2321-8045 Science Park Research Journal Vol-1, Issue-7, 5th Sept 2013

PRIMARY ARTICLE

Algorithmic Review for Voice and Speech Recognition.

Swapnil R. Kurkute And D. G. Agrawal

ABSTRACT

In the noisy environment there is presence of background acoustic noise such as noise produced by vehicle engine inside a vehicle. Also there is utmost requirement to ensure negligible noise components in the recorded speech signal. Due to this speech communication or recording is greatly affected.

One possible way to overcome this problem or satisfy a requirement to obtain a better recording of the desired signal is the use of a simple noise canceller. [6]

To fulfill this requirement generally adaptive algorithms are used which converge rapidly. There are two best choice as adaptive filtering i.e. recursive least squares (RLS) algorithm & normalized least mean squares (NLMS) algorithm. Unfortunately this algorithm has high computational complexity, stability & poor adaptive rate problems respectively.

Proposed algorithm is based on adaptive filtering with averaging (AFA) used for noise cancellation. The main advantages of AFA algorithm could be summarized as follows. It has low computational complexity at the same time high convergence rate comparable to that of the RLS algorithm and possible robustness in fixed-point implementations. The algorithm is illustrated on car and office noise added to speech data. [1]

Swapnil R. Kurkute And D. G. Agrawal
From
[1]Assistant Professor , Associate Professor[2]
Department of E&C Engineering[1,2]
S.S.G.B.C.O.E. &T, Bhusawal[1,2]
N.M.U, Jalgaon[1,2]
Bhusawa, India[1,2]

Te Article is published on September 2013 issue & available at www.scienceparks.in

DOI: 10.9780/23218045/152013/22

KEYWORDS :

Recursive Least Squares (RLS),Normalized Least Mean Squares (NLMS), Adaptive Filtering with Averaging (AFA)

I.INTRODUCTION

Removal of noises from respiratory signal is a classical problem. In recent years, adaptive filtering has become one of the effective and popular approaches for the processing and analysis of the respiratory and other biomedical signals. Adaptive filters permit to detect time varying potentials and to track the dynamic variations of the signals. [8]

An adaptive filter is defined as a self-designing system that relies for its operation on a recursive algorithm, which makes it possible for the filter to perform satisfactorily in an environment where knowledge of the relevant statistics is not available [4]. The purpose of this contribution is to study the application of a new algorithm based on adaptive filtering with averaging in noise cancellation problem. It is well known that two of most frequently applied algorithms for noise cancellation are normalized least mean squares (NLMS) and recursive least squares (RLS) algorithms. Considering the two algorithms, it is obvious that NLMS algorithm has the advantage of low computational complexity. On the contrary, the high computational complexity is the weakest point of RLS algorithm but it provides a fast adaptation rate. Thus, it is clear that the choice of the adaptive algorithm to be applied is always a tradeoff between computational complexity and fast convergence. [1]

In the present work we propose a new adaptive algorithm with averaging applied for noise cancellation. The conducted extensive experiments with different types of noise reveal its robustness maintaining fast convergence and at the same time keeping the computational complexity at a low level.

II. ADAPTIVE FILTERING

A. Adaptive system?

A system is said to be adaptive when it tries to adjust its parameters with the help of meeting some well-defined goal or target that depends upon the state of the system and its surroundings. So the system adjusts itself so as to respond to some phenomenon that is taking place in its surroundings. [8]

B. Adaptive filtering

Discrete-time (or digital) filters are ubiquitous in todays signal processing applications. Filters are used to achieve desired spectral characteristics of a signal, to reject unwanted signals, like noise or interferers, to reduce the bit rate in signal transmission, etc. The notion of making filters adaptive, i.e., to alter parameters (coefficients) of a filter according to some algorithm, tackles the problems that we might not in advance know, e.g., the characteristics of the signal, or of the unwanted signal, or of a systems influence on the signal that we like to compensate. Adaptive filters can adjust to unknown environment, and even track signal or system characteristics varying over time. [7]

