ADAPTIVE NOISE CANCELLATION BY LMS ALGORITHM

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To my husband Abd Kadir Bin Mahamad,
My child's,
Mohamad Azri and Nur Aliah

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ABSTRACT

The research on controlling the noise level in an environment has been the focus of many researchers over the last few years. Adaptive noise cancellation (ANC) is one such approach that has been proposed for reduction of steady state noise. In this research, the least mean square (LMS) algorithm using MATLAB was implemented. Step size determination was done to determine the best step size and effects of the rate of convergence. Sound recorder was used to record sound and saved as .wav file. Graphical user interface (GUI) was created to make it user friendly. The output of the analysis showed that the best step size was 0.008. Smaller step size of 0.001 tend to lower the speed of convergence, and too big a step size, 0.8 tend to cause the system to diverge. Analysis on synthesized data showed that the noise reduction did not eliminate the original signal. The implementation on actual data showed slight difference between the output and input level. In real situation, as in theory, this technique can be used to reduce noise level from noisy signal without reducing the characteristic of the signal.

ABSTRAK

Penyelidikan terhadap pengawalan paras kebisingan dalam persekitaran telah menjadi fokus penyelidikan beberapa tahun kebelakangan ini. Penyesuaian penghapusan kebisingan (ANC) adalah salah satu pendekatan yang dapat mengurangkan keadaan tetap kebisingan. Dalam penyelidikan ini, algoritma purata kuasa dua terkecil (LMS) menggunakan MATLAB digunakan. Penentuan saiz langkah dilakukan untuk menentukan saiz langkah yang terbaik dan kesannya terhadap kadar penumpuan. Perakam bunyi digunakan untuk merakam bunyi dan disimpan sebagai .wav fail. Antaramuka pengguna bergambar (GUI) direka bagi menjadikannya mesra pengguna. Hasil analisis yang diperolehi, didapati saiz langkah yang terbaik adalah 0.008. Saiz langkah yang lebih kecil, 0.001 menyebabkan kadar penumpuan menjadi perlahan dan bagi saiz langkah yang terlalu besar, 0.8 sistem akan mencapah. Analisis keatas data yang direka menunjukkan pengurangan bising tanpa menjejaskan isyarat asal. Perlaksanaan data sebenar menunjukkan hanya sedikit perbezaaan diantara paras isyarat keluaran dan masukan. Dalam situasi sebenar, secara teorinya teknik ini mampu untuk mengurangkan paras bising dari isyarat tanpa mengubah ciri isyarat tersebut.

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LIST OF SYMBOL AND ABBREVIATION

ANC	–	Adaptive noise cancellation
LMS	–	Least mean square
n	–	Time
LMS-AP	–	Augmented predictor LMS
MLMS-AP	–	Modified LMS-AP
SPR	–	Strictly positive real
FIR	–	Finite impulse respond
RLS	–	Recursive least square
AR	–	Autoregressive
ARMA	–	Autoregressive moving average
GAL	–	Gradient adaptive lattice
$d(n)$	–	Desired signal
$x(n)$	–	Reference signal
$y(n)$	–	Output of adaptive filter
$e(n)$	–	Error signal
IIR	–	Infinite impulse response
MSE ($\xi(n)$)	–	Mean square error
$\mathbf{p}$	–	Cross-correlation matrix
$\mathbf{R}$	–	Input correlation matrix
L	–	Filter order
μ	–	Step size
MSD	–	Mean square deviation
SNR	–	Signal to noise ratio
GUI	–	Graphical user interface

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CHAPTER I

INTRODUCTION

1.1 Background

Acoustic problems in an environment has gained more attention due to the tremendous growth of technology that lead to noisy engines, heavy machineries, pumps, air condition, music and other noise sources. These acoustic problems sometime can disturb the neighbours next door. Normally human ears are very sensitive at audio range (lower frequency) from 20 Hz to 20 kHz, even though it depends on the age and physical condition of a person. So, any sound within these frequencies has the tending to disturb human hearing and can be classified as noise.

