Simulation for noise cancellation using LMS adaptive filter

To cite this article: Jia-Haw Lee et al 2017 IOP Conf. Ser.: Mater. Sci. Eng. 211 012003

View the article online for updates and enhancements.

Related content
- Optimal design on digital filter algorithm of Hankel transform
  Yunwei Zhao, Zi-qiang Zhu, Guang-yin Lu et al.
- Study on Underwater Image Denoising Algorithm Based on Wavelet Transform
  Sun Jian and Wang Wen
- A method of background noise cancellation for SQUID applications
  D F He and M Yoshizawa
Simulation for noise cancellation using LMS adaptive filter

Jia-Haw Lee¹, Lu-Ean Ooi¹ *, Ying-Hao Ko² and Choe-Yung Teoh²

¹School of Mechanical Engineering, Universiti Sains Malaysia, 14300 Nibong Tebal, Pulau Pinang, Malaysia
²Department of Mechanical Engineering, Faculty of Engineering and Built Environment, Tunku Abdul Rahman University College (TAR UC), Malaysia

*Email: ooiluean@usm.my

Abstract. In this paper, the fundamental algorithm of noise cancellation, Least Mean Square (LMS) algorithm is studied and enhanced with adaptive filter. The simulation of the noise cancellation using LMS adaptive filter algorithm is developed. The noise corrupted speech signal and the engine noise signal are used as inputs for LMS adaptive filter algorithm. The filtered signal is compared to the original noise-free speech signal in order to highlight the level of attenuation of the noise signal. The result shows that the noise signal is successfully canceled by the developed adaptive filter. The difference of the noise-free speech signal and filtered signal are calculated and the outcome implies that the filtered signal is approaching the noise-free speech signal upon the adaptive filtering. The frequency range of the successfully canceled noise by the LMS adaptive filter algorithm is determined by performing Fast Fourier Transform (FFT) on the signals. The LMS adaptive filter algorithm shows significant noise cancellation at lower frequency range.

1. Introduction

Random fluctuation of the sound wave is known as noise. The presence of noise in a signal alters the original signal and results in inaccurate transmission of the desired signal. It is an issue in many applications such as headphones, engines, pumps and etc. One of the examples is the noise generated by the jet turbine engine that affects the comfort level of the airlines passengers [1]. Another example is the noise presence in hearing aids due to acoustic feedback effect in which reduces the performance of the hearing aids [2]. Due to the concern on the impact of noise on human, many researches have been done in aiming to eliminate the noise in different applications.

Several methods have been proposed in noise cancellation. The traditional noise reduction technique is based on passive noise control, such as using earplugs, ear-protector, sound insulation walls and muffler [3]. Passive noise cancellation (PNC) uses physical object that installed in the system to isolate the background noise from the surrounding. PNC techniques are effective in reducing noise over a wide frequency range. Chen and Yijun (2016) reduced the tonal noise from the centrifugal fan Xu and Mao have implemented PNC in forward-curved centrifugal fan by 5dB after employing the open-cell metal foam [4]. However, the major restriction of passive noise cancellation is the isolation object is relatively big in size and unable to be fitted into small scale applications. It is also limited by the effective frequency range. Thus, active noise cancellation (ANC) is developed in order to cover a wider range of the noise frequencies especially [5]. ANC employs an electro-acoustic
system to cancel the primary noise based on the principle of superposition, where an anti-noise (secondary signal) of equal amplitude but with antiphase is generated and combined with the primary noise in achieving the cancellation of noise [6]. ANC has greater advantages over PNC in terms of the capability to attenuate the low frequencies noise due to the presence of embedded control system which is able for the real time situation and well as low cost. ANC had been used in the microcontroller and achieved the cancellation of 20dB to 25dB of the 60Hz periodic noise component in the electronics signal measurement [7]. However, the implementation of ANC is complicated although it has high reliability in cancelling noise.

