International Conference on Electrical, Control and Computer Engineering
Pahang, Malaysia, June 21-22, 2011

Speech Processing for Makhraj Recognition

The Design of Adaptive Filter for Noise Removal

N.W.Arshad, S.N.Abdul Aziz, R. Hamid, R. Abdul Karim, F. Naim, N. F. Zakaria
Faculty of Electrical and Electronics Engineering
Universiti Malaysia Pahang
Pekan, Pahang, Malaysia
along_lyrs@yahoo.com

***Abstract* – In our daily day, improvement of *makhraj* for Arabic alphabets is a topic that very useful in many applications and environments. The existing system cannot recognize the appropriate pronunciation of each alphabet with the existence of noise. As an example "ha", with the disturbance from the noise, the system may recognize wrong alphabet like "kho". This paper focus on noise removal in *makhraj* recognition using Least Mean Square (LMS) Algorithm based on Adaptive Filter to search for the optimal solution to adaptive filter, including system identification and noise cancellation. There are 30 Arabic alphabets from ا until ي . However, this project will only use 7 alphabets as samples that are from ا until خ. The speech processing will be used to obtain same waveform output from two different situations. The filtered data will be processed to match the standard pronunciations and it will be integrated with filter design process in MATLAB. As a result, the waveform of noise cancellation using LMS algorithm is quite similar with the waveform of reference signal. As a conclusion, it is proved that noise cancellation method remove noise from unknown system.**

***Keywords* – Adaptive filter, Least Mean Square (LMS) algorithm, Normalized Least Mean Square (NLMS) algorithm, System Identification, Noise Cancellation.**

I. INTRODUCTION

Speech recognition has been implemented in many area of research such as criminal [1], wireless communication [2], and speech therapy [3]. *Makhraj* recognition is a part of speech recognition applications. *Makhraj* is the correct position of the organs of speech in order to produce a letter so that it can be differentiated from others. This is equally so whether the letter is a consonant or a vowel. Being able to recite the letters correctly is the foundation of *tajweed*, and this is achieved by knowing where the sound originates. This can then help in practicing the pronunciation of the letters correctly. Both *makhraj* and *tajweed* are the important thing for reciting al-Quran correctly [4].

Speech recognition is a technique for converting a speech to text. In order to convert the speech, noise parameter is the most crucial factor need to consider for successful of the system. Robust adaptive filter need to design for this purpose. There have many types of adaptive filter such as Recursive Least Squares (RLS), Least Mean Square (LMS), Normalized LMS (NLMS) and etc.

RLS is deterministic models which constantly generate same output with starting condition or initial state and minimize the weighted of linear least squares cost function with concern to the input signals. Successful utilization of RLS adaptive filter was proved by [5] with increased the accuracy up to 98%. However, this filter required high computational complexity and stability problem on tracking performance [6].

This research is proposed in order to create a system that can recognize the correct *makhraj* automatically and at the same time help people to improving the pronunciation of the *makhraj.* The main purpose of this research is to remove noise from the speech of *makhraj* that is taken from variety of environment. Besides that, this system is able to differentiate between each letter. As an example *"ha",* with the disturbance from the noise, the system may recognize wrong alphabet like *"kho".*

The technique that applied is Least-Mean-Square (LMS) algorithm based on adaptive filter as shown in Fig. 1. LMS algorithm is great in noise removing and extremely popular because of its simplicity and ease of computation.

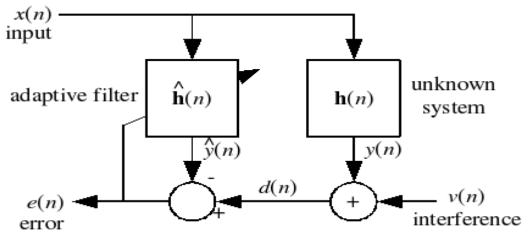


Figure 1. Least-mean-square implementation.