The reduction of acoustic noise in speech has been investigated for many years [3]. The major application of noise reduction is by improving voice communication at noisy sites using noise cancelling microphones [1]. In these microphones, the near-field response is independent of frequency and the far-field response is similar to high-pass frequency. Another technique is by using a single input that exploits the noise model. The noise model is estimated when speech is absent.

In such situation, the approach of *adaptive noise cancellation* (ANC) is applicable. ANC, also called noise reduction is one of such approach that has been proposed for reduction of steady state noise [1]. ANC technique employs two inputs, the *primary input* (speech corrupted by noise) and the *reference input* (noise alone) [1, 2]. The dual input approach tries to estimate the differential path characteristics from the noise source to the primary and reference input.

There are many algorithms that can be use for adaptive filter in ANC, but the simplest and effective algorithm for the operation of adaptive filter is least mean square (LMS) algorithm [10]. The LMS algorithm is a stochastic gradient algorithm that iterates each tap weight of a transversal filter in the direction of the gradient of mean square error of an error signal. The LMS algorithm uses a fixed step size parameter to control the correction applied to each tap weight from one iteration to the next.

Adaptive filters are used for non-stationary signals and environments. Applications of adaptive filters include system identification, layered earth modeling, predictive coding, adaptive noise cancellation, multi channel reduction, radar/solar signal processing, channel equalization for cellular mobile phone and echo cancellation. It consists of two parts, digital filter and adaptive algorithm. Digital filter is used to perform the desired signal processing and the adaptive algorithm is used for adjusting the coefficients or weights of the filter and to minimize the mean square value.

1.2 Problem Statement

Adaptive noise cancellation with *least mean square* (LMS) algorithm is one of the most popular algorithms to solve many problems [7].. Its popularity comes from its ability to perform well for both static and dynamic noise disturbances, easy to implement and effective to use. This adaptive process, mean it does not require knowledge of signal or noise characteristic.

To record the sound recording of an air condition in a room producing unwanted noise, the sound recorder must filter out other disturbances. An adaptive filter can trace that noise, and reduce it so that it can produce suitable sound recording.

Base on this situation, a filter code with least means square (LMS) algorithm using MATLAB was through to be able to eliminate or reduce periodic noise from audio signal and was implement in this project. The filtered audio signal was recorded and so the clean signal could be played back.

1.3 Research Objectives

The project was to implement the least means square (LMS) algorithm using MATLAB for noise reduction level in audio signal.

1.4 Scope of Research

The scopes of the research:-

1. Writing LMS algorithm using MATLAB.
2. Generate model for system identification to determine the suitable step size.
3. Data is an audio signal collected in closed room, using sound recorder.
4. Data will be simulated using LMS algorithm to reduce noise in an audio signal.
5. Implement an existing code to reduce periodic noise in audio signal.

1.5 Report Organization

The next chapter will discuss on reviews of different approaches to the noise cancellation problem. Chapter III will discuss the theoretical parts of adaptive noise cancellation, adaptive filter and algorithm used in this project, the derivation and equations involved in LMS algorithm. Then proposed adaptive noise cancellation by LMS algorithm scheme will be discussed and the methodology of this proposed will be elaborated in Chapter IV.

The main objective of this project is to implement LMS algorithm using MATLAB that can reduce the noise from the noisy signals. Chapter V discusses the graphical user interface that was created for the research and results of the simulation. The conclusion of this project and the recommendation for future work are highlighted in the last Chapter of this report.

CHAPTER II

LITERATURE REVIEW

2.1 Review

The initial work on adaptive echo cancellers started around 1965[10]. It appears that Kelly of Bell Telephone Laboratories proposed the echo cancellation using adaptive filter, where the speech signal itself was utilized in performing the adaptation [12]. In 1975, Widrow *et al* has originated the adaptive line enhancer to cancel 60-Hz interference at the output of an electrocardiography amplifier and recorder. The adaptive echo cancellers and adaptive line enhancer is an example of the adaptive noise cancellation.

