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Performance Evaluation of Active Noise Control Algorithm Using Matlab

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Abstract
Active noise cancellation technique is a promising area of research because it can suppress noise at low frequency with the help of adaptive filter algorithm and advanced digital signal processors. In this paper, the adaptive algorithm used for active noise control (ANC) problem is derived and analysed. Specifically two types of algorithm are studied: filtered-LMS (FXLMS) algorithm for broadband noise control and Waveform synthesis method based algorithm for narrowband noise control. Both the above said algorithms are simulated under different control conditions of noise.

Keywords:* *adaptive filter, ANC, FXLMS, waveform synthesis

1. Introduction

In today's world noise pollution has become a global hazard for mankind leading to hot discussions in various international forums including UNO, WHO etc. Exposure to growing noise pollution has become very hazardous for human being especially children. Prolonged exposure to noise levels at or above 80dB can cause deafness. Deafness, insomnia, vascular diseases are caused mostly due to noise hazards. Psycho-physiological problems such as panic attacks, memory loss, severe depression, headache etc. are some of the problems which are also related to noise hazard. People experiencing high noise levels have increased number of headaches, greater susceptibility to minor accidents, increased reliance on sleeping pills, increased mental sickness. Noise pollution has not only affected human health but also has crippled the wild life by causing hormone imbalance, chronic stress, panic and escape behavior thus putting a question mark to the very survival of wild life.

Major sources of noise pollution which are globally speaking dominant source of noise pollution are : transportation, neighborhood and domestic noise, industrial noise, car alarms and emergency service sirens, construction work, power tools, different Audio systems, loudspeakers and TVs, motor vehicles etc. Sound with high intensity produced by industries called industrial noise. Noise from various machines from the industries, factories, mills creates large scale noise problems has great impact on working people and the surrounding environment. Noise from grinding, welding, pneumatic drills, motors, mechanical saws, crane operation is unbearable to public. Workers in thermal power stations, mines, petrochemical, cement, steel industries due to exposure for long hours suffer from occupational population.

Sound pressure level which is denoted as SPL or Lp in decibel (dB) is defined by the formula as follows.

$$SPL = 20Log_{10}(SP / RP) \tag{1}$$

Where RP =Reference Pressure and SP =Sound pressure.
Human ear is audible to sounds with frequencies from 20Hz to 20,000Hz, which is called audible frequency range.

2. Methods of controlling noise

Noise cancellation technique is basically focused on how to remove or stop undesired sound. It can be done by two methods.

- Passive noise cancellation
- Active noise cancellation

Passive noise cancellation is the traditional approach which is done by preventing sound waves with the help of barriers, enclosures, silencers, earplugs, ear-protectors to reach eardrums. This technique is expensive, bulky and ineffective at low frequencies. To overcome these problems, active noise control (ANC) method is being sincerely investigated for last three decades.

2.2 Active Noise Control [2-6]

Active noise cancellation technique is a promising area of research because of it can suppress noise at low frequency with the help of adaptive filter algorithm and advanced digital signal processors. ANC can be realised by generating a secondary noise of equal amplitude and opposite phase of primary noise so that both the noise cancels each other. ANC is an electroacoustic or electromechanical system that depends upon the principle of destructive interference. The anti noise can be generated by using loudspeakers. The primary noise is either sensed or generated knowing its frequency and is fed to an adaptive filter which will try to generate the required antinoise. The noise and antinoise interfere with each other in the acoustic environment and generate a residual noise. The residual noise is sensed by a microphone and is called as the error microphone. This noise signal is used to monitor the noise cancellation performance as well as it is used to tune the parameters of the adaptive filters so that the residual noise is gradually minimized. The basic principle of noise and antinoise with destructive interference generating the residual noise is shown in Fig. 1. The principle of ANC is first proposed by Paul Lueg in 1936 [1].

The adaptive ANC algorithms are classified in two categories feedforward control and feedback control.

