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Simulation for noise cancellation using LMS adaptive filter

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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:

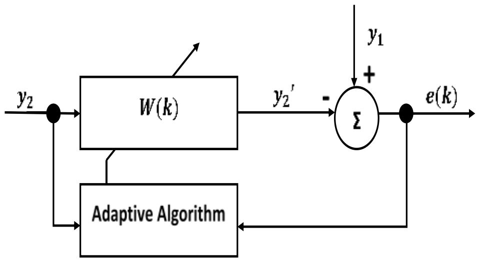

Figure 1. LMS adaptive filter algorithm.

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

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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' 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.

$$\begin{aligned} e(k) &= y_1(k) - y_2'(k) \\ &= y_1(k) - [W^T(k) y_2(k)]^T \end{aligned} \tag{2}$$

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

$$W(k+1) = W(k) + \mu e(k) y_2(k) \tag{3}$$

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 in equation (4) and equation (5) respectively.

$$W(k) = \begin{bmatrix} W_0(k) \\ W_1(k) \\ \vdots \\ W_{L-1}(k) \end{bmatrix} = \begin{bmatrix} 0 \\ 0 \\ \vdots \\ 0 \end{bmatrix} \tag{4}$$

$$e(k) = \begin{bmatrix} e_0(k) \\ e_1(k) \\ \vdots \\ e_{L-1}(k) \end{bmatrix} = \begin{bmatrix} 0 \\ 0 \\ \vdots \\ 0 \end{bmatrix} \tag{5}$$

The initial value for the μ 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'). The intermediate output y_2' 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.

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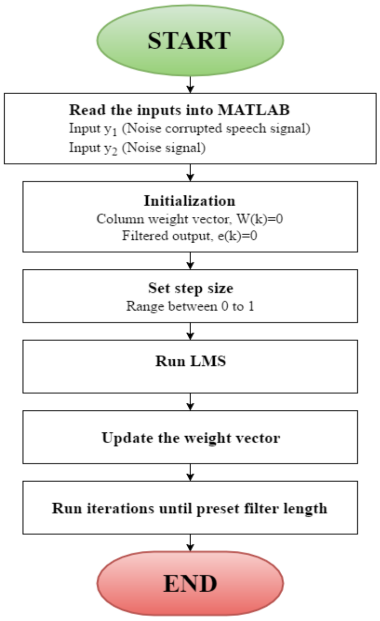

Figure 2. Flow chart of the LMS adaptive filter algorithm implementation.

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.

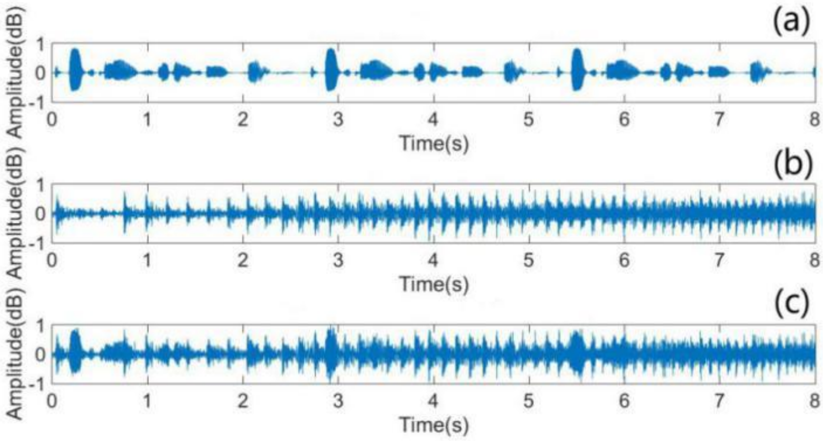

Figure 3(a). Noise-free speech signal **Figure 3(b).** Engine noise signal and **Figure 3(c).** Noise corrupted speech signal.

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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.

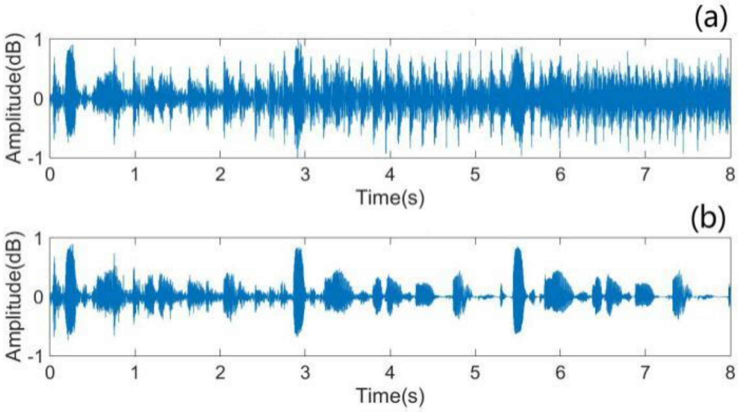

Figure 4(a). Noise corrupted speech signal and **Figure 4(b).** Filtered output.

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].

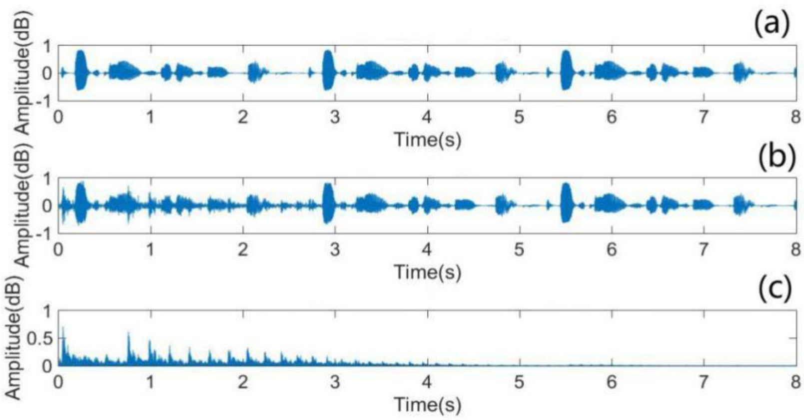

Figure 5(a). original clean speech signal **Figure 5(b).** Filtered signal and **Figure 5(c).** Difference between original clean speech signal and filtered signal.

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.

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