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FPGA based modified FXLMS algorithm for feedforward active noise control systems

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ABSTRACT

Several approaches have been introduced in literature for active noise control (ANC) systems. Since FxLMS algorithm appears to be the best choice as a controller filter, researchers tend to improve performance of ANC systems by enhancing and modifying this algorithm. In this paper, modification is in done the existing FxLMS algorithm that provides a new structure for improving the tracking performance and convergence rate based on the secondary path modeling technique. The convergence rate is improved by the dynamically varying the step size of the error signal. It is also implemented using FPGA.

Introduction

Acoustic noise problems become more and more evident as increased numbers of industrial equipment such as engines, blowers, fans, transformers, and compressors are in use. The traditional approach to acoustic noise control uses passive techniques such as enclosures, barriers, and silencers to attenuate the undesired noise [1] and [2]. These passive silencers are valued for their high attenuation over a broad frequency range; however, they are relatively large, costly, and ineffective at low frequencies. Mechanical vibration is another related type of noise that commonly creates problems in all areas of transportation and manufacturing, as well as with many household appliances. Active noise control (ANC) [3]–[4] involves an electro acoustic or electromechanical system that cancels specifically, an anti-noise of equal amplitude and the primary (unwanted) noise based on the principle of superposition; opposite phase is generated and combined with the primary noise, thus resulting in the cancellation of both opposite phase is generated and combined with the primary noise, thus resulting in the cancellation of both noises.

The most popular adaptation algorithm used for ANC applications is the FxLMS algorithm, which is a modified version of the LMS algorithm [5]. The schematic diagram for a single-channel feed forward ANC system using the FxLMS algorithm is shown in figure.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 (adaptive) 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). [6].

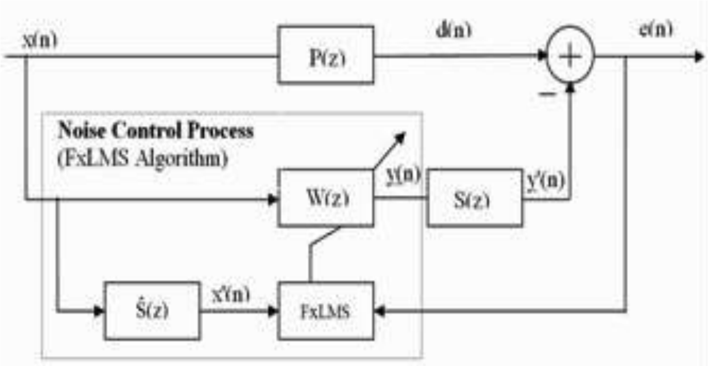

Fig. 1. Blockdiagram of FxLMS based feedforward ANC system.

Secondary Path Effects

In ANC system, the primary noise is combined with the output of the adaptive filter. Therefore, it is necessary to compensate $\hat{S}(z)$ for the secondary-path transfer from $y(n)$ to $e(n)$, which includes the digital-to-analog (D/A) converter, reconstruction filter, power amplifier, loudspeaker, acoustic path from loudspeaker to error microphone, error microphone, preamplifier, anti-aliasing filter, and analog-to digital (A/D) converter. The schematic diagram for a simplified ANC system is shown in figure2. From Figure 2. , the - transform of the error signal is

$$E(z) = [P(z) - S(z)W(z)]X(z) \qquad (1)$$

We shall make the simplifying assumption here that after convergence of the adaptive filter, the residual error is ideally zero $[\text{i.e., } E(z) = 0]$. This requires $W(z)$ realizing the optimal transfer function.

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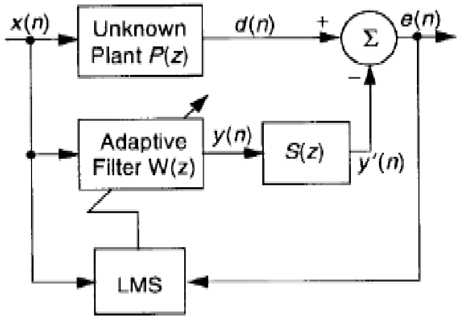

Fig. 2. Block diagram of simplified ANC system

$$W^o(z) = \frac{P(z)}{S(z)} \quad (2)$$

In other words, the adaptive filter has to simultaneously Model $P(z)$ and inversely model $S(z)$. A key advantage of this approach is that with a proper model of the plant, the system can respond instantaneously to changes in the input signal caused by changes in the noise sources. However, the performance of an ANC system depends largely upon the transfer function of the secondary path. By introducing an equalizer, a more uniform secondary path frequency response is achieved. In this way, the amount of noise reduction can often be increased significantly [4]. In addition, a sufficiently high-order adaptive FIR filter is required to approximate a rational function $1/S(z)$ shown in (2). It is impossible to compensate for the inherent delay due to $S(z)$ if the primary path $P(z)$ does not contain a delay of at least equal length.

