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ANKUR KUMAR
Electrical Engg., (N.I.T Patna) India, ankur.bit.2k5@gmail.com

GK CHOUDHARY
Electrical Engg., (N.I.T Patna) India, girishkrchoudhary@yahoo.co.in

amrita sinha
Institute of Technology, Banaras Hindu University, amrita_bhu@yahoo.co.in

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[1]ANKUR KUMAR, [2]GK CHOUDHARY & [3]AMRITA SINHA

[1](M.Tech) Electrical Engg., (N.I.T Patna) India;
[2]Head of Department, Electrical Engg.(N.I.T Patna);
[3]Electrical Engg., (N.I.T Patna) India
Email:ankur.bit.2k5@gmail.com; girishkrchoudhary@yahoo.co.in; amrita_bhu@yahoo.co.in

Abstract*:* The paper proposed the adaptive noise suppression technique for suppression of noise in voice communication. There are different techniques earlier used for adaptive filteration like least mean square, kalman's filter etc.In the paper we used "fuzzy logic" technique for adaptive filteration. We know about the theory of adptive filteration of noise and application of fuzzy logic. We are using the fuzzy logic functions anfis and genfis1 by matlab for simulation. Anfis is the adaptive neuro-fuzzy training of sugeno-type fuzzy inference systems. In this paper we uses anfis system to suppress different types of noise from voice signal.

Keywords*: lms filter, voice communication, k- -alman's filter, fuzzy logic, anfis etc.*

1. INTRODUCTION

Voice communication is very important factor in modern world. Human's life is widely dependent on voice or speech communication like mobile communication or voice synthesis has very important impact on the human life. But background noise like bus noise, car horn noise or truck horn noise disturb the massage signal. Thus background noise results in massage degradation. The use of adaptive filter removes the background noise working on Fuzzy system. Anfis system of Fuzzy works as adaptive network.

Norbert Wiener proposed the filter to reduce the noise in present in a signal by comparison with an estimation of the desired noiseless signal [1] with this the estimation theory implied in noise cancellation. Plackett developed some theorems known as recursive least square family of adaptive filtering algorithms [2] .The LMS algorithm is closely related to the concept of stochastic approximation developed by Robbins & Monro [3] in statics for solving certain sequential parameters estimation. The LMS algorithm was devised by Windrow and Hoff [4] in their study of a pattern-recognition scheme known as the adaptive linear element. Nearly at the same time Zadeh introduced the fuzzy set theory for information control [5]. Another stochastic gradient algorithm, closely related to the LMS algorithm, is the gradient adaptive lattice (GAL) algorithm. Sorenson, H. W.[6] discussed about the Least-Squares estimation in which he summarized the whole development in the area of estimation theory.

Godrad [7] used Kalman filter theory to derive one variant of the algorithm that is sometime reffered to in the literature of Godard algorithm.

In voice communication S.F. Boll [8] work on Suppression of acoustic noise in speech using spectral subtraction. Zames [9] introduced the H. norm (or minimax criterion) as a robust index of performance for solving problems in estimation control and with it the field of robust control took a new research direction. Sayed.et.el.[10] published an expository paper in which the exact relationship between the RLS algorithm and kalman filter theory was delineated for the first time. Before this P. Lockwood et.el done Experiments with a nonlinear spectral subtractor (NSS), hidden markov models and the projection, for robust speech recognition in cars [11].

After some time Yoshinari et al. used clustering technique for construction of fuzzy models through clustering techniques[12]. After some time Hassibi ET. Al.[13] have shown that the LMS algorithm is indeed optimal under the H. norm.

K. Wu and P. Chen in [14] had used the spectral su -btraction method for efficient speech enhance - ment in car hands-free application. Most recently Jan Vanuš used the lms algorithm for noise removal in voice communication[15].

We are using the anfis system for noise suppression in voice signal. We are also comparing the technique with the previously used techniques least mean square, normalised-lms and weiner filter for adaptive noise suppression. We are taking three kind of noise here and suppress the noise from the signal.

2. LMS ALGORITHM

The Least Mean Square (LMS) algorithm, introduced by Widrow and Hoff in 1959 [3] is an adaptive algorithm, which uses a gradient-based method of steepest decent [2]. LMS algorithm uses the estimates of the gradient vector from the available data. LMS incorporates an iterative procedure that makes successive corrections to the weight vector in the direction of the negative of the gradient vector which eventually leads to the minimum mean square error. Compared to other algorithms LMS algorithm is relatively simple; it does not require correlation function calculation nor does it require matrix inversions.