2.2.1 Feedforward ANC

The basic principle of broadband feedforward ANC is that the time delay between the input noise and the speaker which provide enough space to generate antinoise by the time the input noise signal reaches the speaker. Therefore, the noise is sensed from the source by a microphone and is fed to the controller. The controller generates the required antinoise by a loudspeaker. By the time the primary noise reaches the point of cancellation in acoustic medium, the controller should be ready with the antinoise to cancel it. Since the noise propagates through the air medium at a slower speed (340 m/s) and the noise after sensed by a microphone is converted to electrical signal which is moved at a velocity of light. Therefore, noise in the electrical path is moved faster compared to the acoustic medium and hence it is feasible to control the broadband noise using the ANC technique. The input noise signal generates antinoise of equal amplitude and 180° out of phase with the primary noise. This anti noise is output to a loudspeaker and used to cancel the unwanted noise. The error microphone measures the error signal (the difference between primary noise and output of the desired signal i.e. the output of ANC), which is used to tune the adaptive filter coefficient such that the error signal will be minimized. The schematic diagram of broadband ANC algorithm is shown in Fig. 2.

When the noise is predictable such as a narrowband noise or a tonal noise of some discrete frequencies this constraint is not seen and hence the noise need not be sensed from the source and it can be generated locally by a noise generation circuit or an algorithm. The schematic diagram of narrowband ANC algorithm is shown in Fig. 3. When the primary noise is periodic the input microphone can be replaced by a non-acoustic sensor like tachometer, accelerometer and optical sensor. This type of sensor eliminates the problem of acoustic feedback. The non-acoustic sensor signal is used to simulate an input signal that contains the fundamental frequency and all the harmonics of the primary noise.

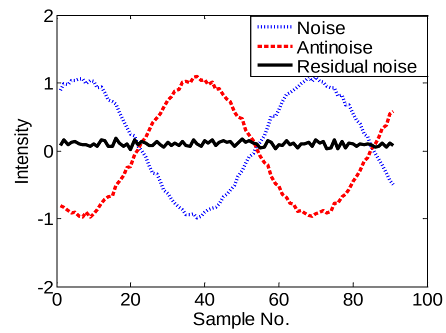

Fig. 1 Basic principle of ANC

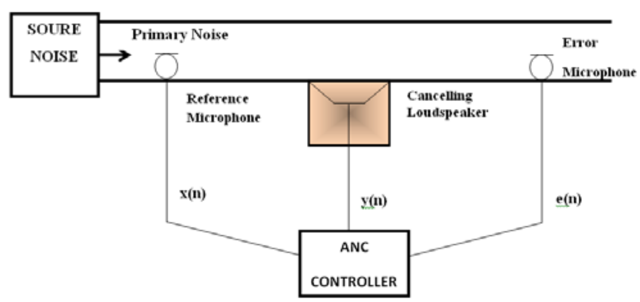

Fig.2 Feed forward ANC

2.2.2 Feedback ANC

Feedback ANC was proposed by Olson and May in 1963. In this type of ANC, the primary noise is neither sensed by an extra microphone, nor it is generated by a noise generator. The primary noise is estimated from the noise sensed by the error microphone. The error sensor signal is returned through an amplifier to produce anti noise at the sensor located near the input microphone. The regenerated reference input is combined with the primary noise to cancel it out. This type of configuration is restricted to

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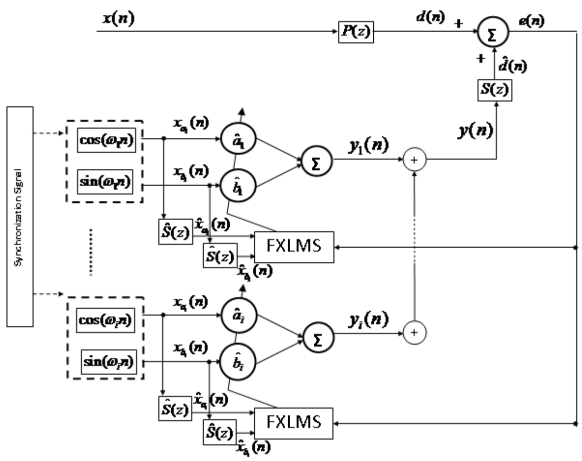

Fig. 3 Narrow band ANC

work for a band limited noise. It suffers from instability because of the possibility of positive feedback at high frequencies. The feedback ANC is normally used in controlling the sound effect in headphones and hearing protectors.

2.3 FXLMS algorithm based Broadband ANC

Active noise control uses a controller uses the input to generate the output by operating the input with its parameters. These parameters are adaptive in case of an adaptive controller. The tuning algorithms are use to tune the parameters such that the controller works to achieve its optimum performance. In this case the performance is measured in terms of minimum mean square error. The error is sensed by a microphone. The error signal is also used to tune the controller parameters to minimize the square error.

