

ACTIVE NOISE CONTROL SYSTEMS USING VARIABLE STEP SIZE FXLMS ALGORITHM

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Abstract-*The most popular adaptation algorithm used for Active Noise Control (ANC) applications is the Filtered-x Least Mean Square (FxLMS) algorithm. This algorithm is mostly preferred, because it used as controller in adaptation filter to update the filter coefficients. In this paper, FxLMS algorithm with variable step size to achieve noise control has been proposed. This new algorithm will improves the convergence as well as reduction in noise compared with conventional FxLMS. The convergence and noise reduction rate are analyzed with the help of MATLAB.*

Keywords: Active Noise Control, FxLMS algorithm, Noise reduction, Variable step size.

I.INTRODUCTION

Acoustic noise reduction is a major real-life application for headphones, automobiles, mobile phones, and some diligences. ANC system using adaptive filters has smaller volume and it acts as best alternatives to those using passive methods. ANC is a technique, in which the unwanted noise is cancelled based on the principal of superposition. The controller of the ANC system generates an anti-noise of equal amplitude and opposite phase and it is combined with the unwanted noise, thus resulting in the cancellation of both noises. The advantage of the ANC system is it efficiently attenuates the noise in low-frequency and also inexpensive, less weight and easy to implement, compared with passive techniques. Further structure of the ANC system can be changed according to the environmental condition, which is the major bottle neck in the passive techniques.

The acoustic noise sources are non-stationary and whose time, frequency, phase, amplitude and velocity of the sound is varied in accordance with the environment. Hence the ANC systems have to adaptive with respect to these variations. The coefficients of the adaptive filters must varies according to the noise level in such a way that to minimize the error signal. The adaptive filter can be realized as finite impulse response, infinite impulse response, lattice and transform-domain filters. The filtered-x least mean square algorithm is the most commonly used algorithm in ANC for its robust, low computational complexity, any easy to implement [1].

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Now a day, various approaches have been introduced for active noise control systems. Since FxLMS algorithm appears to be the best choice for ANC applications, which is a modified version of the Least Mean Square (LMS) algorithm. The LMS algorithm is still used due to its simplicity and robustness [5]. FxLMS algorithm can be beneficial in expressions of faster convergence process in the actual ANC because a pre-filtered output signal through the secondary path is used [6]. FxLMS algorithm is the most popular adaptive algorithm to update the controllers.

Several other approaches like controlling noise using logarithmic transformation which leads to complexity in computation and hybrid time taking approaches were made in order to reduce the noise in acoustic environments.

This paper is organized as follows. The Section 2 briefs about the basic operation of FxLMS algorithm. The Section 3 describes about Variable Step Size algorithms. The Section 4 explains the VSS FxLMS algorithm and Section 5 describes the results and discussion and Section 6 gives the conclusion.

II. FxLMS ALGORITHM

The FxLMS algorithm is widely used adaptation algorithm used for ANC applications, which is an extension version of the LMS algorithm [10]. The block diagram for a single-channel feed forward ANC system using the FxLMS algorithm is shown in Fig.1.

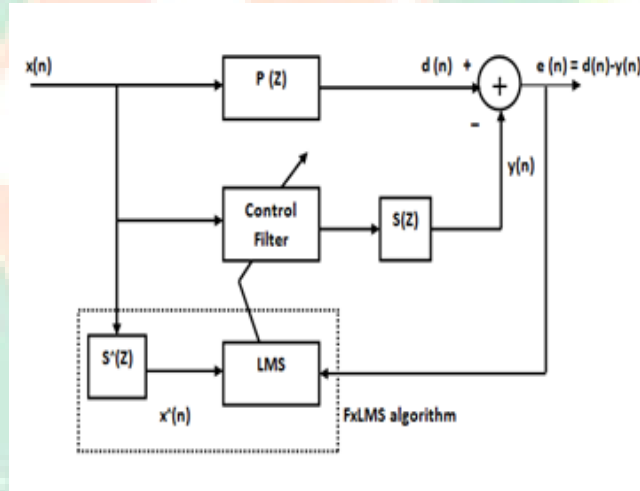


Fig.1. Block diagram of FxLMS based feedforward ANC system.

Here, $P(z)$ is primary acoustic path between the reference noise source and the error microphone and $S(z)$ is the secondary path following the ANC filter $w(z)$.

The reference signal $x(n)$ is filtered through $S'(z)$, and appears as anti-noise signal $y'(n)$ at the error microphone. This anti-noise signal combines with the primary noise signal $d(n)$ to create a zone of silence in the vicinity of the error microphone. The error microphone measures the residual noise $e(n)$, which is used by $w(z)$, for its adaptation to minimize the sound pressure at error microphone.