With each iteration of the LMS algorithm, the filter tap weights of the adaptive filter are updated according to the following formula. .

$$W(n+1)=w(n)+2\mu e(n)x(n) \ldots\ldots (1)$$

Here x(n) is the input vector of time delayed input values, x(n) = [x(n) x(n-1) x(n-2) .. x(n-N+1)]T. The vector w(n) = [w0(n) w1(n) w2(n) .. wN-1(n)]T represents the coefficients of the adaptive FIR filter tap weight vector at time.

3. 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 noise cancellation system is speech, there are still many factors such as signal input power and amplitude which will affect its performance.

The normalised least mean square algorithm (NLMS) is an extension of the LMS algorithm which bypasses this issue by calculating maximum step size value. Step size value is calculated by using the following formula.

Step size=1/dot product
(input vector, input vector) (2)

This step size is proportional to the inverse of the total expected energy of the instantaneous values of the coefficients of the input vector x(n). This sum of the expected energies of the input samples is also equivalent to the dot product of the input vector with itself, and the trace of input vectors auto-correlation matrix.

$$tr(R)=\sum_{i=0}^{N-1} E[x^2(n-i)]=E\sum_{i=0}^{N-1}[x^2(n-i)] \ldots (3$$

The recursion formula for the NLMS algorithm is stated in equation .

$$W(n+1)=w(n)+\frac{1}{x^T(n)x(n)}e(n)x(n) \ldots\ldots\ldots\ldots (4)$$

4. WIENER'S FILTER

In signal processing, the Wiener filter is a filter proposed by Norbert Wiener during the 1940s and published in 1949[1]. Its purpose is to reduce the amount of noise present in a signal by comparison with an estimation of the desired noiseless signal... A Wiener filter is not an adaptive filter because the theory behind this filter assumes that the inputs are stationary. The input to the Wiener filter is assumed to be a signal,s(t), corrupted by additive noise, n(t). The output, ^ , is calculated by means of a filter, g(t) using the following convolution:

$$\hat{s}(t)= g(t)*[s(t)+n(t)] \ldots\ldots\ldots\ldots (5)$$

Where

s(t)is the original signal (not exactly known; to be estimated) n(t)is the noise ^ , is the estimated signal (the intention is to equal s(t+a)) g(t) is the Wiener filter's impulse response.The error is defined as

$$e(t)=s(t+\alpha)- \hat{s}(t) \ldots\ldots\ldots\ldots (6)$$

where a is the delay of the Wiener filter (since it is causal) In other words, the error is the difference between the estimated signal and the true signal shifted by a.The squared error is

$$\epsilon^2(t) = s^2(t+\alpha) - 2s(t+\alpha)\,\hat{s}(t)+\hat{s}(t)^2 \ldots (7)$$

Where s(t+a) is the desired output of the filter e(t) is the error.

5. ANFIS

The proposed method in this paper is Anfis system. It is adaptive system using Fuzzy rules membership functions for prediction.

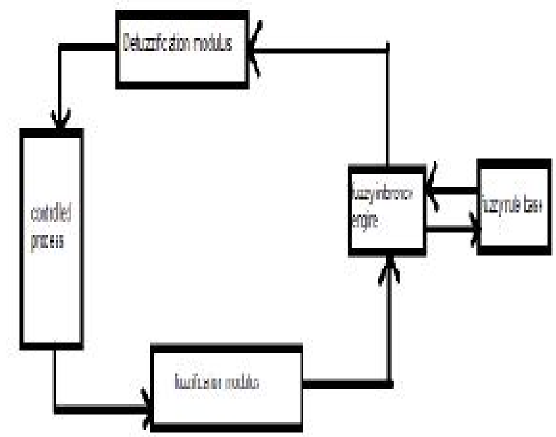

Figure 1 fuzzy system

The fuzzy system contains three steps:

1. Fuzzification of crisp values:- we fuzzifying the crisp value we have using membership functions and applied it to fuzzy inference engine
2. Fuzzy inference system:- Here we generate the fuzzy rule suaitable for the system.
3. Defuzzification of output :- here we defuzzified the output using any of the various methods.
   In Anfis system the system train itself for given set of training datait itself define its rules. In the simulation, the anfis architecture is employed to model nonlinear function.