The block diagram of an adaptive ANC system is shown in Fig. 4 where the secondary path transfer function $S(z)$ is present at the output of the controller, $W(z)$. The residual signal, $e(n)$ is expressed as

$$\begin{aligned} e(n) &= d(n) - \hat{d}(n) \\ &= d(n) - s(n) * y(n) \\ &= d(n) - s(n) * [\mathbf{x}(n)\mathbf{w}^T(n)] \end{aligned} \tag{2}$$

where $d(n)$ is the noise at the point of cancellation and $\hat{d}(n)$ is the antinoise generated by the control loudspeaker. $y(n)$ is the output the control adaptive filter, $s(n)$ is the impulse response of the secondary path transfer function $S(z)$ at time n, ‘ $*$ ’ denotes linear convolution, The coefficient vector of the FIR filter in the controller is denoted as

$$\mathbf{w}(n) = [w_0(n) \quad w_1(n) \quad \cdots \quad w_{N-1}(n)] \tag{3}$$

And its transfer function in z-domain as $W(z)$. The input signal vector formed by accumulating N recent samples at time n is denoted as

$$\mathbf{x}(n) = [x(n) \quad x(n-1) \quad \cdots \quad x(n-N+1)] \tag{4}$$

where N is the order of filter $W(z)$. The objective of the adaptive filter is to minimize the instantaneous squared error. This is regarded as the cost function of this minimization algorithm and is defined as

$\xi(n) = e^2(n)$.

The least mean square algorithm updates the coefficient vector $\mathbf{w}(n)$ as

where μ is the step size parameter and $\nabla\xi(n)$ is an instantaneous estimate of gradient of $\xi(n)$ at time n, and can be expressed as

$$\mathbf{x}'(n) = [x'(n),\ x'(n-1),\ \cdots,\ x'(n-N+1)],$$

and

$$x'(n) = s(n) * x(n), \tag{5}$$

$$\mathbf{w}(n+1) = \mathbf{w}(n) + \mu\mathbf{x}'(n)e(n).$$

This update equation is called filtered-x LMS algorithm as the reference signal $x(n)$ is filtered through the secondary path $S(z)$ for updating the FIR filter coefficients $\mathbf{w}(n)$ of the controller. In practical ANC applications,

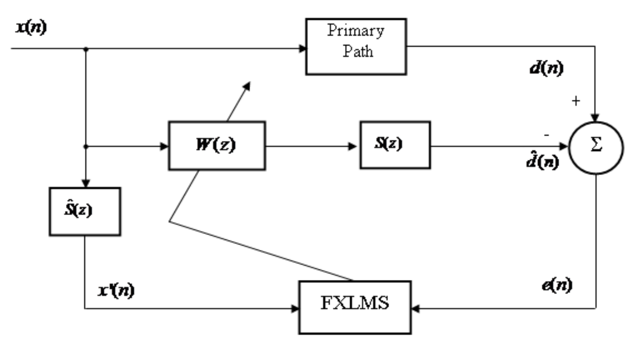

Fig. 4 Block diagram of FXLMS algorithm

2.4 Waveform synthesis approach

The narrowband ANC using a waveform synthesis method is well known to the ANC research community. The narrowband noise which essentially consists of I numbers of tonal components with different frequencies, phases and amplitudes may be represented in discrete form as

$$x(n)=\sum_{i=1}^{I}A_i\sin(2\pi f_i n/f_s+\varphi_i), \tag{6}$$

where A_i, f_i, φ_i are amplitude, frequency and phase of the i th tonal component, respectively. f_s is the sampling frequency and n is the sample index.

The objective of the waveform synthesis method based ANC is to synthesize this waveform when the frequency information is known from the synchronization signal. The estimated waveform represented as $y(n)$ should ideally be equivalent to $x(n)$ to achieve maximum cancellation performance. Since frequencies f_i ($1\le i\le I$) are known from the non-acoustic sensors, the amplitude and phase are only required to be estimated. The output of the waveform synthesizer is represented as

$$y(n)=\sum_{i=1}^{I}[\hat{b}_i(n)\sin(2\pi f_i n/f_s)+\hat{a}_i(n)\cos(2\pi f_i n/f_s)] \tag{7}$$

where $\hat{b}_i(n)$ and $\hat{a}_i(n)$ are the in-phase and quadrature coefficients respectively, which are adaptively estimated in the conventional narrowband ANC algorithms where it is assumed that frequencies f_i's are known and a fixed sampling frequency f_s, well above the maximum frequency of the noise is chosen. These adaptive coefficients should converge to the optimum values: $\hat{b}_i^o(n)=\hat{a}_i\cos(\pi\hat{\varphi}_i)$ and $\hat{a}_i^{\,o}(n)=\hat{a}_i\sin(\pi\hat{\varphi}_i)$.