C. Adaptive noise cancellation

Fig. 2.1 shows the classical scheme for adaptive noise cancellation using digital filter with finite impulse response (FIR). The primary input consists of speech s(n) and noise $n_2(n)$ while the reference input consists of noise $n_1(n)$ alone. The two noises $n_1(n)$ and $n_2(n)$ are correlated and hi(n) is the impulse response of the noise path. The system tries to reduce the impact of the noise in the primary input exploring the correlation between the two noise signals. This is equivalent to the minimization of the mean-square error $E\,[e^2\,(n)]$ where

$e\,(n) = s\,(n) + n_2\,(n) - n_3\,(n)$

Having in mind that by assumption, s(n) is correlated neither with $n_2(n)$ nor with $n_1(n)$ we have

$E\,[e^2\,(n)] = E[s^2(n)] + E[n_2(n) - n_3(n)]^2$

In other words the minimization of $E\ [e^2(n)]$ is equivalent to the minimization of the difference between n_2 (n) and n_3 (n). Obviously $E\ [e^2\ (n)]$ will be minimal when $n_3\ (n) \approx n_2\ (n)$ i.e. when the impulse response of the adaptive filters closely mimics the impulse response of the noise path. The minimization of $e[e^2(n)]$ can be achieved by updating the filter taps $w_i(n)$. [1]

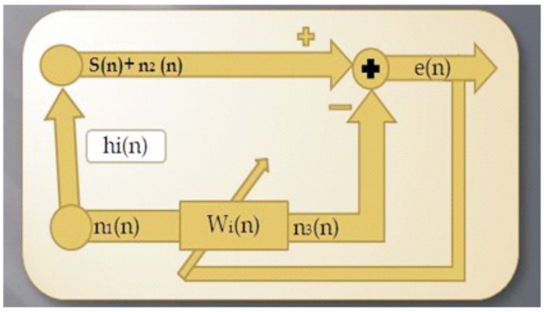


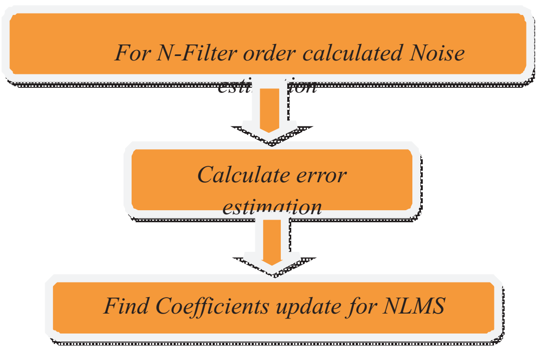
Figure-2.1: Adaptive noise cancellation scheme

III. NOISE CANCELATION

A. Noise Cancelation using NLMS algorithm

One of the primary disadvantages of the LMS algorithm is having a fixed step size parameter for every iteration. This requires an understanding of the statistics of the input signal prior to commencing the adaptive filtering operation. In practice this is rarely achievable. Even if we assume the only signal to be input to the adaptive echo cancellation system is speech, there are still many factors such as signal input power and amplitude which will affect its performance[15].

The normalized least mean square algorithm (NLMS) is an extension of the LMS algorithm which bypasses this issue by calculating maximum step size value. Each iterations of the NLMS algorithm require these steps in the following order:



Mathematical solution for each step is given bellow:

