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A COMPARISION OF PERFORMANCE OF MVSSA WITH OTHER CONVENTIONAL ADAPTIVE ALGORITHM FOR ECHO CANCELLATION

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ABSTRACT

Acoustic Echo Cancellation is a common occurrence in todays telecommunication systems. The signal interference caused by acoustic echo is distracting to users and causes a reduction in the quality of the communication. Echo cancellers are very successful and today almost no echo at all can be perceived while using telephones. The Normalized Least Mean Square Algorithm (NLMS) is always the favorable choice because of fast convergence speed. NLMS has higher ERLE so it works as a better algorithm for Echo Cancellation application. The performance of NLMS is better as compared to that of LMS, VSS LMS[1]. The present work focus on the conventional adaptive algorithms and Modified Variable Step Size LMS algorithm to reduce the unwanted echo. A Variable Step Size Least Mean Square (VSS LMS) Algorithm is given with significant changes in which scalar step size increases or decreases as the squared error increases or decreases, thereby allowing the adaptive filter to track changes in the system and produces a smaller steady state error. A new VSS LMS algorithm is proposed, which can effectively adjust the step size while maintain the immunity against independent noise disturbance. The Modified VSS LMS algorithm allows more flexible control of misadjustment and convergence time without the need to compromise one for the other. Simulation results are presented to support the analysis and to compare the performance of the modified algorithm with the other conventional adaptive algorithms. They show that the MVSS algorithm provides faster speed of convergence among other Variable Step Size algorithms while retaining the same small level of misadjustment and the mean square error[2]. An attempt has been made to examine adaptive filtering techniques as they applied to acoustic echo cancellation, to simulate these adaptive filtering algorithms using MATLAB and to compare the performance of these adaptive filtering algorithms as they applied to the acoustic echo cancellation application.

Index Terms: Acoustic Echo cancellation (AEC), LMS, NLMS, VSS LMS.

1. INTRODUCTION

Echo cancellation is one of the most widely used digital signal processing devices in the world because each telephone call requires a pair of echo cancellers. Basically, a transversal filter, which is adaptively modeling the echo path impulse responses, generates an estimate of the echo, with this an echo estimate is created at the right time to cancel the actual echo[3].

The common problems faced by echo cancellation are the convergence time and the degree of cancellation. Convergence time is the time taken to reach an acceptable level of steady state residual echo. Degree of cancellation is the amount of echo cancelled, measured in Echo Return Loss Enhancement (ERLE)[4]. The LMS is most widely used algorithm for numerous applications, especially channel equalization and echo cancellation. The LMS is introduced as a way to recursively adjust the parameters of w(n) of a linear filter with the goal of minimizing the error between a given desired signal and the output of the linear filter[5]. The NLMS is always a favorable choice because of convergence speed. The estimation error and mean square error are very small and the value of average ERLE is also quite large. NLMS is the best algorithm for the practical implementation of the echo canceller. NLMS has higher ERLE so it works as a better algorithm for echo cancellation application[1]. The performance of VSS algorithm deteriorates in the presence of measurement noise. Hence a new VSS LMS algorithm is proposed, where the step size of the algorithm is adjusted according to the square of the time- averaged estimate of the autocorrelation of e(n) and e(n-1). The MVSS LMS algorithm allows more flexible control of misadjustment and convergence time without the need to compromise one for the other[1]. The performance of the MVSS LMS algorithm is better that the VSS algorithm when applied to echo cancellation application.

2.Modified Variable Step Size Algorithm

There is a vast amount of literature on variable step size methods, which exploit the trade-off between fast convergence and lower steady state error. A number of time-varying step size algorithms have been proposed to enhance the performance of the conventional LMS algorithm. Several criteria have been used: squared instantaneous error, sign changes of successive samples of the gradient, attempting to reduce the squared error at each instant, or cross correlation of input and error. The modified VSS LMS algorithm gives a better performance as compared to other variable step size algorithms.