Using a given input/output data set, the toolbox function ANFIS constructs a fuzzy inference system (FIS) whose membership function parameters are tuned (adjusted) using either a back propagation algorithm alone, or in combination with a least squares type of method. This allows our fuzzy systems to learn from the data they are modeling.

6. EXPERIMENTAL WORK

We generate noise signals due to bubble noise for 5 second and 1000 samples of data has been taken the signal having strength 17.94 db.

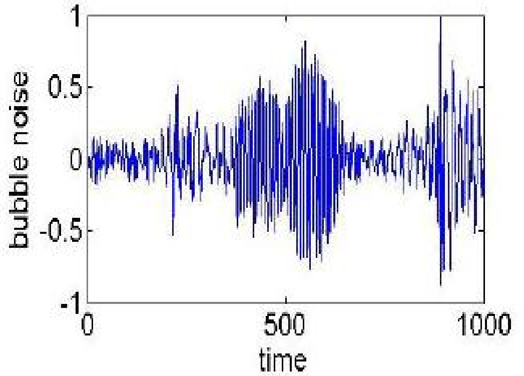

Figure 2: bubble noise

The car horn noise recorded for 5 seconds consisting 1000 samples having magnitude 26.74db is shown in figure3.

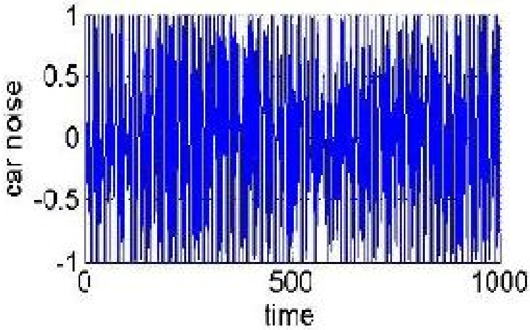

Figure 3: car horn noise

The truck horn noise generated for 5 seconds consisting 1000 samples havine magnitude 24.94 db is shown in figure4.

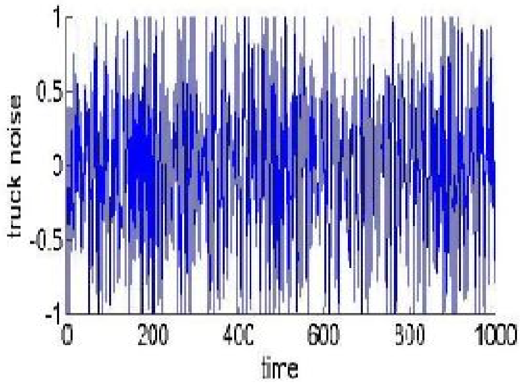

Figure:4 truck horn noise

The original signal generated for 5 seconds as "hello hello hello hello…………" 1000 samples taken with strength of signal is 12.88 db shown in figure5.

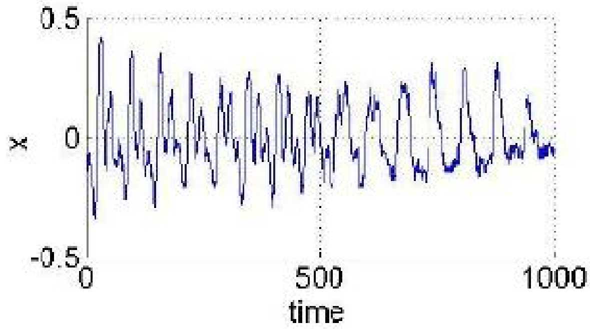

Figure 5: signal

The interference signal which is generated with the non linear characteristic of noise is shown in figure6. This signal is used as second input with m signal contains signal and noise as first input for the system.

