A Research of Noise Estimation and Removal Techniques for Speech Signal

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ABSTRACT--- In this paper, authors made an attempt to implement the active noise control technique (ANC) to decrease the amplitude of noise communicating through the environment using an electro-acoustic (EA) system with the help of measurement sensors such as microphones and output actuators such as loudspeakers. In general, the noise signal is generated from ambient; therefore, it is easy to detect the noise in the vicinity of its source. The main objective of developing the ANC system is to generate an "anti-noise" that reduce the unwanted noise in a desired quiet region using an appropriate adaptive filter. The simulations were performed in the MATLAB 2015 environment and satisfactory results were obtained using the proposed technique. The problem under study is different from traditional adaptive noise cancellation techniques in two ways. Firstly, it is not possible to measure the desired response of a signal directly measured; only the signal with reduced magnitude is present. Secondly, the ANC system is required to take into consideration the secondary loudspeaker-to-microphone error (LME) path in its adaptation.

1. INTRODUCTION

In case of electrical noise cancellation, Adaptive filters are mostly used wherein the desired signal is considered as a filter output. In control applications, the adaptive filter serves as a controller with the capability to control the dynamic systems containing actuators and amplifiers. In such types of systems, estimate (anti sound) is considered as the electrical quantity received from a dynamic system. Since dynamic system exists between the filter output and estimate, the selection of adaptive filter is done with utmost care. In this work, the ANC technique has been proposed to reduce the magnitude of an unwanted noise communicating through the environment using an EA system using measurement sensors such as microphones and output actuators such as loudspeakers. As we know that the noise signal generally comes from environment; therefore, it is possible to detect the noise close to its source. The ultimate goal of developing the ANC is to attenuate the noise signal in a desired quiet region by using an "anti-noise" that could be produced using an appropriate adaptive filter. After implementation of the proposed technique, the simulation results were obtained in MATLAB and the results seems to be quite satisfactory. The problem under study is different from traditional adaptive noise cancellation techniques in two ways. Firstly, it is not possible to measure the desired response of a signal directly measured; only the signal with reduced magnitude is available. Secondly, the ANC system is required to take into consideration the secondary LME path in its adaptation.

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2. METHODOLOGY USED & RESULTS

In control applications, the adaptive filter is used as a controller with the capability to control the dynamic systems containing actuators and amplifiers. In such sort of systems, estimate (anti sound) is considered as the output signal received from a dynamic system. Since dynamic system exists between the filter output and estimate, the selection of adaptive filter is done with utmost care. In this case, a conventional adaptive LMS algorithm is found to be unstable due to the delay provided by the forward path [1, 2]. The x-LMS algorithm is well known adaptive filter algorithm that has proven its worth for active control (AC) applications [3]. LMS algorithm is the foundation of filtered x-LMS algorithm, which is illustrated by Figure 1. In this algorithm, a forward path is introduced between the input signal and the algorithm for the adaptation of the coefficient vectors [3, 4].

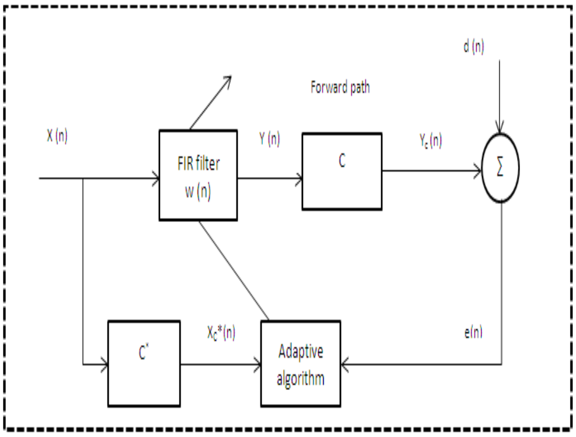

Fig.1. Active control system based on a filtered x-LMS algorithm

The implementation of active noise control technique is discussed as follows step by step:

(a) The Secondary Communication Path
(b) Determination of Secondary Propagation Path (SPP)
(c) SPP Design
(d) Performance evaluation of SPP Estimate
(e) The Primary Communication Path
(f) The Noise Cancellation
(g) ANC Using the Filtered-X LMS
(h) Residual Error Signal (RES) Spectrum

(a) The Secondary Communication Path

SPP is considered as the path that is followed by anti-noise from the output loudspeaker to the error microphone (LTEM). A LTEM impulse response of frequency range 160 - 2000 Hz was produced with a filter of length 0.1[1-2]. In order to perform this operation, a sampling frequency of 8000 Hz was used.