The error microphone receives the residual noise, $e(n)$ which is a combination of primary and the secondary noises $d(n)$ and $\hat{d}(n)$ respectively. The residual noise is represented as

$$e(n)=d(n)+\hat{d}(n). \tag{8}$$

The secondary noise is generated as

$$\hat{d}(n)=y(n)S_j(n)=\sum_{j=1}^{L-1}S_j(n)y(n-j), \tag{9}$$

where $S_j(n)$ represents the impulse response of the secondary path which is represented in Z- domain as $S(z)$ and $y(n)$ is the ANC output. The parameters $\hat{b}_i(n)$ and $\hat{a}_i(n)$ are updated according to

$$\hat{a}_i(n+1)=\hat{a}_i(n)-\mu e(n)\hat{x}_{a_i}(n),. \tag{10}$$

$$\hat{b}_i(n+1)=\hat{b}_i(n)-\mu e(n)\hat{x}_{b_i}(n), \tag{11}$$

where

$$\hat{x}_{a_i}(n)=\sum_{j=1}^{L-1}\hat{S}_j(n)x_{a_i}(n-j), \tag{12}$$

$$\hat{x}_{b_i}(n)=\sum_{j=1}^{L-1}\hat{S}_j(n)x_{b_i}(n-j). \tag{13}$$

Here $\hat{S}_j(n)$ represents the j th impulse response of the secondary path estimate filter of length L and

$$x_{a_i}(n)=\cos(2\pi f_i n/f_s)=\cos(\omega_i n), \tag{14}$$

$$x_{b_i}(n)=\sin(2\pi f_i n/f_s)=\sin(\omega_i n), \tag{15}$$

where $\omega_i=(2\pi f_i/f_s)$.

In this way, $x_{b_i}(n)$ and $x_{a_i}(n)$ are generated for each component of the tonal noise and then multiplied by $\hat{b}_i(n)$ and $\hat{a}_i(n)$ respectively to synthesize the i th tone. The adaptive parameters $\hat{b}_i(n)$ and $\hat{a}_i(n)$ are updated at each iteration to minimize the error using least mean square (LMS) algorithm. Since the reference signals $x_{b_i}(n)$ and $x_{a_i}(n)$ are filtered through the secondary path for the adaptation of $\hat{b}_i(n)$ and $\hat{a}_i(n)$, respectively, the algorithm is also termed as filtered-x LMS (FXLMS) algorithm. The block diagram of this waveform synthesis based narrowband ANC system is shown in Fig. 4.

3. Simulation Study for Broadband ANC

To assess the stability and the convergence rate of the FXLMS algorithm a number of simulation experiments are carried out. In all these experiments, the transfer functions of primary path and the secondary path are chosen from the companion disc of [3]. These transfer functions were IIR filters and a FIR filter of order 127 is used to estimate the secondary path using LMS algorithm. The frequency and phase response of the primary and secondary paths and the secondary path estimates are shown in Fig. and their pole and zero frequencies are shown in the Table 1.

3.1 Experiment I

In this Experiment I the effect of the order of secondary path estimation filter is studied. The order of secondary path estimation filter was chosen as 128, 5 and 3 respectively. The mean square error in dB for all the three condition are plotted in Fig. 5. It is found that by reducing the filter order of the secondary path estimate the performance of noise cancellation is degraded.

Table 1: Pole zero coefficients of Primary and secondary path used in this simulation

