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Variable Step Size for Improving Convergence of FxLMS Algorithm

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Abstract

Several approaches have been introduced for active noise control (ANC) systems. The most popular adaptation algorithm used for Active Noise Control(ANC) applications is the Filtered-x Least Mean Square (FxLMS) algorithm. In this paper, FxLMS algorithm with variable step size to improve the convergence of ANC system has been proposed. This algorithm is mostly preferred, because it used as controller in adaptation filter to update the filter coefficients. This new algorithm is based onarc-tangent function and it will improve the convergence as well as a reduction in noise compared with conventional FxLMS. The convergence and noise reduction rate are analyzed with the help of MATLAB. Simulation results show that the convergence speed and noise reduction of the variable step algorithm are superior to conventional FxLMS algorithm.
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Keywords: Active Noise Control; FxLMS algorithm; Noise Reduction; Convergence analysis; Variable step size.

1. Introduction

ANC has received a great deal of attention in recent years. ANC involves an electroacoustic or electromechanical system that cancels the primary (unwanted) noise based on the principle of superposition. An anti-noise of equal amplitude and opposite phase is generated and combined with the primary noise, thus resulting in the cancellation of

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noises. The ANC system efficiently attenuates low frequency noise where passive methods are either inefficient or tend to be very expensive or bulky. ANC is developing rapidly because it permits improvements in noise control, often with potential benefits in size, weight, volume, and cost. 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]. 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 [2]. 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 vary 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 most common techniques used to update the coefficients of the controller filter is least mean- square (LMS) algorithm. An early duct cancellation system based on adaptive filter theory was developed [3]-[4].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(VSS) algorithms. The Section 4explains the proposed VSS FxLMS algorithm and Section5 describes the results and discussion and Section 6 gives the conclusion.

2. FxLMS Algorithm

The FxLMS algorithm is widely used adaptation algorithm used for ANC applications, which is an extension version of the LMS algorithm [7]-[10]. The block diagram for a single-channel feed forward ANC system using the FxLMS algorithm is shown in Fig.1. 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 $S^{\wedge}(z)$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 $S^{\wedge}(z)$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,

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

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