Research on adaptive filter started earlier than adaptive noise cancellation, that is around 1950s. The least mean square (LMS) algorithm was one of the adaptive filter devised by Windrow and Holf in their study of pattern-recognition scheme, known as the adaptive linear element. Robbins and Monro (1951) highlighted that the LMS algorithm was closely related to the concept of stochastic approximation. The difference between LMS and the stochastic approximation was the usage of step size. The LMS algorithm uses a fixed step-size parameter to control the correction applied to each tap weight for each iteration, but in stochastic approximation methods the step size parameter is inversely proportional to time n or to a power of n.

2.2 Related Research

Adaptive noise cancellation with the LMS algorithm has become a popular solution to the noise canceller. Orgen A.C., *et al* [1] proposed two algorithms to improve steady state residual noise. These two algorithms are LMS algorithm with augmented predictor (LMS-AP) and modified LMS-AP (MLMS-AP). Both the algorithms depend on strictly positive real (SPR) whitening filter. SPR condition uses error filtering to ensure stability and convergence. SPR is a new approach of a whitening mechanism, and it manifests the signal processing task in a number. If the whitening filter is SPR, the residual variance provided by MLMS-AP is larger than that given by LMS-AP, although lower than the LMS. For non-SPR whitening filter, LMS-AP is divergence and MLMS-AP performs at least as well as LMS algorithm.

Ho K. C. and Ching P. C. [2] introduced the new structure for adaptive noise cancellation to improve the convergence characteristics. They have devised the split-path adaptive filters (split canceller) as illustrated in **Figure 2.1** by splitting the original finite impulse respond (FIR) into two linear phase filter connected in parallel [2]. Both filter uses LMS algorithm to minimize the system output error and adapted independently to obtain the performance of the overall system. This method has successfully improved the convergences speed by almost two times. It required around 500 iterations compared to 1000 iterations done by LMS algorithm.

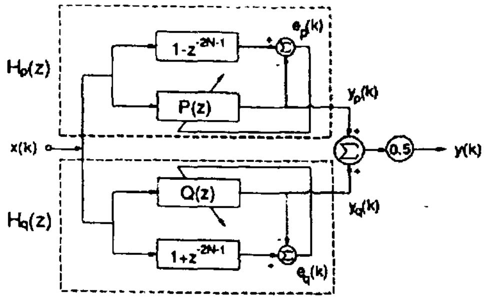

Figure 2.1: The split noise canceller.[2]

Kang G. S. and Fransen L. J. [3], developed an adaptive noise cancellation filter for real time operation using TMS32010 microprocessor. To make the real time noise suppression as simple as possible, the homogeneous adaptation was used. The homogeneous adaptation was an identical convergence factor for all coefficients in each filter. The homogeneous adaptation has slow convergence speed and for this application, slow adaptation was permissible because it was done before the beginning of voice communication. The filter was tested at the noise level of 111 dB, using recorded noise samples and the successful noise reduction ratio was about 10 to 15 dB. They used spectrograms to demonstrate the adaptive noise cancellation results. Spectrogram is one of the time-frequency distributions.

Pulsipher D., Boll S. F., Rushforth C. and Timothy L. [6] studied on LMS adaptive noise cancellation to reduce the non-stationary acoustic noise in speech. The primary noise added to the speech was reduced by subtracting the reference noise. The reference noise filter was updated using LMS algorithm. The effectiveness of noise suppression depends directly on the ability of the filter to estimate the transfer function relating to the primary and reference noise channels. They also presented a study of a desired noise reduction level in a hard-walled room.