Primary path Zeros	**Primary path Pole**	**Secondary path Zeros**	**Secondary path Pole**
-0.237467	-2.616971	-0.073844	-2.126286
0.806502	5.868128	0.280609	4.237403
-1.882598	-10.254027	-0.778907	-6.964616
3.601750	16.472815	-0.321869	10.683328
-5.938800	-23.574261	2.563927	-14.553812
8.734397	31.578068	-5.876712	18.877407
-11.601771	-39.180908	11.504314	-22.745882
13.989734	46.125938	-18.114344	26.226549
-15.218549	-51.048965	25.967108	-28.574720
14.661575	53.950325	-33.871861	29.879738
-11.943175	-54.055084	41.345085	-29.674311
7.090693	51.845238	-47.070049	28.296484
-0.801205	-47.205044	50.785450	-25.745836
-5.311028	41.234455	-51.863720	22.470392
11.085567	-34.143761	50.043129	-18.625055
-15.154731	27.083752	-45.990845	14.694556
16.927727	-20.176018	39.786583	-10.964108
-17.206356	14.395070	-33.078777	7.748295
15.141826	-9.473912	25.692730	-4.980707
-12.420147	5.902801	-18.861567	2.969857
8.971280	-3.258903	12.781915	-1.507602
-5.755672	1.655440	-8.174841	0.719325
3.278546	-0.656055	4.567914	-0.266916
-1.155305	0.208196	-2.076808	0.055631
0.303144		0.728670	

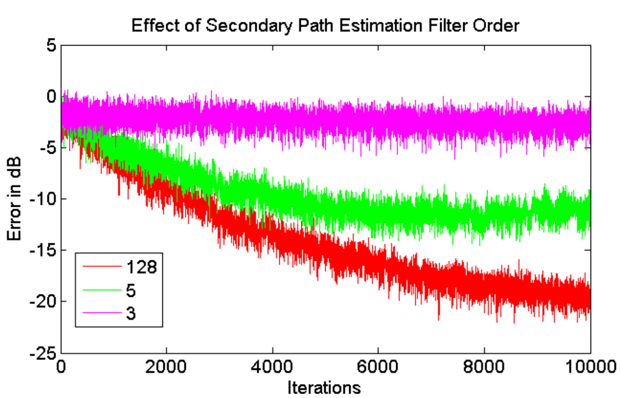

Fig. 5 Convergence Plot of mean square error of ANC for experiment 1

3.2 Experiment II

Experiment II is conducted to study the effect of step size of the FXLMS algorithm as the performance of noise control. The step for the FXLMS algorithm was taken as 0.01, 0.001, 0.0005, 0.0001. The mean square error in dB for the above all condition are plotted in Fig. 6. It is found out that by reducing the step size of the FXLMS algorithm the performance is degraded.

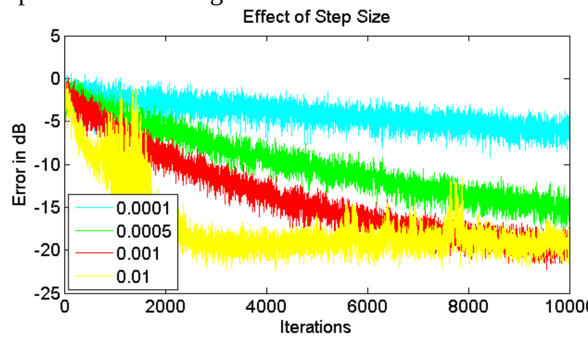

Fig. 6 Convergence Plot of mean square error of ANC for experiment II

3.3 Experiment III

Experiment III is conducted to study the effect of ANC order of FXLMS algorithm as the performance of noise control. The secondary path estimation was done using LMS algorithm. The order of secondary path filter was chosen as 120, 80, 60, 40, 20 respectively. The mean square errors in dB for all the condition are plotted in Fig.7. It is found that by reducing the ANC order the performance of ANC is degraded.

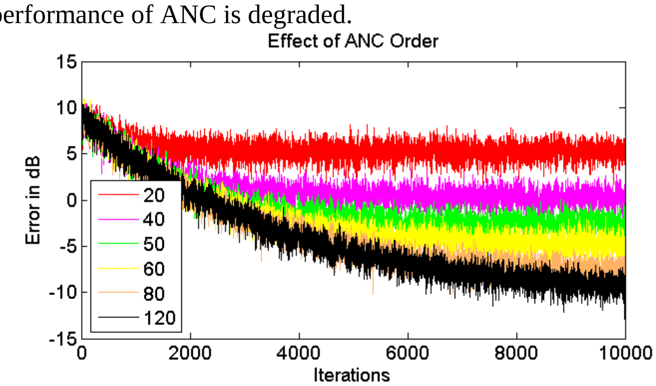

Fig. 7 Convergence Plot of mean square error of ANC for experiment III

3.4 Experiment IV

Experiment IV is governed to study the effect of noise power of the FXLMS algorithm as the performance of noise control.