1) Noise estimation is calculated for N-Filter order using:

$$n3= \sum_{i=0}^{N} Wi(n)n1(n-i)$$

2)Error estimation can be calculated using:

$$e(n)=s(n)+n_2(n)- n_3 (n)$$

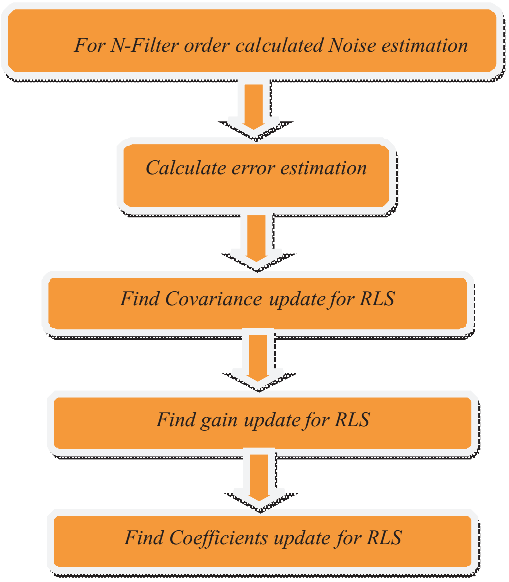
3) Coefficients update for NLMS is given by:

$$w_i(n+1) = w_i(n) + \mu \frac{e(n)\, n_1(n-i)}{\sum_{i=0}^{N} n_1^2(n-i)}$$

For $0 \leq i \leq N$

B. Noise Cancelation using RLS algorithm:

The Recursive least squares (RLS) adaptive filter is an algorithm which recursively finds the filter coefficients that minimize a weighted linear least squares cost function relating to the input signals. This is in contrast to other algorithms such as the least mean squares (LMS) that aim to reduce the mean square error. In the derivation of the RLS, the input signals are considered deterministic, while for the LMS and similar algorithm they are considered stochastic. Compared to most of its competitors, the RLS exhibits extremely fast convergence. However, this benefit comes at the cost of high computational complexity, also Compared to least mean squares (LMS) algorithms, recursive least squares (RLS) algorithms have a faster convergence speed and do not exhibit the eigen value spread problem. However, RLS algorithms involve more complicated mathematical operations and require more than LMS algorithms. Each iterations of the RLS algorithm require these steps in the following order:



Mathematical solution for each step is given bellow:

1) Noise estimation is calculated for N-Filter order using

$$n_3= \sum_{i=0}^{N} Wi(n)n1(n-i)$$

2) Error estimation can be calculated using:

$$e(n)=s(n)+n2(n-d)- n3 (n)$$

3) Covariance update for RLS is given by:

$$P(n) = P(n-1) - \frac{P(n-1)N_1(n)N_1^T(n)P(n-1)}{1/\delta + N_1^T(n)P(n-1)N_1(n)}$$

4) Gain update for RLS is given by:

$$K(n) - \frac{P(n-1)N_1(n)}{1/\delta + N_1^T(n)P(n-1)N_1(n)}$$

5) Coefficient update for RLS is given by:

$$w_i = w(n-1) + k_i(n)e(n)$$

For $0 \leq i \leq N$

$0<\delta<1$ P (0) = αI α large

C. Noise Cancelation using averaging algorithm

As mentioned in the introduction, for application where the fast convergence rate is vital, NLMS algorithm is not applicable. The more complex RLS algorithm maintains a good rate of adaptation but the prize to be paid is an order-of-magnitude increase in complexity. Moreover RLS algorithm is known to have stability issues due to the recursive covariance update formula. In this section we introduce a new adaptive algorithm applied for noise cancellation based on adaptive filtering with averaging. We start with defining the problem in the following manner. To recursively adjust the filter coefficients, so that the mean-square error is minimized, a standard algorithm for approximating the vector of filter coefficients can be written as

W (n+1) = W (n) – a (n) N1(n)e(n) (1)
Where

W (n) = [w0 (n), w1 (n)… wN (n)]
T is the coefficients vector,
$N_1(n)$ = [$n_1(n)$, n1(n-1),…, n1(n-N)]T is the input vector and a(n) is a sequence of positive scalars as a(n)→0 for n→∞
In (1) the estimation error can be given by
e(n) = s(n) + n2(n) - N1
T (n) w(n) (2)