There are several algorithms that can be utilized in noise cancellation such as Least Mean Square Algorithm (LMS) and Filtered-x Least Mean Square Algorithm (FXLMS). LMS algorithm was introduced by Widrow and Hoff in 1960. It uses gradient-based method of steepest decent and updating coefficients of an adaptive filter [8]. The LMS algorithm is classified as adaptive filtering in which is a self-designing and time-varying system that continuously adjust its tap weight in the algorithm [9]. The LMS algorithm is implemented in aiming to minimize the noise in the input signal and producing a noise-free output. Sukhpreet and Sukhwinder used LMS algorithm in suppressing the effect of acoustic noise in the speech signals. The results show that LMS algorithm provides a greater signal-to-noise ratio value and suppressed the effect of acoustic noise [10]. In contrary, the FXLMS algorithm requires a filtered version of the reference signal as input in which the filter is having the same impulse response as the cancellation path. FXLMS had been used on cancelling the periodic noise generated by laptop fan after identification [11]. Recently, Chang and Chu (2013) developed the variable tap length and step size FXLMS in improving the convergence rate and noise reduction ratio in ANC [12]. Yang et al. (2014) reduced 8dB noise level in the cabin of high-speed elevator by implementing modified FXLMS algorithm [13]. However, the FXLMS algorithm has higher complexity compared to LMS algorithm due to the consideration of the secondary path parameter.

In this paper, the simulation of noise cancellation using LMS adaptive filter in MATLAB software is presented. The noise corrupted speech signal is used as the input for the developed LMS adaptive filter algorithm. The filtered signal of the LMS adaptive filter algorithm and the reference signal which is the noise-free speech signal are compared to examine the effectiveness of the LMS adaptive filter in cancelling the noise. Fast Fourier Transform (FFT) is then applied to both the signals in order to identify the frequency range successfully cancelled.

2. LMS adaptive filter algorithm
The LMS adaptive filter algorithm that developed in this study is shown in Figure 1. The parameters $y_1$ and $y_2$ are the inputs of the algorithm in the form of column vector. This is study $y_1$ is the noise corrupted signal and $y_2$ is the noise signal. The parameter $W(k)$ is the column weight vector of the filter at $k^{th}$ time, which is used in the algorithm to update the subsequent column weight vector and can be represented in the equation (1) below:

$$W(k) = \begin{bmatrix} W_0(k) \\ W_1(k) \\ \vdots \\ W_{L-1}(k) \end{bmatrix}$$

![Figure 1. LMS adaptive filter algorithm.](image-url)
The notation is the length of the filter in which is equal to the length of the input and. The weighted-noise signal $y_2^\prime$ is given by the multiplication of the column weight vector and the noise signal. The error $e(k)$ at $k^{th}$ time is then defined as the difference between the noises corrupted signal and the weighted-noise signal as shown the equation (2) as below. The notation is the transpose of the vector.

$$e(k) = y_1(k) - y_2^\prime(k)$$

$$= y_1(k) - [W^T(k)y_2(k)]^T$$

(2)

The adaptive algorithm is representing by the update of the column weight vector after each iteration and is given by equation (3). The parameter $\mu$ is the step-size that determines the convergence and stability of the filter.

$$W(k+1) = W(k) + \mu e(k)y_2(k)$$

(3)

At here, the LMS adaptive filter algorithm will be terminated when column weight vector iterates up to the length of filter and the final output of the algorithm is $e(k)$.

3. Methodology
Two audio files of Waveform Audio File Format (wav.) are used as input for the simulation model, in which the simulation model is developed in MATLAB software environment. These two inputs are of single channel (Monophonic sound reproduction) and with the same sampling rate (44100 Hz). Input $y_1$ is the audio signal of a noise corrupted speech signal (combination of speech and engine noise sound) and Input $y_2$ is the audio signal of the noise signal (engine noise sound). Another audio signal which is the speech sound signal (noise-free speech signal) is used as reference signal for comparison with the filtered output of the algorithm, $e(k)$. The algorithm is initiated by setting the column weight vector $W(k)$ and the filtered output $e(k)$ as a zero column vector as shown equation (4) and equation (5) respectively.

$$W(k) = \begin{bmatrix} W_0(k) \\ W_1(k) \\ \vdots \\ W_{L-1}(k) \end{bmatrix} = \begin{bmatrix} 0 \\ 0 \\ \vdots \\ 0 \end{bmatrix}$$