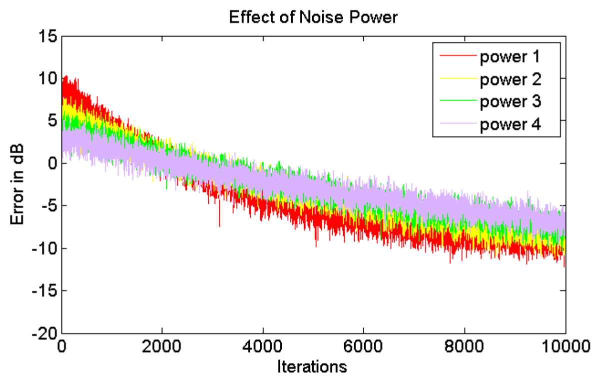

Fig 8: Convergence Plot of mean square error of ANC for

4. Simulation Study of Narrowband noise ANC

4.1 Monotone Noise

In this Experiment the effect of single tone frequency on narrow band ANC is studied. The primary path is chosen as $P(z) = z^{-5} - 0.6z^{-6} + 0.3z^{-7} - 0.9z^{-8} + 0.3z^{-9} + 0.5z^{-10}$

The actual secondary path was taken as
$S(z) = z^{-2} + 0.5z^{-3}$
Step size is taken as 0.01. The reference frequency and noise frequency is known as 300 Hz.

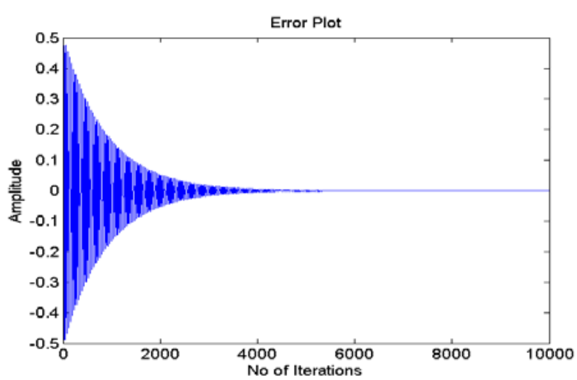

Fig. 9 Convergence plot of mean square error of narrow band

From the Fig 9, it is found out that after the convergence of error to zero, the narrowband ANC generates anti-noise of same amplitude and opposite phase of noise signal to cancel it out.

4.2 Harmonically related tonal

In this experiment the effect of harmonically related tonal noise 100 Hz and 200 Hz, is studied. The primary path and the secondary path is same as in case of the monotone noise. The noise frequency contains harmonically related frequencies 100 Hz and 200 Hz.

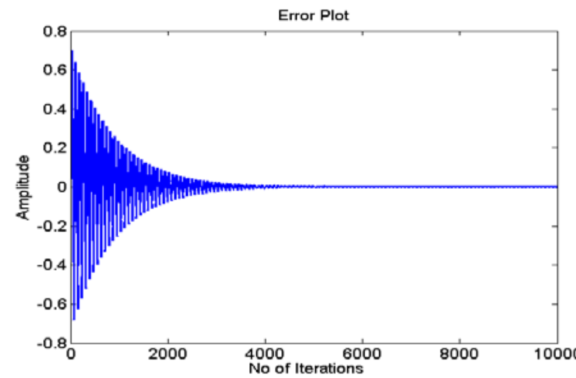

Fig. 10 Convergence plot of mean square error of narrowband ANC for harmonically related tonal noise with frequencies 100Hz and 200Hz

From the Fig. 10 it is found out that when there are harmonically related frequencies present in the primary noise source, the narrowband ANC effectively generates the anti-noise to cancel out the noise signal and make the mean square error to converge to zero.

4.3 Noise with beat frequency

Beat frequency is the small difference in the frequencies of two waves. The primary path and the secondary path are same as in case of the monotone noise.

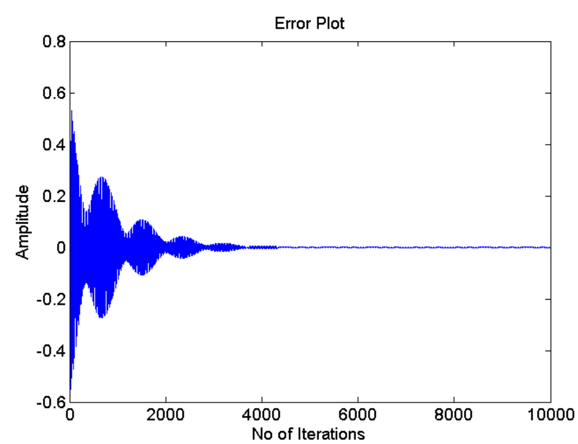

Fig. 11 Convergence plot of mean square error of narrowband ANC beat frequency

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In this experiment the primary noise contains frequency of 310Hz and internally generated reference noise contains frequency of 300Hz, so the beat frequency error is 10Hz. From the Fig. 11, it has been observed that the beat frequency error is present in the mean square error of the narrowband ANC and it is degrading the performance of the narrowband ANC.

