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A Novel Method for the Application of Adaptive filters for Active Noise Control System

Pradyumna Ku.Mohapatra
Asistant Professor
Department of Electronics &Telecommunication
Orissa Engg. College,BBSR,Odisha
er_pradyumna@yahoo.co.in

Jibanananda Mishra
Associate Professor
Department of Electronics & Telecommunication
Orissa Engg. College, BBSR, Odisha

Abstract**—** This paper introduces one novel method for active noise control .Though use of filtered-X LMS FIR Adaptive Filter mature in the literature ,this expression illustrates the application of adaptive filters to the attenuation of acoustic noise via active noise control. The reference signal is a noisy version of the undesired sound measured near its source. We shall use a controller filter length of about 44 msec and a step size of 0.0001 for these signal statistics. The resulting algorithm converges after about 5 seconds of adaptation. We also realize adaptive algorithm using IIR filter with active noise to overcome the ability of acoustic feedback . The direct form IIR filter structure, which faces the difficulties of checking stability and of relatively slow convergence speed for noise composed of narrow band components with large power inequality. To overcome these difficulties along with using the direct form IIR filters filtered-u LMS algorithm is used.

Keywords*- Active Noise, Acoustic noise cancellation, Filtered-XLMS algorithm.*

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I. INTRODUCTION *(HEADING 1)*

Using an electro-acoustic system dimension sensors such as microphones and output actuators such as speakers one can reduce the intensity of an unwanted dB noise propagating through the air. Usually the noise signal comes from any piece of equipment, such as a revolving machine, so that it is possible to measure the noise near its source. The end of the active noise control scheme is to create an"anti-noise" that attenuates the unwanted noise in a desired settle down region using an adaptive filter. In view of the fact that of that inadequacy of the physical barriers, active means to cut low frequency noise (less than 500-1000 Hertz) have been look into by researchers in the area of adaptive acoustic control. Active noise control (ANC) plays an important role in which noise can be reduced in the form of a small package of a DSP controller, microphone(s), and loudspeaker(s). The ANC systems are effective,it may be better or worse only when the intended noise is periodic and so random noises like the white noise will not be reduced. This problem differs from traditional adaptive noise cancellation in that:

- Only the attenuated signal is available but the most wanted response signal cannot be directly measured.
- The secondary loudspeaker-to microphone error path in its adaptation the ANC system must take into account.

The application of adaptive filters to the attenuation of acoustic noise via active noise control FXLMS FIR Adaptive Filter is used. The path in which the anti-noise takes from the output loudspeaker to the error microphone within the quiet zone is called as secondary propagation path. Here we considered as the impulse response which is band limited to the range 160 - 2000 Hz with a filter length of 0.1 seconds. For this, ANC We shall use a sampling frequency of 8000 Hz. The structure of the ANC is normally classified into two classes: feed forward control and feedback control. In the feed forward control, a reference noise is presumed to be usable to the adaptive filter. Depending on the type of primary noise that can be shortened feed forward ANC systems can be categorized as either a broadband or a narrowband. In the broadband feed forward control case, a reference noise is observed by a reference sensor (e.g.,microphone), and thus noise correlating with the reference noise can be scaled down. For ANC, IIR adaptation has been used as filtered–U algorithm but this combines the main path adaptation with the cancellation of the reverse feedback from secondary source to reference input [1].

II. ADAPTIVE ANC SYSTEM

The ANC system using filter x-LMS algorithm are shown in figure-1. The system is excited with i/p signal x (n) and output of adaptive filter y (n) to cancel the primary noise at the error microphone location. Though the use of Filtered-X LMS FIR Adaptive Filter is mature in literature, adaptive filters are defined for problems such as electrical noise cancelling where the filter output is an estimate of a preferred signal. In many applications, the adaptive filter works as a dynamic system Controller which containing actuators and amplifiers etc. The estimate (anti-vibrations or anti-sound) in this case can thus be seen as the output signal from a dynamic system, i.e. a forward path. A conventional adaptive algorithm such as the LMS algorithm is likely to be unstable in this application due to the phase shift (the delay) introduced by the forward path [1, 2]. The well-known filtered-x LMS-algorithm is, however, an adaptive filter algorithm which is suitable for active control applications [3].