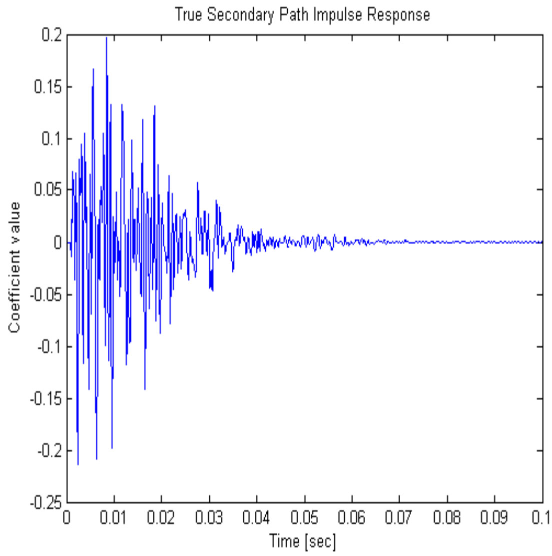

Figure 2: Impulse Response of True SPP

(b) Determination of SPP

In the ANC system, first we calculate impulse response of the secondary communication path. This step is generally implemented before controlling the noise through generation of random signal. It is obtained from the output loudspeaker in absence of noise. To accomplish this, the random signal of 3.75 seconds time duration along with the measured signal at the error microphone was generated [3]. The impulse response of secondary path is illustrated in Figure 2.

(c) SPP Design

Most often, the length of the secondary communicating path filter is not equal to the length of the actual secondary communicating path. It does not require being so for an effective control in major active noise control techniques. A secondary filter with a length of 250 taps was used. This was used in response to an impulse response that has the length of 31 msec. While any adaptive FIR filtering technique may be used to perform it, the LMS algorithm is mostly preferred due to its reliability, robustness and higher efficiency [4-6]. The secondary path impulse response is plotted and illustrated in figre2. The output response of the normalized LMS filter is presented in Figure 3.

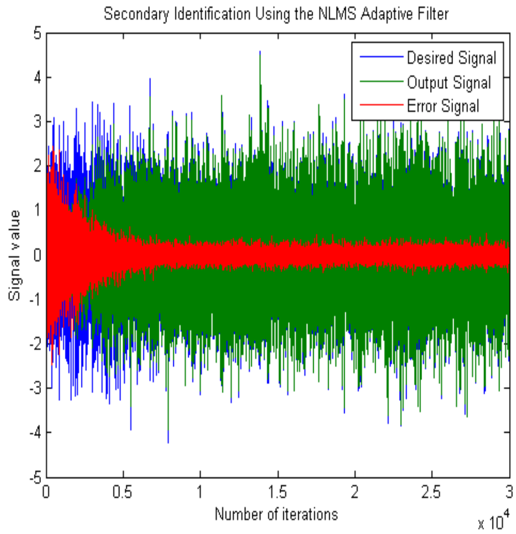

Figure 3: Output Response of Normalized LMS Filter

(d) Performance evaluation of SPP Estimate

The accuracy of the SP impulse output response is calculated from the plot that depicts the coefficients of both the actual and approximated path. However, it was not possible to estimate the actual output of the tail accurately. The ANC system remains unaffected due to this error while deployed for a particular task [7].The output impulse response for a secondary communicating path is displayed as figure 4.

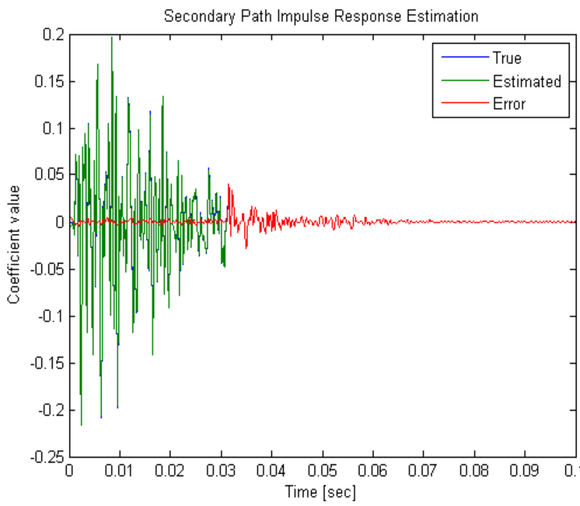

Figure 4: Output Impulse Response for Secondary Communicating Path

(e) The Primary Communicating Path

A linear filter was designed for the cancellation of noise in the propagation path. For this, the impulse response was generated that is shown in Figure 5 in the band of 200-800 Hertz.