4.4 Mixer audio noise

In this experiment a recorded mixer noise is analyzed in frequency domain. The mixer noise in time domain plot is shown in Fig. 12. Then the noise is studied in frequency domain by FFT analysis.

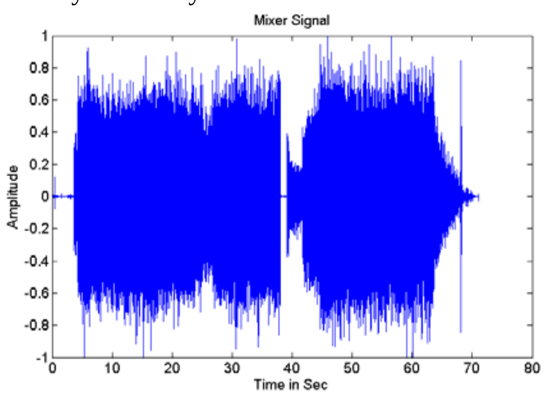

Fig. 12 Mixer noise plot in time domain

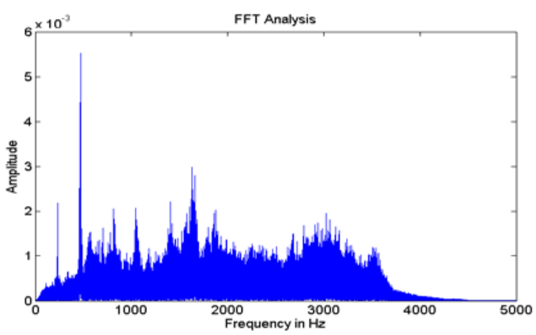

Fig. 13 Plot of FFT based frequency estimation of mixer noise

The FFT plot shows that the mixer noise is dominated by some low frequency components which can be controlled by narrowband ANC. The FFT of the mixer noise is shown in Fig. 13.

4.5 Primary path change

In this experiment the effect of the order of the primary path and secondary path estimation filter is studied in the case of narrowband ANC. The primary path and secondary path are chosen as before. The order of the primary path is chosen as 10 and 2 respectively. The mean square error in dB for above two conditions is plotted in Fig. 14 and Fig. 15 respectively. From the simulation result it is observed that the change in primary path conditions have no effect on the control performance of this particular narrowband ANC system.

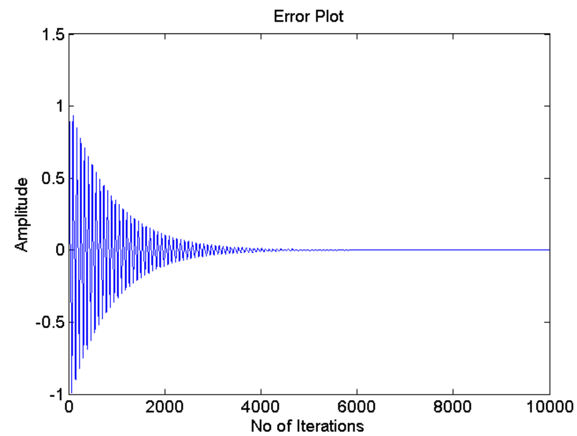

Fig. 14 Convergence plot of mean square error of primary path of order 10.

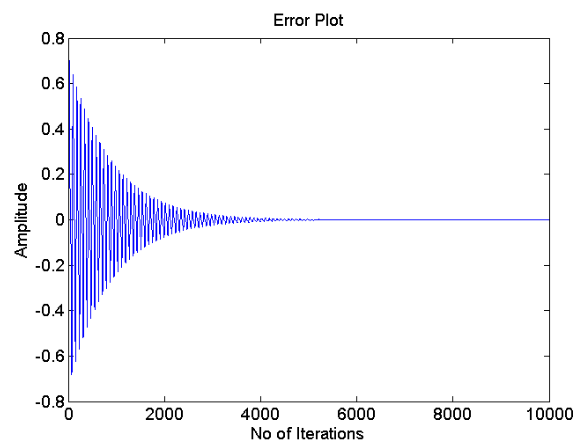

Fig. 15 Convergence plot of mean square error of primary path of order 2.