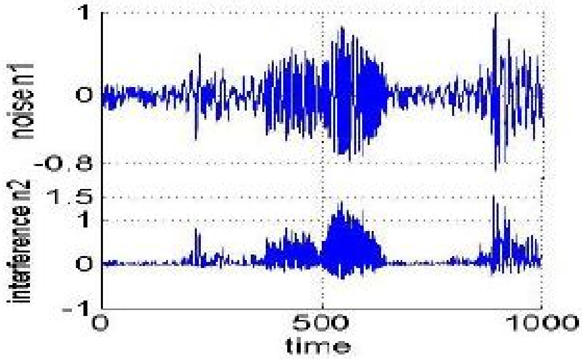

Figure 6: interference for bubble noise

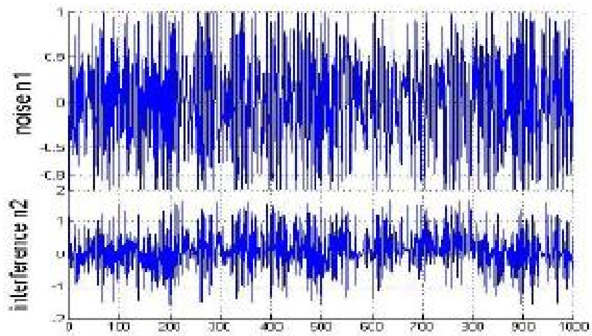

Figure7: interference signal for truck horn noise

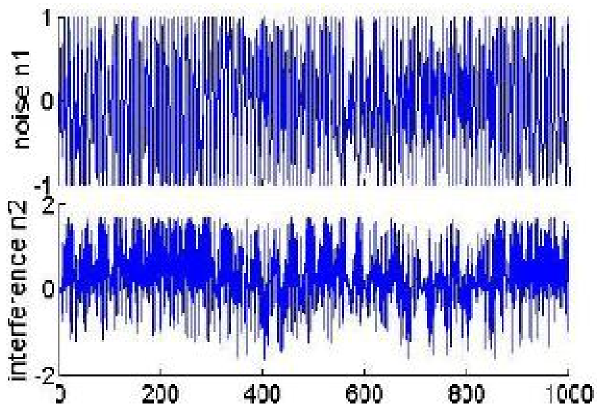

Figure8: interference signal for car horn

Now we take 1000 samples of data of the m signal and the interference noise and of the desired output

.With these data we make a training data set having 1000 sample now we generate adaptive fuzzy system. The message signal mixed with truck horn noise is shown in figure9.

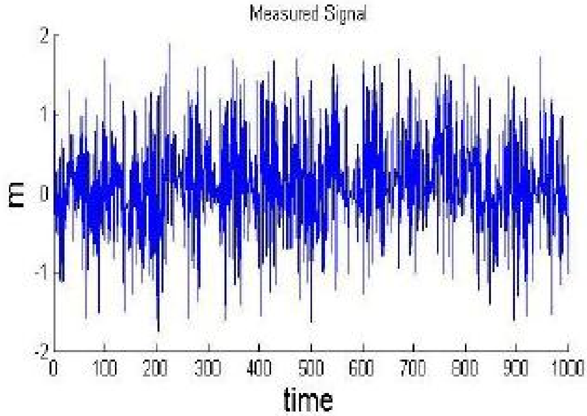

Figure9: message signal with truck horn

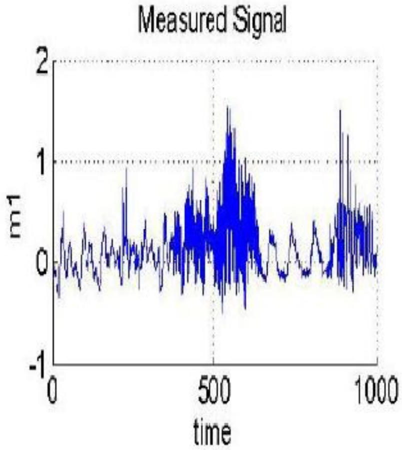

Figure10: message signal with bubble noise

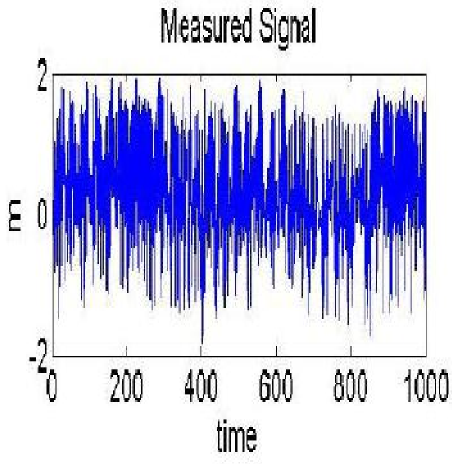

Figure 11: message signal with car horn

The output is shown in figure8. We have output estimated as the signal. We have a training set of data having two inputs and one output. Applied to an Anfis system for two membership function and step size is 0.2 the no of epochs are 10.