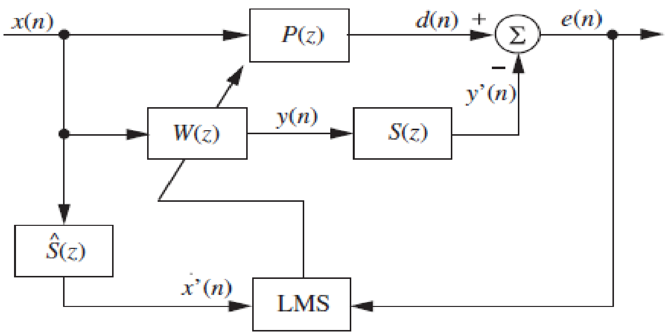

Fig. 1. Equivalent sampled-time block diagram of the broadband feedforwardANC systemshown in Fig. 1. In this figure, *P (z)* is the primary path, *S(z)* is thesecondary path,*W(z)* is the control filter, and *ˆS(z)* is the secondary-pathmodel..

A.**The FXLMS algorithm:** Figure-1 shows the block diagram of an ANC system in which the FXLMS algorithm is applied.The output of the adaptive filter i.e y(n) can be obtain as $y(n) = W^T(n)x(n)$ (1)

Where w(n)=$[w_0\ (n)\ w_1\ (n) \dots\ w_{L-1}(n)]^T$ and $X(n) = [x(n)x(n-1)\ .\ .\ .\ x(n-L+1]^T$ are the coefficient of signal vector $W(Z)$and L is the filter length. The adaptive filter updates coefficient of signal vector $W(Z)$ using FXLMS algorithm which can be expressed as $w(n+1) = w(n) + \mu e(n)x'(n)$ (2)

Where μ is the convergence factor that determines the convergence speed.

$x'(n) = \hat{s}(n) \otimes x(n)$ (3)

is the filtered reference signal vector and $\hat{s}(n)$ is the impulse response of the secondary-path estimation filter. The accurate estimation of the secondary-path model the ANC system requires FXLMS algorithm. This algorithm shows that when secondary path, S(z) , follows the adaptive filter, this transfer function must also be placed in the reference signal path. The basic ANC system described above perform quite well in reducing broadband as well as narrowband noise in ducts under plane wave conditions. In this arrangement primary path and secondary path are assumed to be linear in nature and so represented by linear transfer functions. To implement FXLMS algorithm the secondary path filter S(z) is to be estimated first. Many offline and online methods are available for identifying the secondary path filter. FXLMS algorithm found to be tolerant to errors made in the estimation of $\ddot{S}(z)$. FXLMS algorithm generally converge even with a phase estimation error of up to 0 ± 90 , within the limit of slow adaptation [2]. The maximum allowable step size for FXLM algorithm is approximately[5]

$$\mu_{max} = \frac{1}{P_x'\ (N+\Delta)} \qquad (4)$$

Where $P_x' = E[x'^2(n)]$ is the mean square value or power of the filtered reference signal $x'(n)$

N is the number of adaptive filter coefficients and Δ is the number of samples corresponding to the overall delay in the secondary path. The the most significant factor which influencing the convergence behavior of the ANC system is delay in the secondary path, thus reducing the maximum step size in the FXLMS algorithm.

***B.* Feedback path ANC :**. The mere approach to solving the feedback problem is to use a feedback cancellation filter that models the feedback path from the secondary loud speaker to the reference sensor, which is precisely the same technique used in acoustic echo cancellation [2]. In the broadband feed forward ANC, when a feedback path is present, the optimal solution is generally an infinite impulse response (IIR) function with poles and zeros. The purpose of the IIR filter can be considerably more efficient for the realization than FIR filters, because an IIR filter may require much fewer coefficients than an FIR filter to model any resonance systems.