II. THE ALGORITHM

The LMS algorithm is a linear adaptive filtering algorithm that consists of two basic processes. Firstly is the filtering process, which involves computing the output of a transversal filtering produced by a set of tap inputs and generating an estimation error by comparing this output to a desired response. Secondly, adaptive filtering, which involves the automatic adjustment of the tap weights of the filter in accordance with the estimation error.

The computational steps for the LMS algorithm are as follows [7]:

1. Initializes the filter coefficients, $w_k(i)$ to zero.

2. At each sampling period:

a) Compute the filter output:

$$\hat{n}_k = \sum_{i=0}^{N-1} w(i).x_{k-i} \quad (1)$$

b) Calculate the error estimate:

$$e_k = y_k \hat{n}_k \quad (2)$$

c) Update the new filter weights:

$$W_{k+1} = W_k + 2\mu e_k X_k \quad (3)$$

But when the LMS algorithm looks at the error to minimize, it considers only the current error value and slow convergence (due to Eigen value spread) compared to RLS give high convergence rate [8]. LMS algorithm is based on the steepest descent method but simplifies it further by undertaking just one iteration per sample, and by calculating only an estimate of the gradient-vector but a new one $\hat{\nabla}_k$ at each sample, *k*. The constant, μ is a step-size directly affects how quickly the adaptive filter will converge toward the unknown system. If μ is very small, then the coefficients change only a small amount at each update, and the filter converges slowly. But if μ is too large, the coefficients may change too fast and the filter will diverge [9]. Even RMS and LMS are capable to solve actual problem [10], the usefulness and selection of suitable adapter filter still depends on the applications.

NLMS algorithm is an improvement from LMS algorithm. LMS performances are poor if the signal conveys high amplitude variations because it is sensitive to the step size scaling. NLMS overcome this problem by normalize the step size, μ with the power of the input to guarantees stability of the algorithm. Based on [11], it is proven that NLMS performs better than LMS in terms of convergence rate, but however produces similar results in terms of bit error rate (BER).

III. MAKHRAJ RECOGNITION

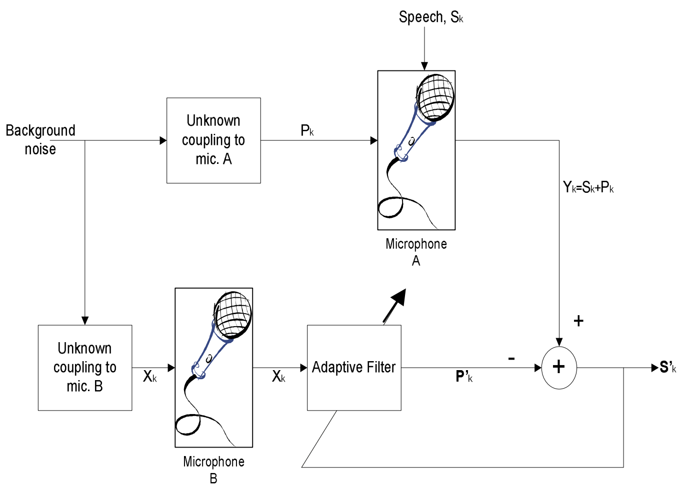


Figure 2. Noise canceller system for *makhraj* recognition.

In this paper, we present a system that consists of two input signals, which are the background noise and speech of *makhraj*. Fig. 2 depicts the noise canceller for *makhraj* recognition. Microphone A will pick-up the *makhraj* pronunciation, S_k and background noise, P_k while microphone B will pick-up the background noise, X_k that is correlated with that picked by microphone A. The filter works to adjust the signal from microphone B to be as close a replica as possible to that contain in the speech.

As long as the background noise remains correlated to the unwanted noise accompanying the desired signal, the adaptive filter adjusts its coefficients to reduce the difference between output signal, S'_k and desired signal, S_k, hence removing the noise and produce a signal.

Table I summarized the steps required for adaptive noise cancellation scheme using NLMS [6].