Here $\hat{S}(z)$ account for the model of the secondary path $S(z)$ between the output of the controller and the output of the error microphone. The filtering of the reference signals $x(n)$ through the secondary-path model $\hat{S}(z)$ is demanded by the fact that the output $y(n)$ of the adaptive controller $w(z)$ is filtered through the secondary path $S(z)$. The expression for the residual error $e(n)$ is given as,

$$e(n) = d(n) - y'(n) \quad (1)$$

Where $y'(n)$ is the controller output $y(n)$ filtered through the secondary path $S(z)$. The $y'(n)$ and $y(n)$ computed as,

$$y'(n) = S^T(n)y(n) \quad (2)$$

$$y(n) = w^T(n)x(n) \quad (3)$$

Where, $w(n) = [w_0(n), w_1(n), \dots, w_{L-1}(n)]^T$ is tap weight vector, $x(n) = [x(n), x(n-1), \dots, x(n-L+1)]^T$ is the reference signal picked by the reference microphone and $S(n)$ is impulse response of secondary path $S(z)$. It is assumed that there is no acoustic feedback from secondary loudspeaker to reference microphone. The FxLMS update equation for the coefficients of $w(z)$ is given as,

$$w(n+1) = w(n) + \mu e(n)x'(n) \quad (4)$$

Where, μ is the step size. It determined the convergence rate [7] and $x'(n)$ is given by,

$$x'(n) = S^{\wedge T}(z)x(n) \quad (5)$$

Where $x'(n)$ is reference signal $x(n)$ filtered through secondary path model $S^{\wedge}(z)$.

III. VARIABLE STEP SIZE ALGORITHM

There is a tradeoff between the rate of convergence, the amount of excess mean-square error, and the ability of the filter to track signal as their statistics change while selecting the step size μ in FxLMS algorithm. The selection of the step size μ in FxLMS algorithm is the important phenomenon in ANC system. Since there is always a compromise between the convergence rate, the excess amount of mean-square error, and also the capability of the filter to track signal in according to the change in signal statistics. At the beginning the filter coefficient of the adaptive filter w is far away from the optimal solution. Hence the system uses large step size μ in order to adapt the weight vector rapidly toward the desired solution.

As the filter starts to converge towards desired steady-state solution, then the algorithm should reduce the excess mean square error at the error microphone, by simply decreasing the step size. This method introduces the possibility of using a variable step size (VSS) in the FxLMS adaptive filter [8]-[13]. However the practical difficulty is in defining a set of rules for changing the step size in such a way that the adaptive filter has a yield small excess mean square error while, at the same time, maintaining the ability of the filter to respond quickly to changes in the signal statistics.

The method of changing the step size in according to the environmental condition is referred as VSS FxLMS algorithm. The filter coefficient update equation for this algorithm is given in equation (6)

$$w_{n+1}(k) = w_n(k) + \mu_n(k)x'(n-k) \quad (6)$$

Here it has been assumed that both $x'(n)$ and $d(n)$ is real value processes and $\mu_n(k)$ is VSS and it is varied independently for each coefficient w . The rules for adjusting step size are tied to the rate at which the gradient estimate changes sign. With an estimated gradient given in equation (7).

$$\nabla^2 e^2(n) = -2e(n)x(n-k) \quad (7)$$

These rules are based on the premise that if the sign of $e(n)x(n-k)$ is changing frequently, then the coefficient $w_n(k)$ should be close to its optimum value where the gradient is equal to zero. On the other hand, if the sign is not changing very often, then the coefficient $w_n(k)$ is probably not close to its optimum value. Therefore $\mu_n(k)$ is decreased by a constant c_1 , if m_1 successive sign changes are observed in $e(n)x(n-k)$, whereas $\mu_n(k)$ is increased by a constant c_2 , if $e(n)x(n-k)$ has the same sign for m_2 successive updates. Therefore, the step-size $\mu_n(k)$ can be varied between two successive values given in equation (8).

$$\mu_{\min} < \mu_n(k) < \mu_{\max} \quad (8)$$

To ensure that the VSS FxLMS algorithm converges in the mean with only a modest increase in computation, the VSS FxLMS algorithm may result in a considerable improvement in the convergence rate.

The contradiction between the convergence speed and the convergence precision fixed step LMS algorithm can be solved in the variable step LMS algorithm. In the initial stages of adaptive and tracking phase, a larger step size is used in order to have fast convergence speed, when the algorithm is in the steady state; smaller step is used for a small steady-state error.

Some approximation is used as a measure to control step size in adaptive processes. Simple and effective method is to use the adaptive error signal in the process, trying to establish some kind of function between the step size and the error signal.