III. ANC USES IIR FILTER

IIR adaptive filters have the ability to give up matching Characteristics with lesser filter coefficients compared to FIR adaptive filters. Fall in number of filter coefficients leads to decrease in computational complexity for ANC implementation. ANC introduces poles in the system in the case of acoustic feedback. So IIR adaptive filters can better match the physical system as they have zeros as well as poles whereas FIR filters have only zeros. The ANC using an IIR filter is mature in the literature.The algorithm used in this paper for IIR adaptive filter is called to filter-u recursive LMS algorithm. The only drawback of this algorithm is that even though experimentally it works well, the stability and global convergence is not guaranteed [4].

IV. SIMULATION OF ANC USING FXLMS:

To give emphasis to the difference we run the system with no active noise control for the first 200 iterations. Listening to its sound at the error microphone before cancellation, it has the quality industrial "whine" of such motors. Once the adaptive filter is enabled, the resulting algorithm converges after about 5 (simulated) seconds of adaptation. Comparing the spectrum of the residual error signal with that of the original noise signal, we see that most of the periodic components have been attenuated considerably. The steady-state cancellation performance may not be uniform across all frequencies,

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however. Such is often the case for real-world systems applied to active noise control tasks. Listening to the error signal, the annoying "whine" is reduced considerably.

Step-1:Generate sine wave with random phase.
Step-2: Generate synthetic noise by adding all sine waves.
Step-3:Propagate noise through primary path.
Step-4:Add measurement noise
Step-5:No noise control for first 200 iterations.
Step-6: Play noise signal.
Step-7: Show spectrum of original and attenuated noise.