(4)

$$e(k) = \begin{bmatrix} e_0(k) \\ e_1(k) \\ \vdots \\ e_{L-1}(k) \end{bmatrix} = \begin{bmatrix} 0 \\ 0 \\ \vdots \\ 0 \end{bmatrix}$$

(5)

The initial value for the $\mu$ is taken as any random number with the condition of $0<\mu<1$ [14]. Both the Input $y_1$ and Input $y_2$ then inserted to the developed LMS adaptive filter algorithm. Input $y_2$ is multiplied by the column weight vector $W(k)$ to generate a weighted-noise signal ($y_2^\prime$). The intermediate output $y_2^\prime$ is then subtracted from the Input $y_1$ to obtain the error $e(k)$. After each interaction, the column weight vector would adaptively updates. The filtering process would be continuing until it reaches the prescribed filter length L, which is same as the length of the inputs. The absolute difference between the filtered output $e(k)$ of the LMS adaptive filter algorithm and the input $y_1$ (noise corrupted speech signal) are calculated. Fast Fourier Transform (FFT) is applied on both $e(k)$ and $y_1$ to obtain the frequency function of the output. The percentage of difference for the reference signal, noise corrupted speech signal and filtered signal are calculated to compare the recovery of the speech signal from the noise corrupted speech signal upon implementing the LMS adaptive filter algorithm. Figure 2 shows the idea of the LMS adaptive filter algorithm in filtering the engine noise from the speech signal.
4. Results and discussions
The simulation using the LMS adaptive filter algorithm attenuate the noise gradually and finally cancelled the engine noise in the noise corrupted speech signal. The results show the clean speech signal is obtained after the LMS adaptive filter is applied. The obtained clean speech signal shows low deviation with the original-free speech signal. Figure shows the signal waveform of the noise-free speech signal, engine noise signal and the noise-corrupted speech signal that had been used in this study. The signal waveform in Figure 3(a) is the noise-free speech signal in which the signal amplitude that approaches zero at intervals indicates a momentary pause in between the words during the speech. Figure 3(b) is the engine noise signal where the waveform pattern indicates that the engine noise signal can be generally classified as periodical noise due to the repetitive waveform. Figure 3(c) is the input for the simulation model which is the noise corrupted speech signal. This signal consists of the fluctuation caused by the engine noise signal and the speech signal is also altered by the noise signal. From Figure 3(c), it is difficult to identify the speech signal by analyzing the waveform.

![Flow chart of the LMS adaptive filter algorithm implementation.](image)

**Figure 2.** Flow chart of the LMS adaptive filter algorithm implementation.

![Signal waveform images](image)

**Figure 3(a).** Noise-free speech signal **Figure 3(b).** Engine noise signal and **Figure 3(c).** Noise corrupted speech signal.
Figure 4 shows the comparison in between the noise corrupted speech signal and the LMS adaptive filter algorithm filtered output. It is clear that the implementation of the LMS adaptive filter algorithm on the noise corrupted speech signal reduces the engine noise signal significantly. The amplitude and the noise of the signal are reduced. This filtered output single will then compare with the original clean speech signal (noise-free speech signal). This comparison shows that the implementation of the LMS adaptive filter algorithm able to attenuate the engine noise which is classified as periodical noise effectively.

![Figure 4(a). Noise corrupted speech signal and Figure 4(b). Filtered output.](image)

Figure 5 below shows the comparison of the filtered signal with the original clean speech signal (noise-free speech signal). From Figure 5(c), the absolute difference between the noise-free reference signal and the filtered signal decrease with time. High difference is found at the initial filtering process. This is because at the beginning stage, the column weight vector is set to be a zero column vector. The algorithm is unable to approximate the noise signal accurately and thus results in higher error in between the original clean speech signal and the filtered signal. As the algorithm continues, the column weight vector adaptively adjusts the weight values to approximate the noise signal and thus the difference in between two signals reduced along with the iterations. The reduction implies that the LMS adaptive filter algorithm has recovered the speech signal from the noise-corrupted speech signal. The signal obtained from the simulation is folded by using Fast Fourier Transform as negative frequencies do not exist in practical. The frequency function is obtained from 0-5000Hz. The percentage difference base on the standard deviation in between the filtered signal and the original clean speech signal is 8.9% for this frequency range. The small percentage is similar with the finding of Mendiratta and Jha (2014) where the noise was suppressed with little signal distortion [15].