TABLE I. NLMS ALGORITHM

Noise estimation:
$P'_k = \sum_{i=0}^{N-1} w_k(i).x_{k-i} \quad (4)$
N- filter order
Error estimation:
$S'_k = y_k - P'_k \quad (5)$
Coefficients update:
$w_{k+1}(i) = w_k(i) + \mu \dfrac{S'_k X_{k-i}}{\sum_{i=0}^{N} X_{k-i}^2} \quad (6)$
for $i = 0$ to N-1

IV. RESULT AND ANALYSIS

In order to test the system efficiencies, 140 samples have been record from male and female students on age of 22 years old. All samples have been check by the expert as a correct *makhraj* pronunciation. For each letter 20 samples are acquired with 10 samples for each noise background. Fig. 3 shows the noisy signal of Y_k. This signal contains the uncorrelated noise component, which should exist in the recording environment.

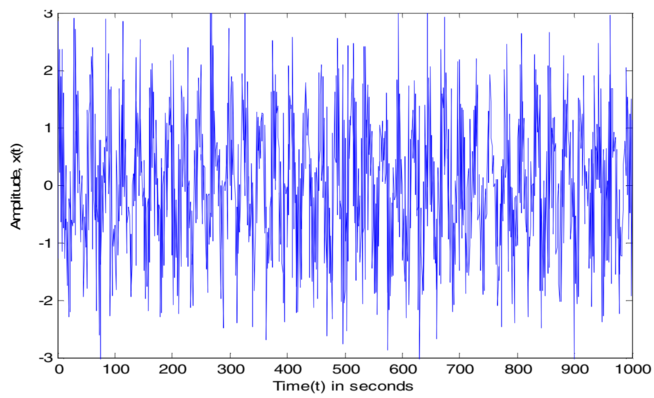


Figure 3. Waveform of original speech added with noise, Y_k.

After corrupt the desired signal to create a noisy signal, a signal need to be complete by requires a reference signal as shown in Fig. 4.

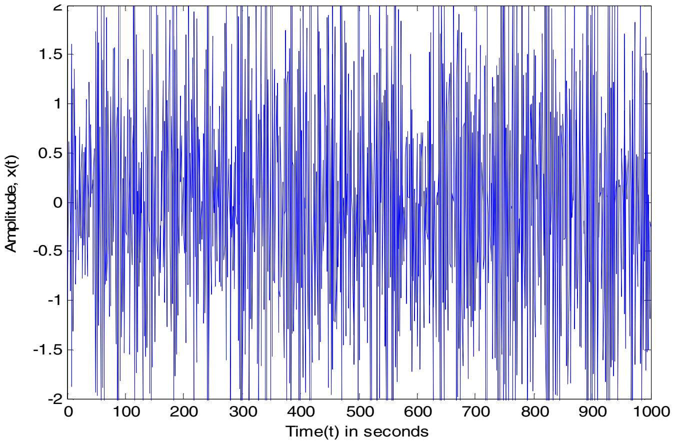


Figure 4. Waveform of reference signal.

Filter coefficients is find for an adaptive filter. Actual coefficients can be compared to the coefficients found by adaptfilt.lms after run the object, desired signal, and the input to the filter, that contain the noisy signal. Fig. 5 shows the filtered signal after noise cancellation process.

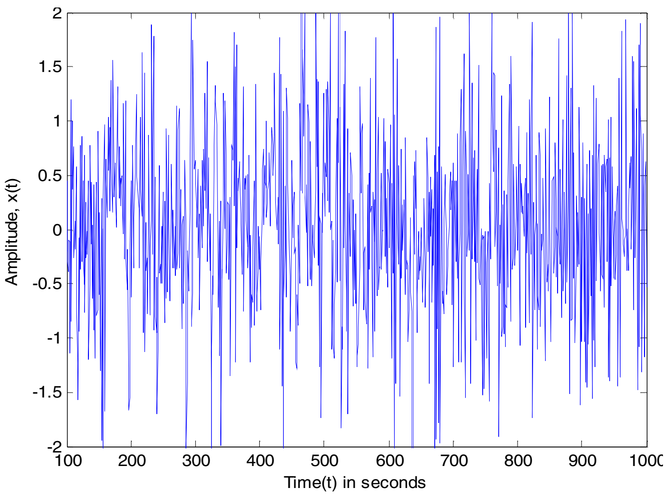


Figure 5. Waveform of filtered signal.