From the above Fig.14 and Fig. 15 it is found out that there is no change in the convergence property of the mean square error.

4.6 Secondary path Change

The secondary path order is chosen as 2 and 3 and the simulation result is given below.

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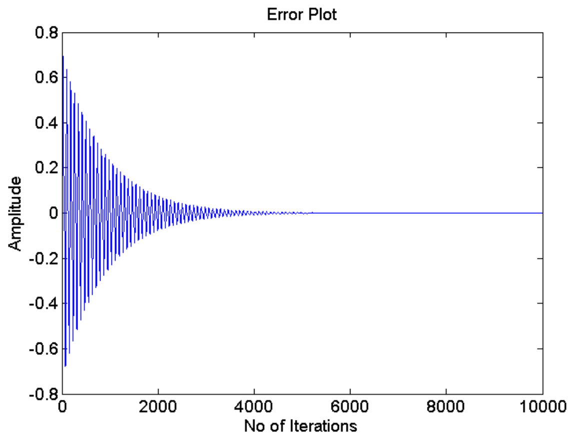

Fig. 16 Convergence plot of mean square error of ANC in order of secondary path 2

From the Fig. 16, it is found that there is no change in convergence property.

4.7 Frequency mismatch

If the actual frequency of the noise and the frequency used to generate antinoise has a mismatch, the algorithm fails to provide optimal solution of the ANC parameters. In addition, the performance of the ANC system degrades when there is a mismatch between the actual and the estimated secondary path. In this simulation study the internally generated primary noise contains 310Hz and the reference signal contains frequency of 300Hz. So there is a frequency mismatch of 10Hz in the primary noise and reference noise.

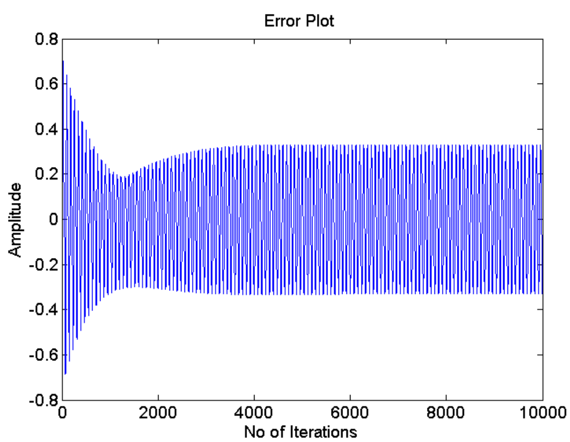

Fig. 17 Convergence plot of mean square error of ANC in frequency mismatch of 300Hz & 310 Hz

From the Fig.17 it is found out that, when there is a frequency mismatch, the performance of narrowband ANC is degraded.

4.8 With Random Signal

In this experiment the primary noise contains a tonal signal of 300Hz and a random noise generated by MATLAB.

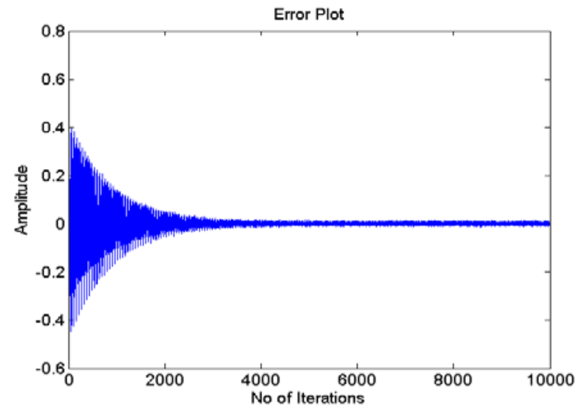

Fig. 18 Convergence plot of mean square error of ANC with random signal

From the simulation result it is observed that the narrowband ANC efficiently control the tonal noise of 300Hz but from the convergence plot of the mean square error, it is found that the high frequency generated by the random noise is present in the error signal.

5. Conclusions

In this thesis, the active noise control techniques are studied which is found to be an effective tool to control. The feed forward and narrow band ANC is studied with different conditions. The characteristic equations for both the cases are derived and analyzed. The extensive simulation experiments are done to verify the study. To overcome the issues which affect the performance of the ANC system include the future work.

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