2.1 Variable Step Size LMS Algorithm

Based on the error-squared power, Kwong and Johnston proposed a simpler Variable Step Size Least Mean Square Algorithm[1]. The error power reflects the convergence state of the adaptive filter, where a converging system has a higher error power while the converged system has a smaller error power. Therefore, scalar step size increases or decreases as the squared error increases or decreases, thereby allowing the adaptive filter to track changes in the system and produces a smaller steady steady state error. To ensure stability, the variable step size m(n) is constrained to the pre-determined maximum and minimum step size values of the LMS algorithm, while a and b are the parameters controlling the recursion. Simply, the VSS algorithm's step size value change by tracking the error square or the error power. A large error increases the step size to provide faster tracking while a small error reduces the step size for smaller steady state error. Although this approach can improve the step size trade-off effect, the drawback is that the maximum and minimum step sizes would require to be known a priori. The variable step size algorithms (except for the gradient adaptive step size) are based on some heuristic rules on the step size adjustment which are translated into numerical formulae. An overall weakness of the variable step size algorithms are that they require the user to select additional step size recursion constants and an initial step size to control the adaptive behavior of the step size sequence. More importantly, anticipation of the maximum and minimum limits are needed to avoid instability as well as to maximize performance. Nevertheless, the body of work in this field has enabled adaptive performance to be achieved that is comparable to the RLS algorithm. However, the performances gained are always followed by an increase in complexity and additional parameters to manage. The gear shifting approach and the VSS algorithm are the most simple and effective algorithms for fast convergence. Alternatively, to increase the convergence speed, the normalized LMS algorithm is the natural choice, as it is independent of the parameter selection state error.

2.2 Modified Variable Step Size Algorithm

A number of time-varying step-size algorithms have been proposed to enhance the performance of the conventional LMS algorithm. Experimentation with these algorithms indicates that their performance is highly sensitive to the noise disturbance[2]. The present work discusses a robust variable step-size LMS-type algorithm providing fast convergence at early stages of adaptation while ensuring small final misadjustment. The performance of the algorithm is not affected by existing uncorrelated noise disturbances. Simulation results comparing the proposed algorithm to current variable step-size algorithm clearly indicate its superior performance. Since its introduction, the LMS algorithm has been the focus of much study due to its simplicity and robustness, leading to its implementation in many applications. It is well known that the final excess Mean Square Error (MSE) is directly proportional to the adaptation step size of the LMS while the convergence time increases as the step size decreases. This inherent limitation of the LMS necessitates a compromise between the opposing fundamental requirements of fast convergence rate and small misadjustment demanded in most adaptive filtering applications. As a result, researchers have constantly looked for alternative means to improve its performance. One popular approach is to employ a time varying step size in the standard LMS weight update recursion. This is based on using large step-size values when the algorithm is far from the optimal solution, thus speeding up the convergence rate. When the algorithm is near the optimum, small step-size values are used to achieve a low level of misadjustment, thus achieving better overall performance. This can be obtained by adjusting the step-size value in accordance with some criterion that can provide an approximate measure of the adaptation Process State. Several criteria have been used:

Squared instantaneous error [1]

Sign changes of successive samples of the gradient [7]

Attempting to reduce the squared error at each instant [8]

Cross correlation of input and error [9].

Experimental results show that the performance of existing variable step size (VSS) algorithms is quite sensitive to the noise disturbance. Their advantageous performance over the LMS algorithm is generally attained only in a high signal-to-noise environment. This is intuitively obvious by noting that the criteria controlling the step-size update of these algorithms are directly obtained from the instantaneous error that is contaminated by the disturbance noise. Since measurement noise is a reality in any practical system, the usefulness of any adaptive algorithm is judged by its performance in the presence of this noise. The performance of the VSS algorithm deteriorates in the presence of measurement noise. Hence a new VSS LMS algorithm is proposed, where the step size of the algorithm is adjusted according to the square of the time-averaged estimate of the autocorrelation of e(n) and e(n-1) . As a result, the algorithm can effectively adjust the step size as in [1] while maintaining the immunity against independent noise disturbance. The MVSS LMS algorithm allows more flexible control of misadjustment and convergence time without the need to compromise one for the other.