After noise cancellation process, the output waveform shows a good result and therefore makes this system reliable. Fig. 6 and Fig. 7 show the comparison between filtered signal and reference signal after applying adaptive filters. There are two different result of waveform after filter the signal that uses two samples of alphabet *"alif"*. These two original *alif* is use as a sample to filter noise that has been recorded at the crowded place. Filtered waveform for these samples produces similar result. The similarities of the waveform are more obviously seen through their accuracy. The accuracy that gets for sample *alif1* is 100% and for sample *alif2* is 99%.

This system also tested using different background noise which are recorded at two different place. Fig. 6 and Fig. 7 show two samples of alphabet "*alif*" that use as a sample to filter noise that has been recorded at the road and crowded place. From the two waveform of filtered noise, it shows that these two waveforms are quite similar but have a slight different in its amplitude.

The accuracy of this system is check using signal-to-noise ratio (SNR). Overall performances for letter *alif* are shown in Table II. We can say that this proposed system achieve at least 97% accuracy for *alif* with different background noise. So, it is proved that this method can filter any noise background to get the needed waveform.

TABLE II: SYSTEM ACCURACY FOR ALPHABET ALIF AT CAFETERIA AND ROAD

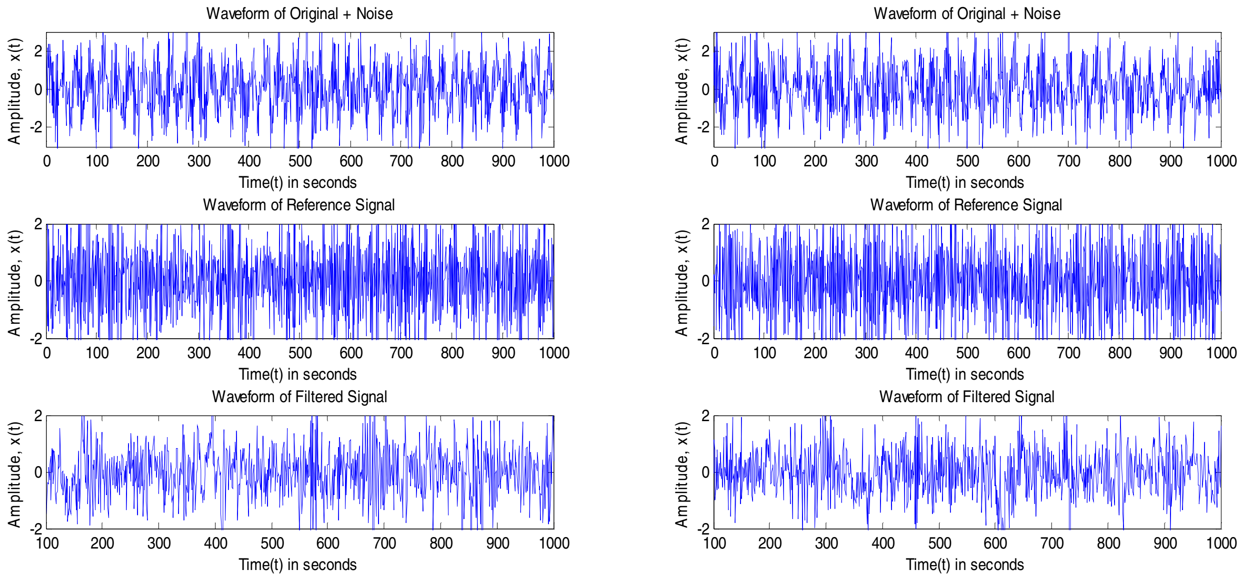
Alphabet ***aliff***	Accuracy (%) (Cafeteria)	Accuracy (%) (Road)
Alif1	98.9	99
Alif2	99.6	100
Alif3	100	99
Alif4	100	98
Alif5	100	97
Alif6	99.5	99
Alif7	99.6	99
Alif8	99.5	97
Alif9	99.6	100
Alif10	98.8	100

From the result in Fig. 6 and 8 we can analyze the performance of the adaptive filter in convergence speed, steady state error and stability.