There were also many researches done on adaptive noise cancellation in the time domain. Ball S. F. [4] implements the frequency domain approach to the noise canceling procedure using short time Fourier transform. By using this method a computational rate is proportional to the log of the filter length. For acoustic noise suppression where the filter length can be on the order of one thousand points, this approach offers a viable alternative for real time implementation and suitable for the long filter lengths required for acoustic noise reduction. **Table 2.1** shows a comparison between time and frequency domain noise cancellation.

Table 2.1: Opposition and similarity of time domain and frequency domain adaptive noise cancellation [4]

Opposition		
	Time Domain	Frequency Domain
Computational requirement per sample	Linear with filter length	Proportional to the log of the filter length
Real time implementation	Limited, because of a linear computational relationships	Limited, but the efficiency of the FFT can be used to reduce the computing requirements
Feedback to update the filter	Required	No
Similarity		
Convergence characteristic of the frequency domain is equivalent to the time domain methods.		

Zhigang Feng, Xizhi Shi and Huang H. [5] managed improved the limitation of noise cancellation LMS methods. This method, known as modified LMS (MLMS) algorithm, was done by introducing the delay between reference noises. A noise cancellation by a filter of lower order can be achieved compared to LMS algorithm MLMS speeded up the convergence rate and reduced the amount of calculation.

Equation below shows the output and the next weight vector of the linear adaptive filter for LMS algorithm,

$$y_j = \sum_{i=1}^{N} W_i X_{ij} \tag{2.1}$$

$$\mathbf{W}_{j+1} = \mathbf{W}_j + \mu e_j \mathbf{x}_j \tag{2.2}$$

Consider the delay than the corresponding equation is modified as Eq. (2.3) and Eq. (2.4) and are called the MLMS algorithm.

$$y_j = \sum_{i=1}^{N} W_i X_{ij} - N_0 \tag{2.3}$$

$$\mathbf{W}_{j+1} = \mathbf{W}_j + \mu e_j \mathbf{x}_{j-N_0} \tag{2.4}$$

where N_0 is a delay.

Abdullah A. H., Yusof. M. I. And Zaki S. R. M. [7] have analysed two type of adaptive noise cancellation algorithms, the LMS algorithm and the recursive least square (RLS) algorithm, and outlined the strange and weakness of both algorithms. RLS have faster convergence; they show that RLS has convergence at one taps compared to 25 taps of LMS algorithm. At the same 25 taps, LMS show an error level around 2.5 % but the error level was 0 % for RLS algorithm. Error level is a residual noise, but RLS implementation is more complicated than LMS algorithm.

Abutaleb A. S. [8] introduced a nonlinear filter that was used for adaptive noise cancelling. The performance of this filter was measured by the signal to noise ratio between the signal and its estimate. This filter was based on Pontryagin minimum principle and the method of invariant imbedding. The filter assumed that the interference at the primary input was the output of a linear time varying filter driven by the reference input, and white noise with two unknown time functions that described the filter. Estimation of the time varying functions was achieved by modeling them using autoregressive (AR) or autoregressive moving average (ARMA), as controllers which drived the system to follow a noisy observation. The Pontryagin minimum principle was employed which resulted in a two point boundary value problem. The method of invariant imbedding was then used to transform the problem to initial value problem that can be solved in real time.

CHAPTER III

THEORY

In a general sense, adaptive filters are systems that vary with time because the characteristics of its input may be varying. That is what separates it from classical digital signal processing; the digital system itself changes through time.

Therefore, adaptive filters must be non-linear because superposition does not hold, but when their adjustments are held constant after adaptation, then some can become linear and belong under the well-named class of linear adaptive filter.

3.1 Application of Adaptive Filter

The ability of an adaptive filter to operate in an unknown environment and track time variations of the input signal makes it a powerful device for signal processing and control applications. In general, there are four basic classes of adaptive filtering application: identification, inverse modeling, prediction, and interference cancelling, as shows in **Figure 3.1** [10].

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