3. PERFORMANCE ANALYSIS

As given in the paper [10], for the purpose of comparison, the VSS and the MVSS algorithms were used in an echo cancellation configuration. The echo path taken in the paper was simulated using the echo path impulse response of **CSA** loop ***#2*** from the set of HDSL2 test loops, at the central office end. The echo path impulse response was truncated to 124 *ps* to avoid the residual echo inherent in untruncated impulse responses. The sampling interval was set at **1** ps, which resulted in an impulse response vector having a length of 124. The echo path was excited with a zero-mean, white Gaussian signal to generate the echo. The echo signal thus generated was contaminated with the received signal, which was also a zero mean white Gaussian signal to result in an SNR of 0 dB. The two algorithms were simulated with the following parameter values:

a=0.97, g =0.08, b= 0.999, mmax=0.01, and mmin=0.001

To measure the performance of the two algorithms, the value of the misadjustment ½½ w***** - w(n)½½ where w***** is the echo path impulse response vector and w(n) is the estimated coefficient vector of the LMS algorithm at the nth iteration, was calculated and plotted against the number of iterations in fig.3.1. As can be observed, the proposed algorithm yields a lower steady-state misadjustment.

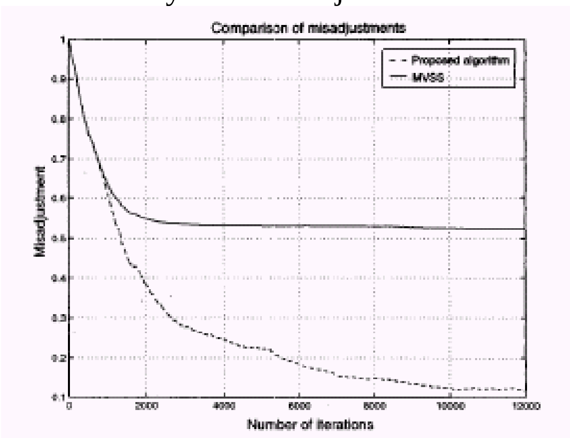

Fig 3.1: Comparison of the Misadjustments Using the MVSS and proposed VSS Algorithms[10]

4. CONCLUSION

Adaptive Digital Signal Processing is a specialized branch of DSP, dealing with adaptive filters and system design. There are number of adaptive algorithms available in literature and every algorithm has its own properties, but aim of every algorithm is to achieve minimum mean square error at a higher rate of convergence with lesser complexity. The various results obtained and compare the performance of various algorithms used for Echo Cancellation. The values of average ERLE obtained from the plots for various adaptive algorithms shows that the average ERLE is maximum for RLS algorithm. The estimation error and the mean square error are several orders smaller for RLS algorithm. The convergence rate of NLMS algorithm is greater than the LMS algorithm and the RLS algorithm has a far greater convergence rate compared to the LMS algorithm. Though the RLS algorithm gives much better results compared to other algorithms, still it is not used, as each iteration requires 4N2 multiplications. For echo cancellation systems the FIR filter order is usually in the thousands. Thus the number of multiplications required are very large because of which the RLS algorithm is too costly to implement. In practice the LMS based algorithms, although poorer performers, are preferred. Also the results obtained for Modified VSS-LMS algorithm are almost similar to the LMS algorithm but the Modified VSS-LMS algorithm can effectively adjust the step size while maintaining the immunity against independent noise disturbance. Also the MVSS LMS algorithm allows more flexible control of misadjustment and convergence time without the need to compromise one for the other. Therefore, the overall analysis shows that the MVSS-LMS algorithm supersedes VSS-LMS adaptive algorithm under similar conditions, for different applications as far as misadjustment and rate of convergence is concerned.

5. FUTURE SCOPE

The various signal processing applications demand for reduction in trade off between misadjustment and convergence rate, taking realization of algorithm into account. There is a scope of improvement in existing algorithms, which reduces complexity and fulfills stringent conditions for stability. This thesis dealt with transversal FIR adaptive filters; this is only one of many methods of digital filtering. Other techniques such as infinite impulse response (IIR) or lattice filtering may prove to be effective in an echo cancellation application. The real time echo cancellation system can be implemented successfully using the TI TMSC6711 DSK. However, this system can only cancel those echoes that fall within
the length of the adaptive FIR filter.

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