FxLMS Algorithm

The FxLMS algorithm can be applied to both feedback and feed forward structures. Block diagram of a feed forward FxLMS ANC system of Figure 1.Here P (z) accounts for primary acoustic path between reference noise source and error microphone. $\hat{S}(z)$ is obtained offline and kept fixed during the online operation of ANC. The expression for the residual error e (n) is given as

$$e(n) = d(n) - y'(n) \quad (3)$$

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

$$y'(n) = s^T(n)y(n) \quad (4)$$

$$y(n) = w^T(n)\,x(n) \quad (5)$$

Where w (n) = [w0 (n) w_1 (n) ..w_{L-1}(n)]^T is tap weight vector, x(n)= [x(n) x(n-1).. 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:

$$w(n+1) = w(n) + \mu e(n)x'(n) \quad (6)$$

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

$$x'(n) = \hat{s}^T(n)\,x(n) \quad (7)$$

Proposed Method

The proposed method is based on varying the step size of FxLMS algorithm based on the secondary path modeling technique. This method utilizes VSS-LMS algorithm for modelling filter and uses *f(n)* as error signal for both $\hat{S}(z)$ and $W(z)$.The VSS-LMS algorithm is used to update modeling filter $\hat{S}(z)$ coefficients. Assuming that is an FIR filter of tap-weight length L , the output signal y (n) is computed as

$$y(n) = w^T(n)\,x(n) \quad (8)$$

Where w (n) = [w_0 (n) w_1 (n) …..w_{L-1}(n)]^T is tap weight vector, x(n)= [x(n) x(n-1)…. ….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). An internally generated zero mean white Gaussian noise signal v(n), uncorrelated with the reference signal ,x(n), is injected at the output y(n) of the control filter. Thus, the residual error signal is given as

$$e(n) = [d(n) - y'(n) + v'(n)] \quad (9)$$

Where d(n) = p(n) * x(n) is the primary disturbance signal, $y'(n) = s(n) * y(n)$ is the canceling signal, $v'(n) = v(n) * s(n)$.

Assuming that the modeling filter $\hat{S}(z)$ is an FIR filter of tap-weight length M, and its output $v''(n)$ is given as

$$v''(n) = S^T(n)v(n) \quad (10)$$

The output of $\hat{S}(z), v''(n)$ is subtracted from e(n) to the error signal for the modeling filter $\hat{S}(z)$ which is given as

$$f(n) = [d(n) - y'(n)] + [v'(n) - v''(n)] \quad (11)$$

The tap-weights of the modeling filter $\hat{S}(z)$ are updated using LMS algorithm

$$\hat{S}(n+1) = \hat{S}(n) + \mu(n)f(n)v(n) \quad (12)$$

The step size μ of the filter can be now updated as

$$\mu(n) = \mu(n) \times \exp\{x(n) \times f(n)\} \quad (13)$$

It has been note that initially the error f(n) of the system is large then it allows large step size then as the number of iteration continuous then the error of system is going to decreases, then it retains original step size.

In modified FxLMS, the step size is varied dynamically with respect to the error signal. Since error at the beginning is large, the step size of the algorithm is also large. This in turn increases convergence rate. As the iteration progresses, the error will simultaneously decreases. Finally, the original step size will be retained.Figure.3 shows the block diagram for proposed method.

Initially the error in the system is very high and so very large step size is selected. Hence the convergence rate is also very high .Then the step size is varied for the instant and the previous value of the error signal e (n). Finally the error is reduced greatly by the implementation of the dynamic step size algorithm.