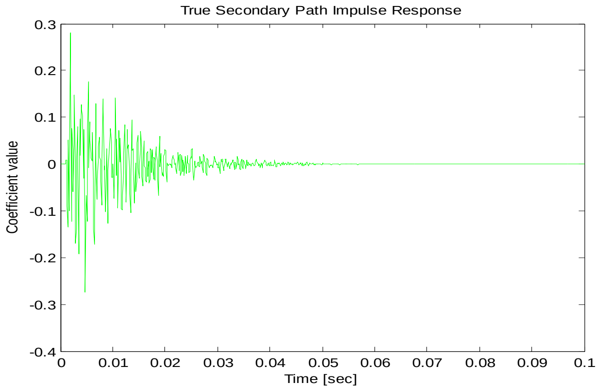

Fig 2:The secondary path impulse response

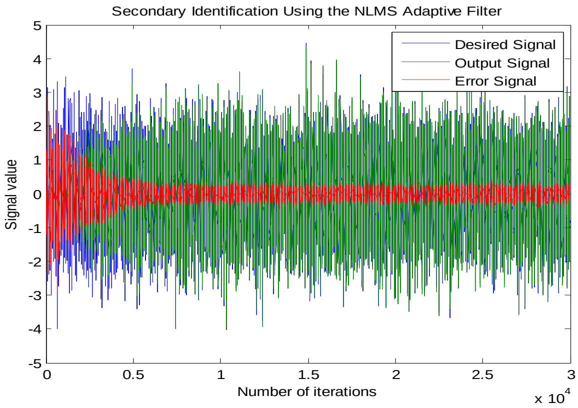

Fig 3:The secondary identification using NLMS

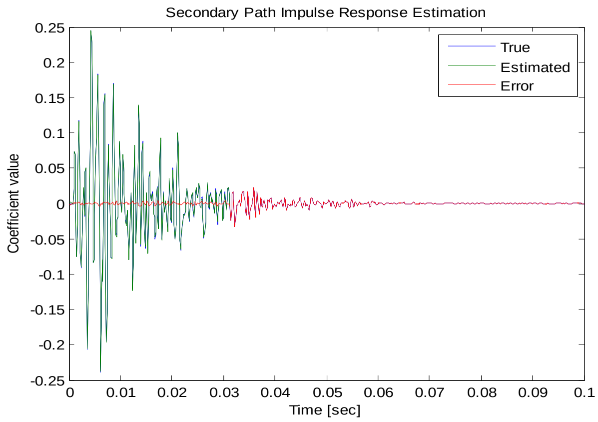

Fig 4:Secondary path impulse response estimation

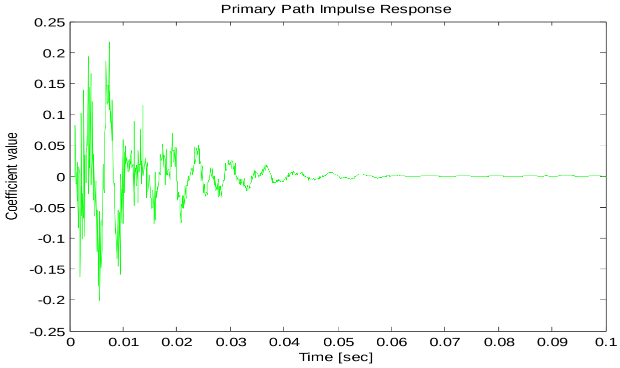

Fig 5:Primary path impulse response

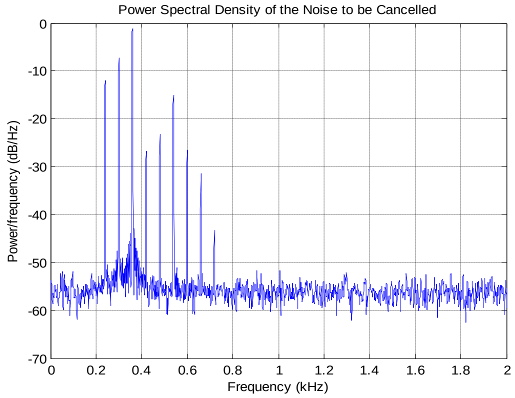

Fig 6: PSD of the Noise to be cancelled

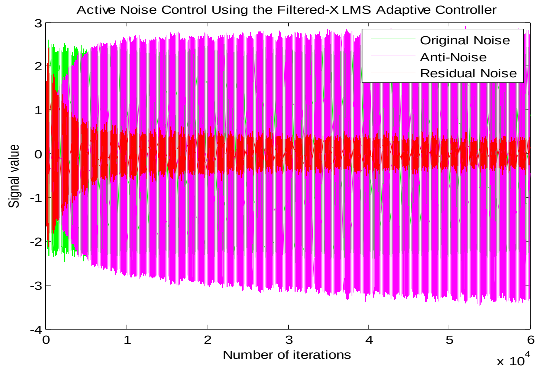

Fig 7:ANC Using the FXLM adaptive Controller

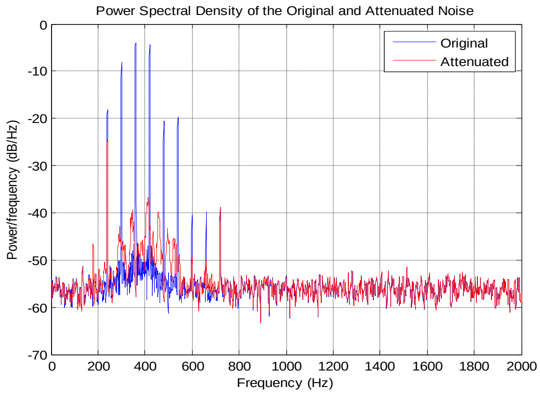

Fig 8:PSD of the Original and Attenuated Noise

V. SIMULATION OF ANC ANC USING FIR FILTER

The limitations of FXLMS algorithm is the magnitude response and the convergence rate of the adaptation since it depends on loop gain of adaptation path. So the proposed New filtered MFXLMS AlGORITHM is used to overcome not only the said problem but to achieve more predictive power. In this case We choose number of taps of filter is 20. step-size parameter mu=0.001. We have to compute reference singals d1= x*Hp1(Z) and d2=x*Hp2(Z). Aso compute filteroutput s1=r1*w and s2=r2*w where r1=x*Hs1(z)and r2=x*Hs2(z) update weights and plot noise and filtered signal