Convergence is the process of minimizing the power of the error signal by adjusting the filter coefficients and fast convergence indicates that the adaptive filter takes a short time to calculate the appropriate filter coefficients that minimize the power of the error signal [12]. Convergence speeds can be known by displaying the error signal and it also affected by the step size and the filter length. Filtered signal in Fig.6 shows the convergence speed of this algorithm is not too fast and not too slow by adjusting the step size, μ. For $\mu = 0.8$ is a good compromise between being large enough to converge well within the 1000 iterations and small enough to create an accurate estimate of the unknown filter.

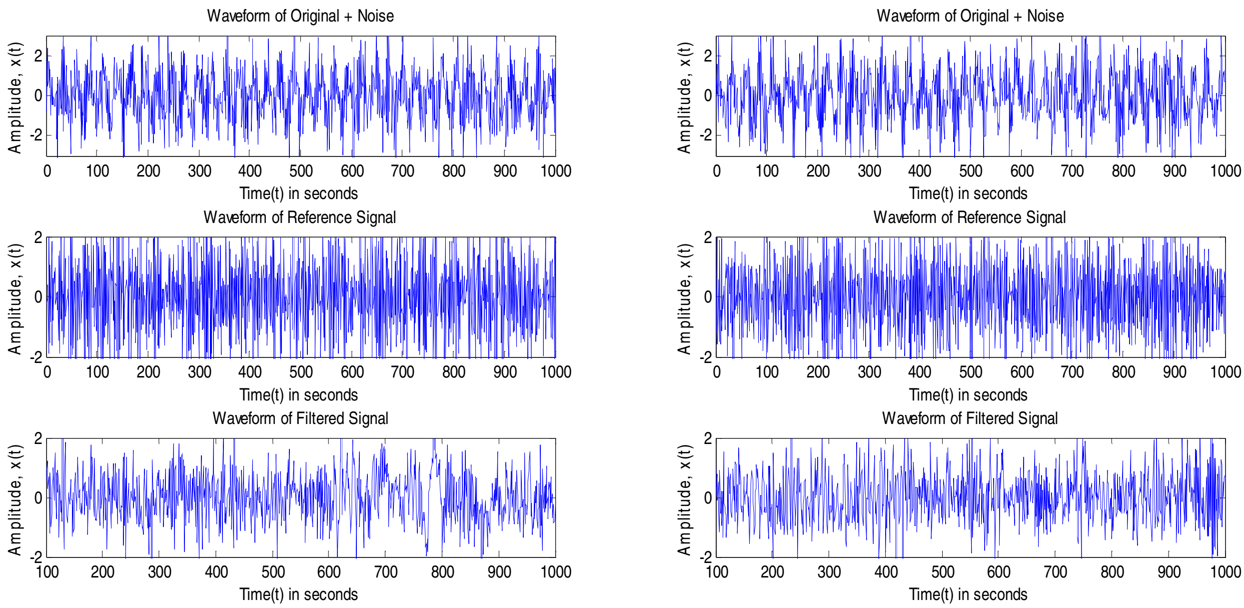
TABLE III: SYSTEM ACCURACY FOR ALPHABET AT CAFETRIA AND ROAD

Alphabet	Accuracy (%) (Cafeteria)	Accuracy (%) (Road)
Alif	90	100
Ba	98	90.9
Ta	96.8	100
Tsa	97.2	99.8
Jim	99	93
Ha	100	100
Kho	98	100



(a) Sample 1 (*alif1*) (b) Sample 2 (*alif2*)

Figure 6. Result of filtering alphabet *"alif"* from crowd noise.