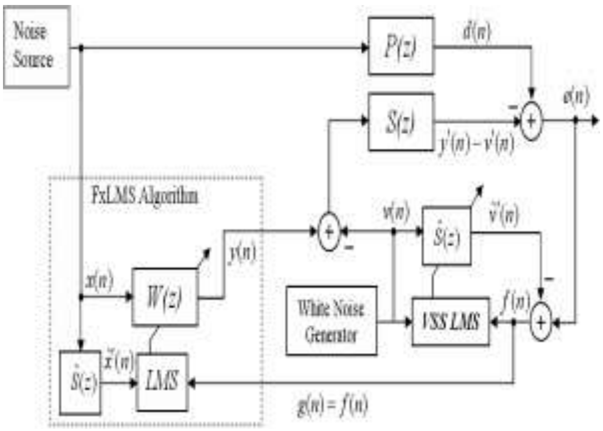

Figure.3 Block diagram for proposed Method

This idea of dynamic step size calculation is represented in "(14)" and "(15)".

$$w(n+1) = w(n) + \mu(n) f(n) x'(n) \tag{14}$$

The above equation represents the updating of the tap weights the modified FxLMS algorithm. From (11) $\mu(n)$ is the dynamic step size which is given by the below

$$\mu(n) = \mu(n) \times \exp\{x(n) \times f(n)\} \tag{15}$$

Thus the "(15)" is the dynamic step obtained based on secondary path modeling technique. Thus the "(14)" is called as modified FxLMS algorithm for improving the performance of existing algorithm.

Hardware Description Languages

The most popular hardware description languages are Verilog and VHDL. Both are text based depictions of the behavior of the digital circuit, and their syntax contains explicit notations for expressing time and concurrency. Gateway Design Automation Inc. started the Verilog language around 1984 as a proprietary hardware modeling language. The language went public in 1990 and has since been very popular in the semiconductor industry for VHDL is a hardware description language that grew out of the VHSIC program sponsored by the Department of Defense and was first released in 1985. The acronym VHDL, stands for VHSIC Hardware Description Language, with the acronym VHSIC standing for Very High speed Integrated Circuit.

High level Languages

There is increasing interest in using high level programming languages for FPGA design. Some, such as Celoxica's DK Design Suite, generate HDL from a C like language. The Confluence language, based on Python, also takes this approach. The custom language is compiled to generate a VHDL or Verilog circuit description. The Accel FPGA tool from AccelChip similarly produces a register transfer level (RTL) circuit description from a Matlab m file.An alternate approach is to generate the device netlist directly from the high level description. This is what the Lava language, still under research by Xilinx and others, does. Lava is based on the lazy programming language Haskell, but is not yet available for system design. A shortcoming of the highlevel design languages is their inability to instantiate vendor specific functions, such as block RAMs and DSP blocks. With the move toward incorporating further highly specific blocks, such as microprocessors, this shortcoming will need to be overcome before any of these languages takes hold.

Simulation Results

In this section the performance of the proposed modified FxLMS algorithm is demonstrated using computer simulation. The performance of the proposed algorithm is compared with that of FxLMS algorithm on the basis of noise reduction R (dB) and convergence rate is given in "(9)" and "(10)".

$$\text{R (dB)} = -10 \log\left(\frac{\sum e^2(n)}{\sum d^2(n)}\right) \tag{15}$$

$$\text{Convergence Rate} = 20 * \log 10\{abs(g)\} \tag{16}$$

The large positive value of R indicates that more noise reduction is achieved at the error microphone. The computer simulation for modified FxLMS algorithm performance is illustrated in figure (4), (5) and (6).

