^T(n)x(n) \quad (1)$$

2. An error signal is calculated as the difference between the desired signal and the filter output.

e(n)=d(n)-y(n) (2)

3. The step size value for the input vector is calculate

$$\mu(n)=\frac{\mu}{\rho+\|\mathbf{x}(n)\|^2} \quad (3)$$

where ρ is a small positive constant used to insure that if $\|\mathbf{x}(n)\|^2$ is zero or close to it, the instability due to division by zero is avoided.

4. The filter tap weights are updated in preparation for the next iteration.

$$\mathbf{w}(n+1)=\mathbf{w}(n)+\frac{\mu}{\rho+\|\mathbf{x}(n)\|^2}\mathbf{x}(n)e(n) \quad (4)$$

The NLMS algorithm takes 3N+1 multiplications for each iteration, which is only N more than the regular LMS algorithm. Taking into account the improvements in stability and echo attenuation, this increase is reasonable [6].

6. Simulation Results for Echo Cancellation Using NLMS Algorithm

The normalized LMS approach was simulated using MATLAB. "Fig. 5" shows the adaptive filter output of the normalized least mean square (NLMS) approach. That is the estimated value of the echo signal.The adaptive filter output separates the echo signal from the input signal with an order filter length of 1000. For every iteration, the NLMS approach requires 3N additions and 3N+1 multiplications.

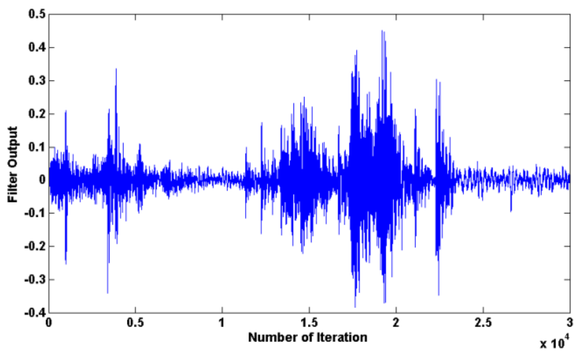

Fig. 5: The adaptive filter output of NLMS algorithm.

The error signal vs the number of repetitions using the NLMS method is shown in "Fig. 6". (Count the extent to which the method reduces echo).

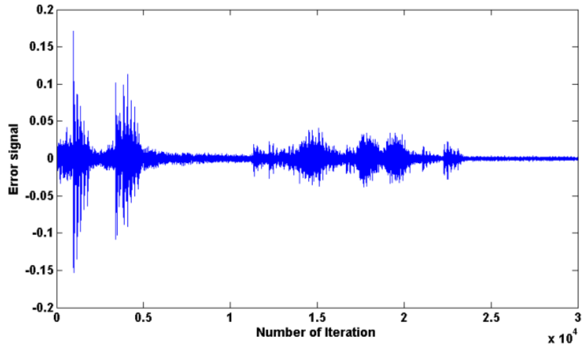

Fig.6: The error signal versus number of NLMS algorithm.

"Fig. 6" and "Fig. 7" shows estimation error and mean square error signal of the NLMS algorithm which is to calculate the difference between desired signal and the output of the filter, it is shown from Fig.7 that. The mean square error decreases as the number of iteration increases, and equal to 0.0044. As the adaptive filter output is equal to desired signal the error signal tends to zero. In this situation the echo signal would be completely cancelled and the far user would not hear any of their original speech returned to them.

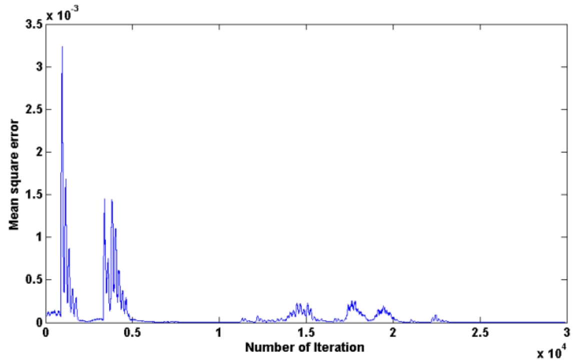

Fig. 7: Mean Square Error of NLMS algorithm.

The measure of the echo canceller efficiency is expressed by Echo Return Loss Enhancement (ERLE) factor which is defined as:

$$ERLE=\frac{P_d(n)}{P_e(n)} \quad (5)$$

$$ERLE=\frac{E|d^2(n)|}{E|e^2(n)|} \quad (6)$$

Where ERLE is measured in dB, $d^2(n)$ is the microphone signal's power and $e^2(n)$ is the residual error signal's power."Fig. 8". shows ERLE performance with using NLMS algorithm.

