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ADAPTIVE NOISE SUPPRESSION IN VOICE COMMUNICATION USING ANFIS SYSTEM

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Abstract **—This paper will propose the adaptive noise suppression technique for suppression of noise in voice communication. There are different techniques earlier used for adaptive noise suppression like least mean square, kalman's filter etc. In the paper we will use "fuzzy logic" technique for adaptive filtration. 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 use Anfis system to suppress different types of noise from voice signal.**

Index Terms **— lms filter, voice communication, kalman's filter, fuzzy logic, Anfis etc.**

I. INTRODUCTION

Voice communication is integral part of Human's life. Life is widely dependent on voice or speech communication like mobile communication. 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 removal of the noise from signal is an underlying problem in several areas of research in communication. The presence of noise between transmitter and receiver degrade and corrupt the input signal.

Norbert Wiener proposed the filter to reducing noise present in a signal by comparison with the desired noiseless signal [1]. With this the estimation theory derived in noise reduction technique. After that Placket 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 concept developed by Robbins & Monro [3] in statics for solving certain sequential parameters estimation. The LMS algorithm was derived by Windrow and Hoff [4] in the 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 referred to in the literature of Godard algorithm. In voice communication S.F. Boll [8] works on Suppression of acoustic noise in speech using spectral subtraction. Zames [9] introduced the H. norm (or minimal 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 sub tractor (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] has shown that the LMS algorithm is indeed optimal under the H. norm. K. Wu and P. Chen in [14] had used the spectral subtraction method for efficient speech enhancement in car hands-free application. Most recently Jan Vanuš used the lms algorithm for noise removal in voice communication [15].

We will use the Anfis system for noise suppression in voice signal. We will also compare the technique with the previously used techniques least mean square, normalised-lms and Weiner filter for adaptive noise suppression. We will take three kind of noise here and suppress the noise from the signal.

II. ADAPTIVE NOISE CANCELLATION:

Adaptive noise cancellation is used to remove background noise signals. This is an extremely useful technique where a signal is submerged in a very noisy environment. A typical example is inside a jet aircraft. The engine of jet aircraft can produced noise at a level over 140decibles with normal human speech is at a level of 30-40decibles, the pilot's communication is impossible in such a environment if there no noise cancellation equipments inside the cockpit. Usually the background noise does not keep steady and it will change from time to time. For example, the noise from the jet engine will be different at various flight states. So the noise cancellation must be an adaptive process: it should be able to work under changing conditions, and be able to adjust itself according to the changing environment.

Adaptive filters are digital filters with an impulse response or transfer function that can be adjusted or changed over time to match desired system characteristics. It adapts, automatically, to changes in its input signals.

An Adaptive filter consists of two parts: one is a digital filter with adjustable coefficients and another is an adaptive algorithm which is used to adjust or modify the coefficients of the filter.

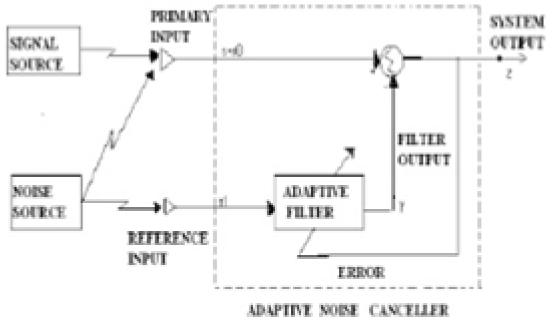


Figure 1: Adaptive Noise Cancellation

In the system shown in the Figure 1 the reference input is processed by an adaptive filtering adaptive filter differs from a fixed filter in that it automatically adjusts its own impulse response. These adjustments are accomplished through an algorithm that responds to an error single dependent, among other things, on the filter's output. Thus with the proper algorithm, the filter can operate under changing conditions and can readjust itself continuously to minimize the error signal.

III. 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,

$$X(n) = [x(n)\ x(n-1)\ x(n-2) \ldots x(n-N+1)]T.$$

The vector

$$W(n) = [w(n)\ w1(n)\ w2(n) \ldots wN\text{-}1(n)]T$$

represents the coefficients of the adaptive FIR filter tap weight vector at time.

IV. 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 normalized 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. The recursion formula for the NLMS algorithm is

Stated in equation.

V. ANFIS

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

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 suitable 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 data itself define its rules. In the simulation, the anfis architecture is employed to model nonlinear function.

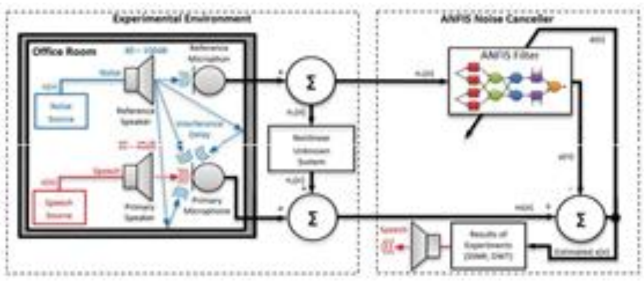
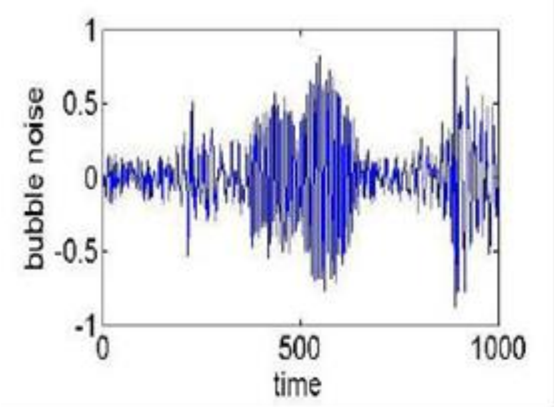


Figure 2: Simplified Principal Scheme of Adaptive Noise Suppression in Voice Communication Using the Neuro- Fuzzy Inference System.