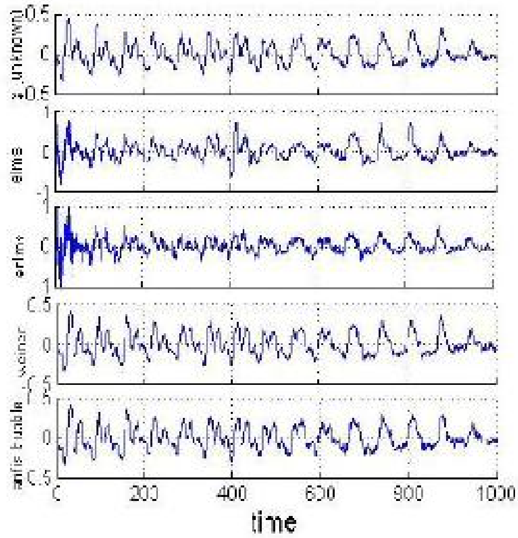

Figure12: output for car horn noise

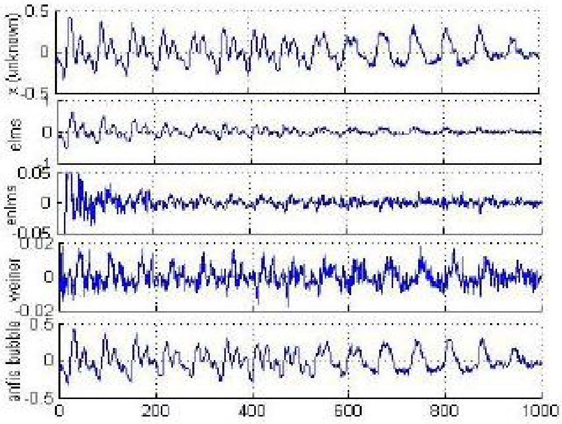

Figure13:output for truck horn noise

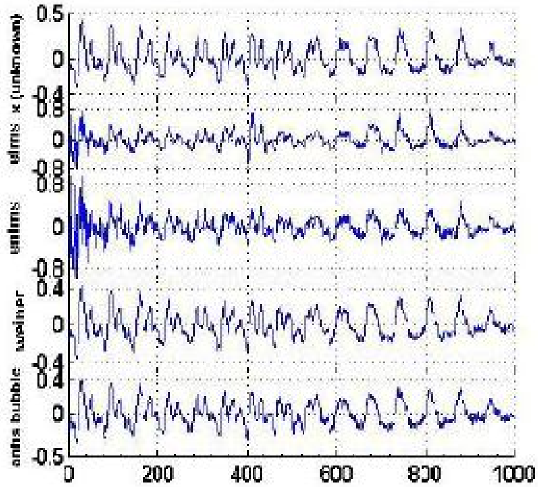

Figure14: for bubble noise

For different types of error the minimum square error for lms and anfis system. The mean square error for the two processes.

Added noise with signal	LMS (db) Rms error	NLMS(db) Rms error	Wiener's filter(db) Rms error	Antis system(db) Rms error
Car horn	4.5618	4.5225	4.5184	0.3243
Truck horn	2.8009	2.7936	2.6842	0.1743
Bubble noise	6.6269	4.7130	4.0818	0.4756

Table 1

By using the formula for signal to noise ratio we got set of result for the different types of noises

$$\mathrm{SNR}=\mathbf{10log}\frac{d^2(k)}{n^2(K)}$$ the improvement in SNR.

Added noise with signal	LMS (db)	NLMS(db)	Wiener's filter(db)	Anfis system(db)
bubble horn	3.6696	4.6723	7.3943	7.3993
Truck horn	10.5594	12.5437	14.0025	14.00578
Car noise	13.093	14.0120	16.7340	16.7395

Table 2

7. CONCLUSION AND FUTURE WORK

The proposed system provides an incorporation of ttechnique and it provides a complete solution for the background noise cancellation system in voice communication and in hearing aids. Fuzzy based noise cancellation system results better performance than the classical algorithm. The result shows that the SNR improvement with .fuzzy anfis algorithm shows about 4 dB improved performance than LMS algorithm near about 3dB from NLMS Wiener filter shows nearly similar result but wiener filter is not adaptive algorithm and the step size is fixed. ANFIS system better improvement in reducing the mean square error about 7% better result shows by this system .The entire system is tested only for three different noise signals, but still for mobile environment, various environmental noises are to be considered. In real time, many different noise signals may be combined with the original information. To get better result two or more stages of this noise cancellation system may be used.

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