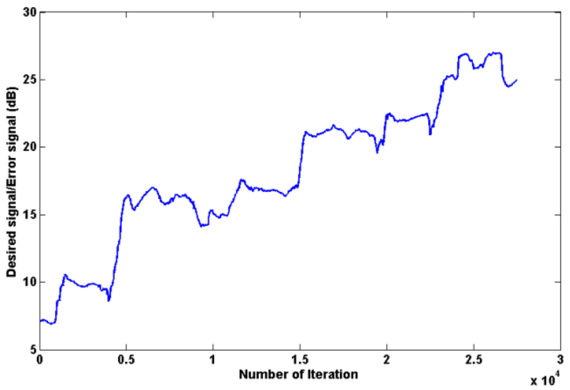

Fig. 8: Echo Return loss Enhancement (ERLE) Performance with NLMS.

Following are the conclusions drawn from the numerical and simulation results: The LMS with a variable step size needs 2N+1 multiplications and 2N additions for real-valued data each iteration. Because of its simplicity, the NLMS is quick to operate in real time and inexpensive to construct computationally. Additionally, it is numerically stable and resilient. However, because it depends on changes in the step size and strength of the input signal, its convergence speed is sluggish.

the algorithm known as NLMS While every LTI system represented as a FIR filter is stable in the sense of BIBO (bounded input bounded output) (since its transfer function does not contain poles within the unit circle), it might become unstable in accordance with certain principles if its coefficients are changing dynamically. Therefore, rather than the filter type, the instability can be brought on by changes in the filter coefficient. The robust technique which uses the input signal's Eigen-decomposition is mentioned as a solution to this issue. These methods use the Eigen-decomposition of the input signal to update stepsize and determine the new weights of a taped filter. The next part presents the Eigen-decomposition performance evaluation of the Robust Adaptive filter.

7. Robust Adaptive filter using Eign Diagonal Loading.

The tap weights of the FIR vector are updated in using equation. This section offers a diagonal loading Robust Adaptive filter that successfully overcomes the divergence mean square error on the impact of the adaptive filter. The classic diagonal loading approach involves repetitive computations based on the ideal weight vector of the NLMS algorithm from the previous chapter. Equation (5)'s optimization problem is solved by referring back to the input signal's equation of equality before updating the stepsize. Equation is used to update the FIR vector's tap weights.

$$W(n+1) = W(n) + \mu(n)x(n)e(n) \qquad (7)$$

The Eigndecompsion technique is used to enhance the NLMS algorithm and obtain the Load stepsize, which is based on the sampling weight algorithm, while loading discretion. To lessen the signal divergence of errors, this loading approach adds a fixed value diagonally from the input signal's correlation matrix to update trhe stepsize. This yields the robust correlation matrix.

$$R_{xx} = E[x(n) * x(n)^H] \qquad (8)$$

We know the R_{XX} decompose to V and U are left-singular and right-singular Eign-vectors respectively also S is Eign-valusdiagonal matrix.

$$S = eig(R_{xx}) \qquad (9)$$

Where the S is diagonal of R_{xx} Eigen-values

$$S = \begin{bmatrix} \lambda_1 & 0 & 0 & 0 & 0 \\ 0 & \lambda_2 & 0 & 0 & 0 \\ 0 & 0 & \lambda_3 & 0 & 0 \\ 0 & 0 & 0 & \lambda_4 & 0 \\ 0 & 0 & 0 & 0 & \lambda_n \end{bmatrix} \qquad (10)$$

Where β is a max positive diagonal loading factor, available update stepsize [8].

$$\mu(n) = \beta + \frac{\beta}{\sigma + \|\mathbf{x}(n)\|^2} \qquad (11)$$

Where σ is safety factor (small positive constant) introduced to prevent division by zero if $\|\mathbf{x}(n)\|^2$ is very small.Substituting (1) into (3).

$$W(n+1) = W(n) + \beta + \frac{\beta}{\sigma + \|\mathbf{x}(n)\|^2} x(n)e(n) \quad (12)$$

Where w(n) is the weight vector, x(n) is the input signal.Equation (4) is the weight vector updating equation for NLMS algorithm.

8. Simulation Results and Comparison of modified NLMS and NLMS algorithm.

The modified update weight has been examined for modified NLMS, the same assumption in simulation of conventional NLMS. “Fig. 10”shows the adaptive filter output of NLMS and modified NLMS algorithm is an estimate of the echo signal.

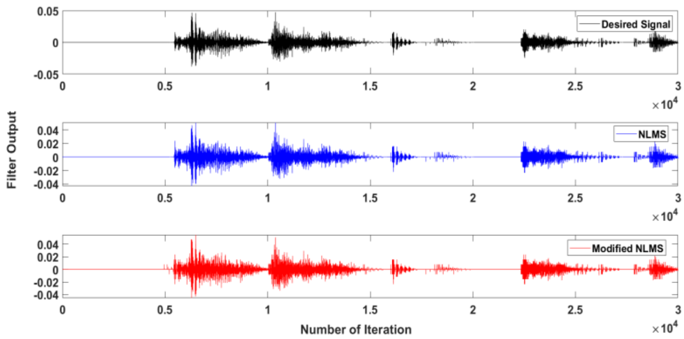

Fig. 9: The adaptive filter output of NLMS algorithm.