The equation (1) could be transformed through taking the averages of W:

$$W(n+1)=W(n)+\frac{1}{n^{\gamma}}N1(n)e(n)$$

$$\overline{w}(n)=\frac{1}{n}\sum_{k=1}^{n}W(k)$$

$$\frac{1}{2}<\gamma<1$$

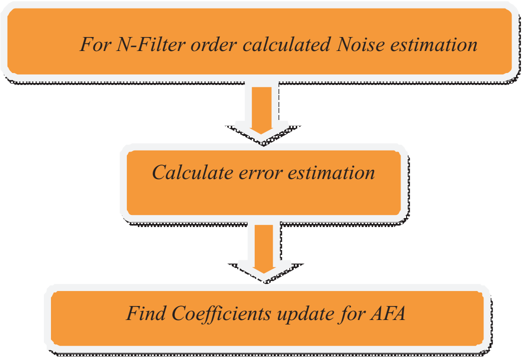
In order to improve the stability we undergo the second step, namely to average not only trough the approximation sequence but also through the observed signals N1 and e.[1] This leads us to an adaptive algorithm with averaging (AFA):

$$\overline{w}(n)=\frac{1}{n}\sum_{k=1}^{n}W(k)$$

$$W(n-1)=\overline{w}(n)+\frac{1}{n\gamma}\sum_{k=1}^{n}N1(k)e(k) \quad (4)$$

$$\frac{1}{2}<\gamma<1$$

Each iterations of the AFA algorithm requires following steps in order given bellow:



Mathematical solution for each step is given bellow:
1) Noise estimation is calculated for N-Filter order using:

$$n_3=\sum_{i=0}^{N}Wi(n)n1(n-i)$$

2) Error estimation can be calculated using:

e (n)=s(n)+n2(n)- n3 (n)

5) Coefficient update for AFA is given by

$$\overline{w}(n)=\frac{1}{n}\sum_{k=1}^{n}Wi(k)$$

$$\overline{n1ei}(n)=\sum_{k=1}^{n}n1(k-i)ei(n)$$

$$w_i(n+1)=w_i(n)+\frac{1}{n^{\gamma}}n1(n)ei(n)$$

For 0 = i = N and 1/2<γ<1

IV. DISCUSSION REGARDING PROPOSED WORK

In many applications of noise cancellation the changes in signal characteristics could be quite fast. This requires the utilization of adaptive algorithms, which converge rapidly. From this point of view here we try to work on AFA algorithm for the expected performance of averaging in noise cancellation problem in the scenario where to achieve a high convergence rate in order to meet the requirements imposed by applications where the changes in signal characteristics could be quite rapidly. According to this aspect AFA algorithm is very promising & significantly useful because of their high adaptation rate, comparable to that of the RLS algorithm & low computational complexity and possible robustness in fixed-point implementations. [1]

In experimental analysis we take into account a speech signal (world "ssgb") corrupted with noise of some db SNR using RLS, NLMS & AFA (proposed) algorithm.

According to [1] RLS and AFA have

good parameters than NLMS. But for the computational complexity and stability point of view it is expected that AFA (proposed) algorithm gives better results/performance than RLS algorithm by our upcoming experimental work.

V. CONCLUSION

Adaptive filtering for noise cancellation is an extensive research work hence adaptive filtering is a wide area of researcher in present in the field of communication because in many applications of noise cancellation the changes in signal characteristics could be quite fast. This requires the utilization of adaptive algorithms, which converge rapidly.

The main goal of this revive paper is to investigate the application of an algorithm based on adaptive filtering with averaging in noise cancellation problem in todays many application regarding the computational complexity and stability point of view. Here the main concern is to achieve a high convergence rate in order to meet the requirements imposed by applications where the changes in signal characteristics could be quite rapid. In this aspect the expected results show the AFA algorithm is very promising & efficient than others existing. Its main advantages have high adaptation rate, low computational complexity and possible robustness in fixed-point implementations. The method is applicable for producing good experimental results in noise cancellation using the concept of adaptive filtering.