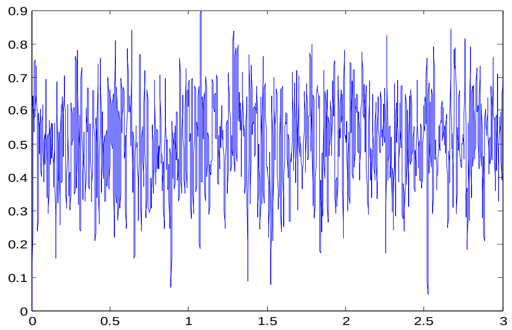

Fig9 : Generate signal

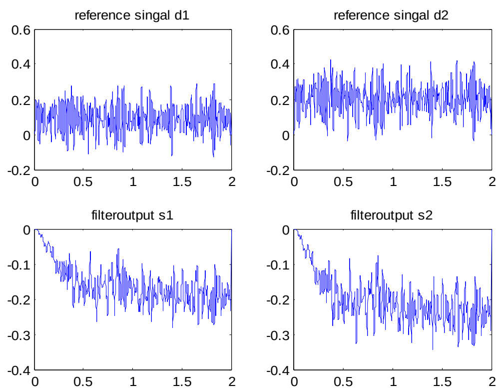

Fig10: Noise and filtered signals

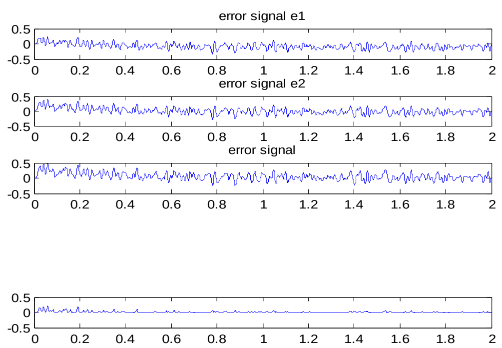

Fig 11: Error signals

VI. SIMULATION OF ANC USING IIR FILTER

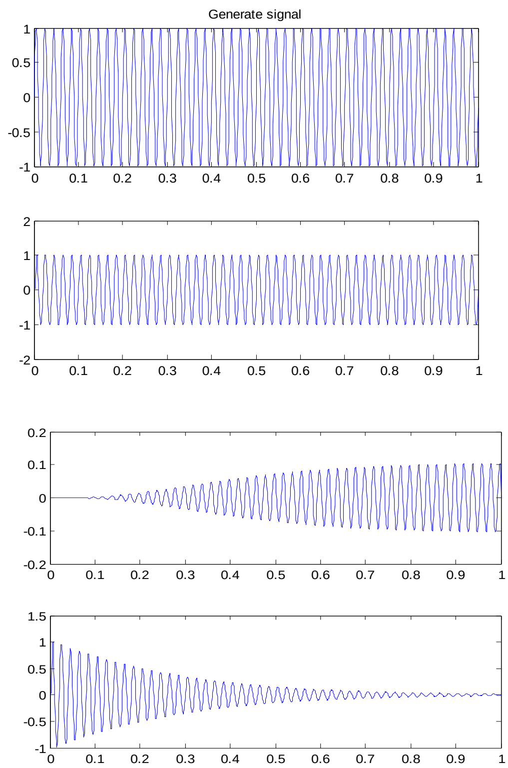

Fig12: Noise and filtered signals using IIR Filter

VII. CONCLUSION AND FUTURE WORK:

This paper deals with active noise control using a filtered-X LMS FIR Adaptive Filter. We also realize adaptive algorithm using IIR filter with active noise to overcome the ability of acoustic feedback .We have to concentrate more to design efficient filter structures based on new topologies to deal with the ANC problems. Different novel methods based on evolutionary and bio-inspired techniques be developed with the basic purpose to optimally train the weights of the adaptive filter structures.

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