From “Fig. 10”, we see that there is significant improvement for all mean square error and with robust and more stability than conventional LMS . In case modified NLMS, From Fig. 9, and “Fig. 10”, it is clear that technique of modified have better and converge faster than the traditional NLMS technique.

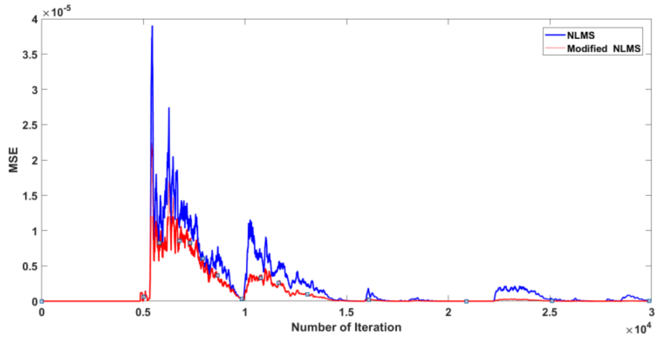

Fig. 10: Comparison of the convergence for modified NLMS and NLMS algorithm.

“Fig. 10”, shows the error versus the number of iterations using adaptive filtering conventional and modified NLMS, algorithm (measure how much the algorithm minimize the echo), it is clear that technique of modified have better.

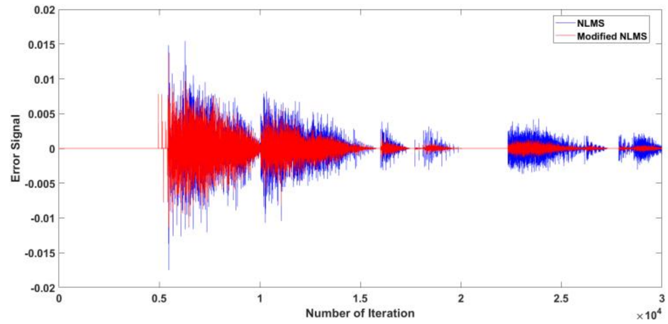

Fig. 11: Comparison of the error for modified NLMS and NLMS algorithm.

The ERLE curve for the classic and modified NLMS algorithms is shown in “Fig. 12”. As can be seen in this image, at the end of the convergence phase, the NVSSLMS method outperformed all other algorithms in terms of echo suppression, as measured by its higher ERLE value. When utilizing the suggested technique in place of the conventional NLMS algorithm, the ERLE boost is around 11 dB.

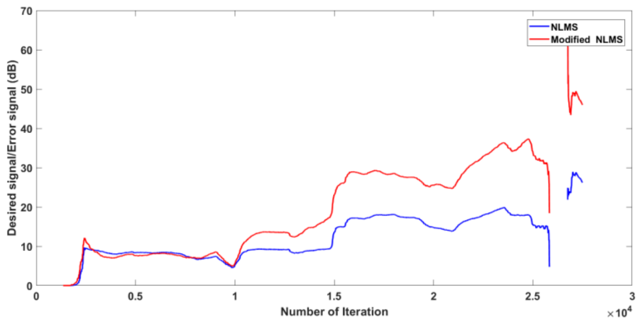

Fig. 12: Comparison of the (ERLE) for modified NLMS and NLMS algorithm.

If there is a difference, the modified NLMS is better; that is, it increases more quickly and has an attenuation that is greater than the standard NLMS. High attenuation, quick convergence, and real-time operation should all be offered by the adaptive method. Because of its high attenuation, the modified NLMS algorithm performs better when used for echo cancellation applications.

9. Conclusion

The studied adaptive filter and modified worked successfully for Acoustic Echo Cancellation of VOIP and here are some conclusions of the analysis for the obtained results.

The tap weights of the FIR vector are updated in using simple equations. Because of its excellent attenuation and straightforward complexity, the NLMS is the easiest algorithm to apply in real time. As a result, it is a superior method for echo canceling applications. Simulation results support this claim. If step size μ is too high, it might cause instability in the NLMS and provide incorrect results. However, for large step sizes, a somewhat big value of promotes quicker convergence. to converge substantially more quickly. Given the high FIR order needed for echo cancellation, this efficiency comes at the expense of computational complexity, making real-time implementation impracticable.

The resilient solutions are based on the inverse transformation of the input signal, or diagonal loading, in order to leverage update step size to overcome this issue. The mean square error is nearly identical to that of the (fixed step size least mean square) modified NLMS algorithm under similar conditions. The performance evaluation of the robust adaptive filter using diagonal loading offers faster speed of convergence among conventional NLMS with variable step size while maintaining the same small level of maladjustment, more robust stability of the LMS algorithm, for example when take the number of iteration = 2500 the Modified NLMS attenuation is better than traditional NLMS attenuation by 11 dB .exhibits a better balance between simplicity and performance than the pure conventional NLMS algorithm. Due to its good properties it will been used in real-time applications.

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