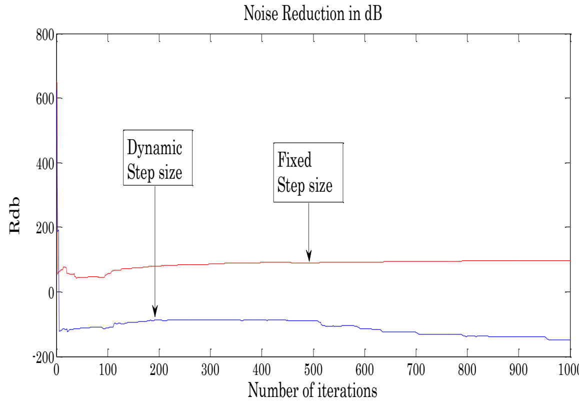

Figure.4 Noise reduction versus iteration time *(n)*

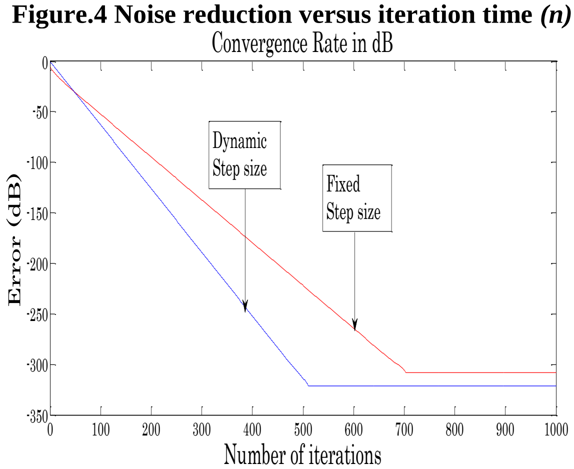

Figure.5 Characteristics of convergence rate

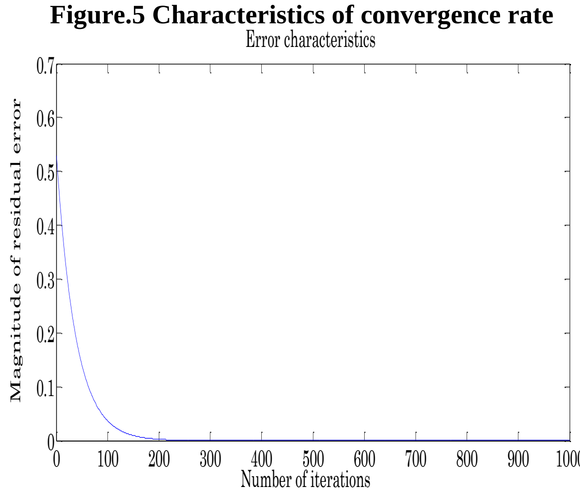

Figure.6. Characteristics of residual error f(n)

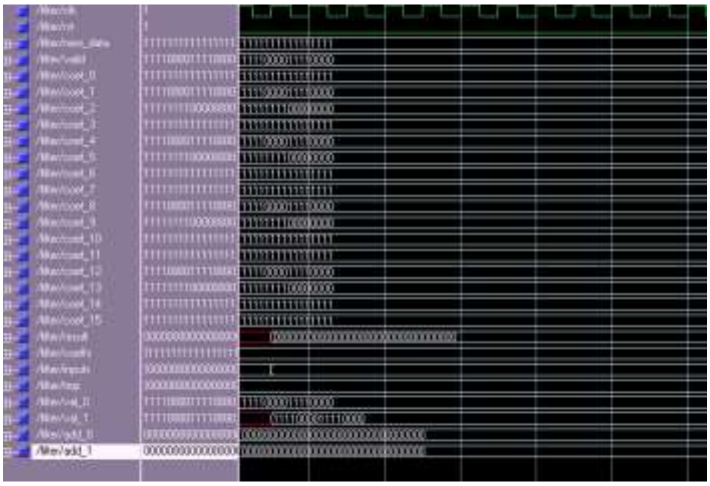

Fig.7 Simulation results obtained using VHDL

Figure.4 shows the characteristics of Noise reduction versus number of iteration times. It has been seen that the modified FxLMS with dynamic step-size produce better noise reduction compared with FxLMS with fixed step size. Figure.5. shows the characteristics of convergence rate in dB with respect to number of iterations. It has been seen that the convergence rate of modified FxLMS with dynamic step-size increases by reducing the number of iterations compared with modified FxLMS with fixed step size.

Figure.6. shows the characteristic of error versus number of iterations. It has been seen that error at the beginning is large, the step size of the algorithm is also large. This in turn increases convergence rate. As the iteration progresses, the error will simultaneously decreases and approaches to zero. Finally, the original step size will be retained. Thus the convergence rate as well as error can be minimized. Fig.6 shows the simulation obtained using the Modelsim, where the program was coded using VHDL.

Conclusions

Here we propose a modified FxLMS structure for ANC system. This structure combines the concept of modified FxLMS algorithm with the dynamic variable step size. It shows better tracking performance and convergence rate than the modified FxLMS algorithm with fixed step and conventional FxLMS algorithm. The main feature of this method is that it can achieve improved performance than the existing methods.

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