Currently, the main variable step size algorithm is to establish the nonlinear relationship between the step size and the error signal to adjust the step. The working principle is: the error is large in the initial iteration stage along with a larger step size to speed up the convergence rate; when the error is close to zero, a smaller step is accessed to achieve smaller stable-state error.

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IV. PROPOSED METHOD

In this paper, algorithm has been proposed is the modified version of the FxLMS algorithm. The LMS algorithm has been used extensively for the adaptive algorithm of the ANC system due to the easy computation and simplicity of the algorithm. The LMS algorithm actively estimates the coefficients of the adaptive filter using the steepest descent method. This FxLMS algorithm is given by,

$$w(n+1) = w(n) + \mu e(n)x'(n) \quad (9)$$

where w is the coefficient vector in the FIR filter, $e(n)$ is the error signal, and μ is the step size that determines the convergence speed and stability of adaptation. If the step size μ is high, the convergence rate is fast, but the error is not minimized. The optimized step size provides a fast convergence with stability. To improve the convergence performance of the FxLMS algorithm, the variable step size FxLMS algorithm (VSSFxLMS) was developed. The VSSFxLMS is expressed as follows:

$$\mu(n+1) = \alpha\mu(n) + \beta e^2(n) \quad (10)$$

$$w(n+1) = w(n) + \mu(n+1)e(n)x'(n) \quad (11)$$

$$\mu(n) = \begin{cases} \mu_{max}, \mu(n) > \mu_{max} \\ \mu_{min}, \mu(n) < \mu_{min} \\ \mu(n), \text{else} \end{cases} \quad (12)$$

where α and β are the constants and major parameters in the VSSFxLMS algorithm. The step size μ is limited between μ_{max} and μ_{min} to provide stability to the algorithm. The equations (10) are called as proposed variable step size (VSS) parameter. The equations (11) are called as proposed variable step size FxLMS (VSSFxLMS) algorithm.

V. RESULTS AND DISCUSSION

This section represents the performance of the proposed VSSFxLMS algorithm with variable step size is demonstrated compared with conventional FxLMS algorithm with fixed step size. It is observed that proposed algorithm will provides better noise reduction compared with conventional algorithm about 20dB. The Fig.2. represents the noise level characteristics. From the characteristics it can be known that the noise level reduced by using variable step size. The noise level for FxLMS algorithm with fixed step size of 0.044 is 116.23dB. By using variable step size the noise level can be reduced to 89dB. Number of iterations used here is 1000. The Fig.3.represents the convergence characteristics.

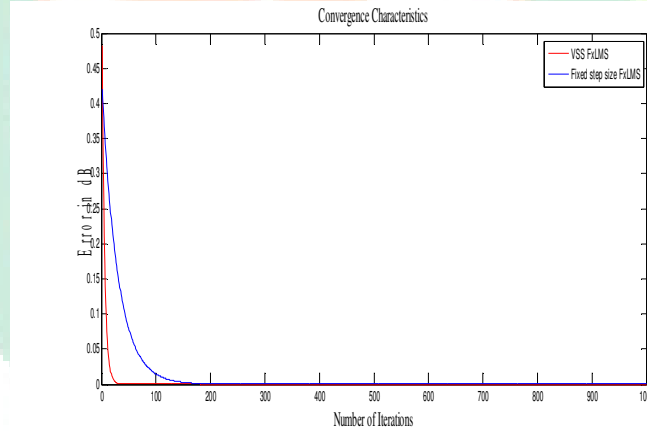


Fig.2. Characteristics of Convergence rate

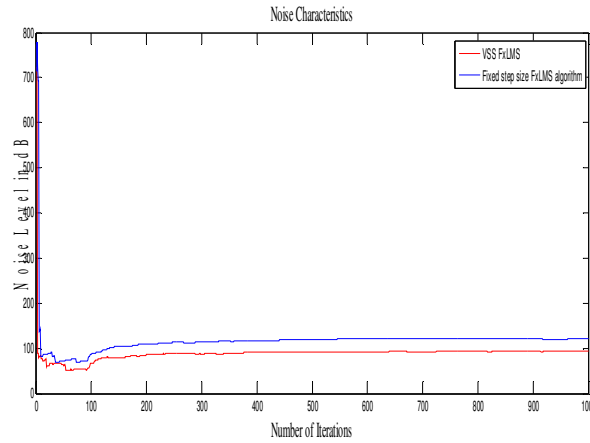


Fig.3. Characteristics of Noise level

VI. CONCLUSION

In this paper, a new variable step size FxLMS algorithm for active noise control has been proposed. The proposed algorithm has better noise reduction when compared with fixed step size FxLMS algorithm. It is inspected using MATLAB simulations. This new algorithm improves the convergence speed as well as reduction in noise.

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