REFERENCES

[1] Georgi Iliev and Nikola Kasabov, “Adaptive Filtering with Averaging in Noise Cancellation for Voice and Speech Recognition”.

[2] Shigeji Ikeda and Akihiko Sugiyama, “An Adaptive Noise Canceller with Low Signal Distortion for Speech Codecs”, IEEE TRANSACTIONS ON SIGNAL PROCESSING, VOL. 47, NO. 3, MARCH 1999.

[3] Sayed. A. Hadei” A Family of Adaptive Filter Algorithms in Noise Cancellation for Speech Enhancement” International Journal of Computer and Electrical Engineering, Vol. 2, No. 2, April 2010. 1793-8163

[4] Simon Haykin ,McMaster University,Hamilton, Ontario, Canada L8S 4K1” Adaptive Filters”

[5] BooGyeon Park and Thomas Kailath” New Square-Root Smoothing Algorithms” IEEE TRANSACTIONS ON AUTOMATIC CONTROL, VOL. 41, NO. 5, MAY 1996.

[6] CHAI SZE CHIANG “Adaptive noise cancellation using hybrid algorithum”may2003.

[7] Christian Feldbauer, Franz Pernkopf, and Erhard Rank Signal Processing and Speech Communication Laboratory Inffeldgasse 16c” Adaptive Filters”

[8] A.Bhavani Sankar, D.Kumar & K.Seethalakshmi” Performance Study of Various Adaptive Filter Algorithms for Noise Cancellation in Respiratory Signals”

[9] Lilatul Ferdouse,Nasrin Akhter, Tamanna Haque Nipa and Fariha Tasmin Jaigirdar” Simulation and Performance Analysis of Adaptive Filtering Algorithms in Noise Cancellation” IJCSI International Journal of Computer Science Issues, Vol. 8, Issue 1, January 2011 185 ISSN (Online): 1694-0814.

[10] BERNARD WIDROW, JOHN R. GLOVER, JR, JOHN M. MCCOOL, JOHN KAUNITZ, CHARLES s. WILLIAMS, ROBERT H. HEAN, JAMES R. ZEIDLER, EUGENE DONG, JR., AND ROBERT C. GOODLIN” Adaptive Noise Cancelling: Principles and Applications” PROCEEDINGS OF THE IEEE, VOL. 63, NO. 12, DECEMBER 1975.

[11] Beanard Widrow,Michael Lehr,Francoise Beaufays,Eric Wan,Michel Bilello”Adaptive Signal Processing”

[12] Jashvir Chhikara, Jagbir Singh” Noise cancellation using adaptive algorithms” International Journal of Modern Engineering Research (IJMER) Vol.2, Issue.3, May-June 2012 pp-792-795 ISSN: 2249-6645.

[13] J. M. Górriz, Javier Ramírez, S. Cruces-Alvarez, Carlos G. Puntonet, Elmar W. Lang, and Deniz Erdogmus" A Novel LMS Algorithm Applied to Adaptive Noise Cancellation" IEEE SIGNAL PROCESSING LETTERS, VOL. 16, NO. 1, JANUARY 2009.

[14] Alok Pandey, L.D. Malviya,Vineet Sharma" Comparative Study of LMS and NLMS Algorithms in Adaptive Equalizer" International Journal of Engineering Research and Applications (IJERA) ISSN: 2248-9622, Vol. 2, Issue 3, May-Jun 2012, pp.1584-1587.

[15] Radhika Chinaboina, D.S. Ramkiran,Habibulla Khan, M.Usha,B.T.P.Madhav,K.Phani Srinivas & G.V.Ganesh"ADAPTIVE ALGORITHMS FOR ACOUSTIC ECHO CANCELLATION IN SPEECH PROCESSING" Volumes/Vol7Issue1/IJRRAS_7_1_05, April 2011.