![Figure 5(a). original clean speech signal Figure 5(b). Filtered signal and Figure 5(c).](image)

Difference between original clean speech signal and filtered signal.
5. Conclusion
In this paper, the simulation model of noise cancellation using LMS adaptive filter algorithm has been presented. The simulation model shows that the LMS adaptive filter algorithm is capable to update the column weight vector adaptively in approximating the noise signal to be cancelled. The implementation of LMS adaptive filter algorithm on the noise corrupted speech signal has proven that the adaptive filter can effectively remove the noise signal at lower frequencies range (0Hz to 5000Hz). The declination in the difference between the reference signal and the filtered signal has proven that speech signal is recovered from the noise corrupted speech signal. The filtered speech signal recovered from the noise corrupted speech signal has low percentage difference which is 8.9% compared to the noise-free speech signal. This shows the performance of the LMS adaptive filter in filtering the noise signals.

References
[1] Brett RCM, Marion B and Daniel K 2013 The use of noise cancelling headphones to improve concurrent task performance in a noisy environment Applied Acoustics 74(1) pp 110-15
[2] Kim N, Toon VW, Mads GC and Marc M 2013 Improved prediction error filters for adaptive feedback cancellation in hearing aids Signal Processing 93(11) 3062-75
[3] Kajikawa Y, Gan WS and Sen MK 2012 Recent advances on active noise control: open issues and innovative applications APSIPA Transactions on Signal and Information Processing 1(3)
[4] Chen X and Yijun M 2016 Passive control of centrifugal fan noise by employing open-cell metal foam Applied Acoustics 103 Part A pp 10-9
[5] Hu S, Rajamani R and Yu X 2013 Directional cancellation of acoustic noise for home window applications Applied Acoustics 74(3) 467-77
[6] Wu L, Qiu X and Guo Y 2014 A simplified adaptive feedback active noise control system Applied Acoustics 81 pp 40-6
[7] Wesson KD, Ochshorn RM and Land BR 2009 Low-cost, high-fidelity, adaptive cancellation of periodic 60 Hz noise J Neurosci Methods 185(1) pp 50-5
[8] Widrow B and Hoff ME Adaptive switching circuits IRE WESCON Convention Record 4 pp 96–104
[9] Hamidia M and Amrouche A 2016 Improved variable step-size NLMS adaptive filtering algorithm for acoustic echo cancellation Digital Signal Processing 49 pp 44-55
[10] Sukhpreet S and Sukhwinder SS 2012 Acoustic Noise Cancellation Using Block LMS Filter In MATLAB SIMULINK International Journal of Computer Science 3(1) pp 183-186
[11] Cordourier-Maruri HA and Orduna-Bustamante F 2009 Active Control of Periodic Fan Noise in Laptops: Spectral Width Requirements in Delayed Buffer Implementation Journal of Applied Research and Technology 7(2) pp 124-35
[12] Chang DC and Chu FT 2013 Active noise cancellation with a new variable tap length and step size FXLMS algorithm IEEE International Conference on Multimedia and Expo
[13] Yang IH, Jeong JE, Jeong UC, Kim JS and Oh JE 2014 Improvement of noise reduction performance for a high-speed elevator using modified active noise control Applied Acoustics 79 pp 58-68
[14] Apolin JA, Diniz PSR, Laakso TI and Campos MLR 1998 Step-size optimization of the BNDR-LMS algorithm 9th European Signal Processing Conference
[15] Mendiratta A and Jha D 2014 Adaptive Noise Cancelling for audio signals using Least Mean Square algorithm International Conference Electronics Communication and Instrumentation (ICECI)

Acknowledgment
The author would like to gratefully acknowledge Universiti Sains Malaysia (USM) Short Term Grant [account no. 60313028] for the financial funding on this project.