(a) Sample 1 (*alif1*) (b) Sample 2 (*alif2*)

Figure 7. Result of filtering alphabet *"alif"* from road noise.

Overall system performance for all letter are shown in Table III. Alphabet a*lif* taken in crowded background give the lowest accuracy, 90% while alphabet *ba* taken nearby road as a background noise give the accuracy about 90.9%. Alphabet *ha* shows the highest accuracy for both different backgrounds.

V. Conclusion

The developed software in this project is successfully able to remove noise in road and cafeteria backgroundThe system is designed based on Least Mean Square (LMS) algorithm technique on adaptive filter is able to remove noise from human voice that produces filtered speech *makhraj* recognition. A comparison between noisy and filtered waveform can be seen in Fig. 6 and Fig. 7. The accuracy calculated in Table II shows the small error between noise and desired signal. The accuracy is also compared between each samples of alphabet as shown in Table III. Therefore, it is concluded that we can develop a *makhraj* recognition software using Adaptive Filter by using MATLAB environment.

LMS algorithm is relatively simple to implement and stable. It is powerful enough to evaluate the practical benefits that may result from the application of adaptivity to the problem at the hand. For input signals that changes slowly over time.

For further research we can use NLMS algorithms for more efficient LMS approach. Besides that, more samples are needed especially to challenge this system using different *makhraj* pronunciation to make sure it has a robust performance against different signal conditions.

REFERENCE

[1] N. Zheng and X . Li, "*A Robust Keyword Detection System for Criminal Scene Analysis,*" IEEE Trans. on Industrial Electronics and Applications, pp. 2127-2131, 2010.

[2] Weerackody, V. Reichl, W. Potamianos and A., *"An Error-Protected Speech Recognition System for Wireless Communications,"* IEEE Trans. on Wireless Communications, vol. 1, pp. 282-291, 2002.

[3] Georgopoulos, V.C. "*An Investigation of Audio-Visual Speech Recognition as A Applied to Multimedia Speech Therapy Applications,"* IEEE Trans. on Multimedia Computing and Systems, vol. 1, pp. 481-486, 1999.

[4] The Makhaarij of the Letter [Online]. Available: http://www.readwithtajweed.com/tajweed_Makhaarij.htm

[5] S.A.R. Al-Haddad, S.A. Samad, A. Hussain, K.A. Ishak and A.O.A. Noor, "*Robust Speech Recognition Using Fusion Techniques and Adaptive Filtering*," American Journal of Applied Sciences 6 (2), pp. 290-295, 2009.

[6] Georgi Iliev and Nikola Kasabov,**"***Adaptive Filtering with Averaging in Noise Cancellation for Voice and Speech Recognition*," IEEE Trans. on Acoust., Speech, Signal Processing. Jan. 2002.

[7] Digital Signal Processing, *"Adaptive Filtering Module",* Newcastle University.

[8] Khan. S, Arif. M and Majeed. T, *"Comparison of LMS, RLS and notch based adaptive filter algorithms for noise cancellation of a Typical Industrial Workroom,"* IEEE Trans. on Multitopic International, pp.169-173, 2004.

[9] Adaptive Filtering: LMS Algorithm [Online]. Available: http://cnx.org/content/m10481/latest/

[10] Abdullah. A.H, Yusof. M.I, Baki.S.R.M, *"Adaptive Noise Cancellation: A Practical Study of the Least-Mean Square (LM) over Recursive Least-Square (RLS) Algorithm,"* IEEE Trans. on Research and Development, pp. 448-452, 2002.

[11] Joy, Arun, Chakka and Vijaykumar, *"Performance Comparison of LMS/NLMS based Transceiver Filters for MIMO Two-Way Relaying Scheme,"* IEEE Trans. on Communications and Signal Processing, pp. 105-107, 2011.

[12] Filter Design Toolbox, "*Noise or Interference Cancellation"*, MATLAB Help Directory.