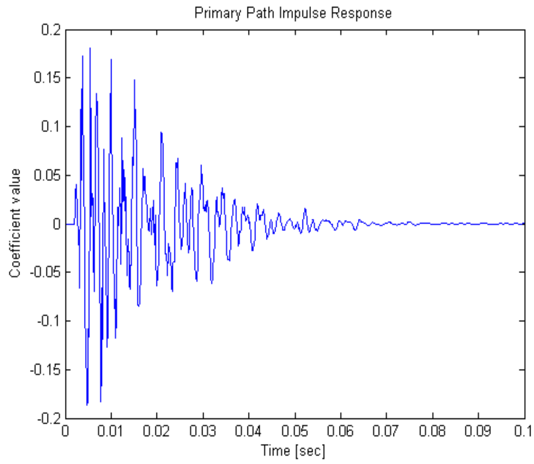

Figure 5: Output Response of Primary Communicating Path

(f) The Noise Cancellation

In our research work, a typical active noise control application was applied by synthetically generating 7.5 seconds of an ambient noise [8]. The spectrum of the sound was also generated. The plot for the power spectral density (PSD) of the estimated noise is given in Figure 6.

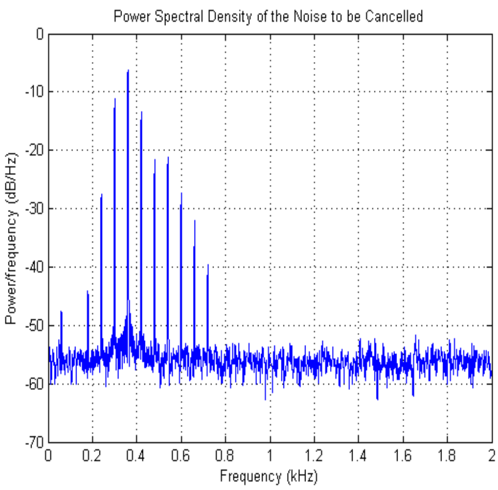

Figure 6: PSD of the Estimated Noise

(g) ANC Using the Filtered-X LMS

The filtered-X LMS algorithm is very popular adaptive system for ANC. In this we approximated the secondary path to yield an electrical output response which get mixed with the noise at the error sensor. Here, we get the noisy signal. In this, a controller filter length of about 44 msec with a size of 0.0001 has been incorporated for this signal [9, 10]. The resulting algorithms tend to converge after 5 seconds. With the error signal, the output volume level is decreased significantly. The output response of Filtered LMS is illustrated in Figure 7.

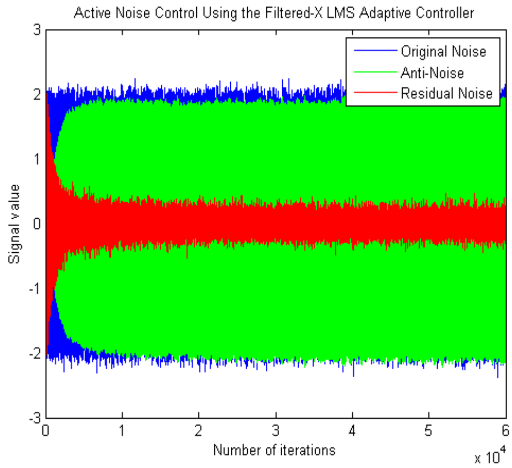

Figure 7: ANC Using the Filtered LMS Adaptive Controller

(h) REC Spectrum

After having compared the spectrum of the RE signal with the original one, we found that most of the periodic components were suppressed highly. The noise cancellation was not done equally over all frequencies range; however, this is acceptable for real-world systems applied to ANC systems [19-20]. PSD plot of original and suppressed noise signal is given in Figure8.

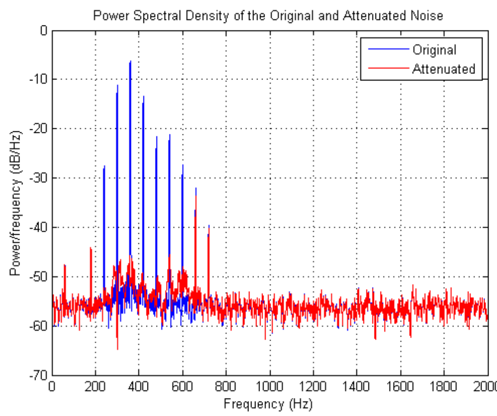

Fig. 8: PSD of the Original and suppressed Noise Signal

3. CONCLUSION

In the current work, a filter of 0.1 seconds duration was generated for SSP by using a LTEM impulse response having band limit range from 160 - 2000 Hz. A sampling frequency of 8000 Hz was used in this. To approximate the secondary propagation path, 3.75 seconds of random noise signal was activated along with the detected signal at the microphone. SPP was designed using a secondary communicating path filter with a length of 250 taps, in response to an impulse signal 31 milliseconds duration. The LMS filter has been employed here due to its better performance and robustness. The output and error illustrated by different figures demonstrates that the algorithm converges after running 10,000 iterations. For creating the primary propagation path, a filter with a length of 0.1

seconds was employed using an input-to-error microphone impulse response in the frequency range of 200 - 800 Hz. In our research work, a typical ANC application has been applied by synthetically generating 7.5 seconds of an ambient noise.

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