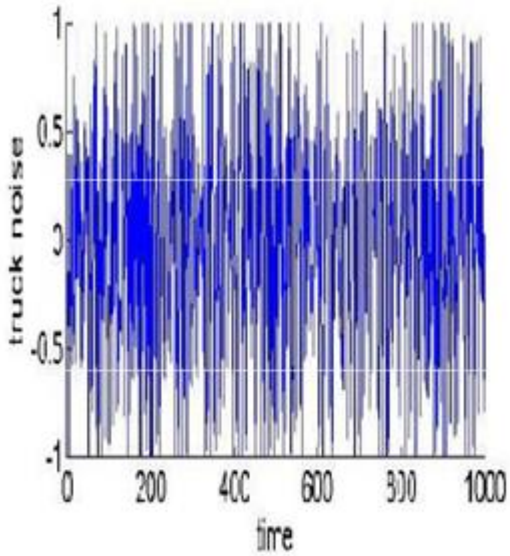
VI. EXPERIMENTAL WORK

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

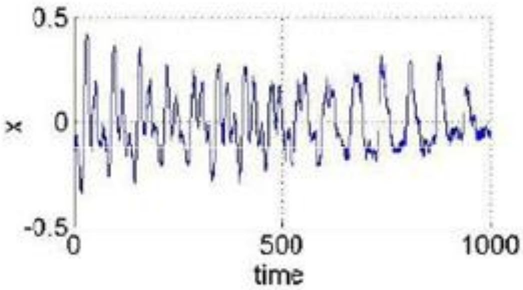


The car horn noise recorded for 5 seconds consisting 1000 samples having magnitude 26.74db is shown in figure3

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



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



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.

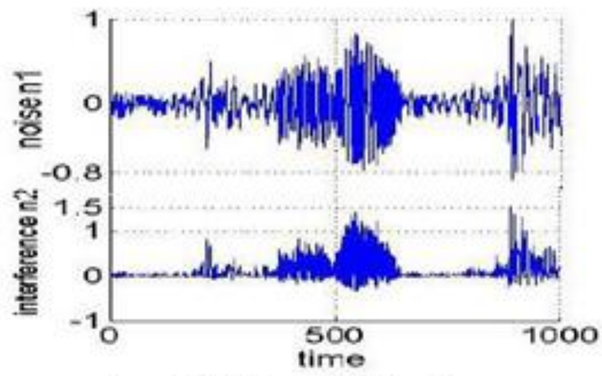


Figure 6: interference bubble noise

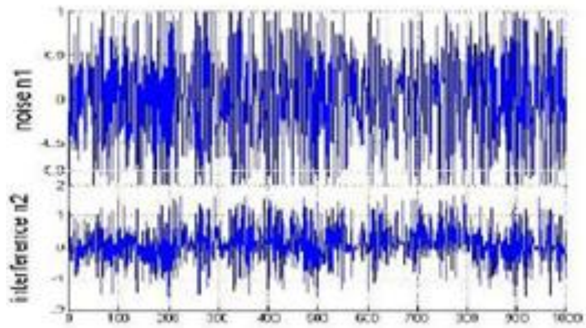


Figure7: interference signal for truck horn noise

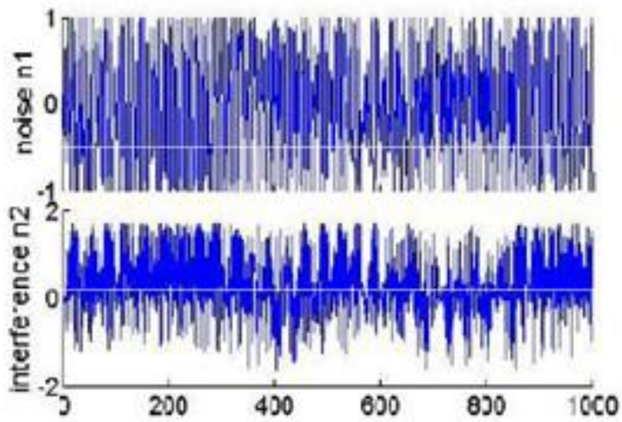


Figure8: interference

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.

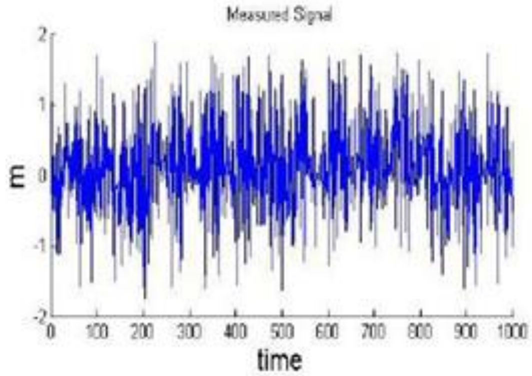


Figure9: message signal with truck horn

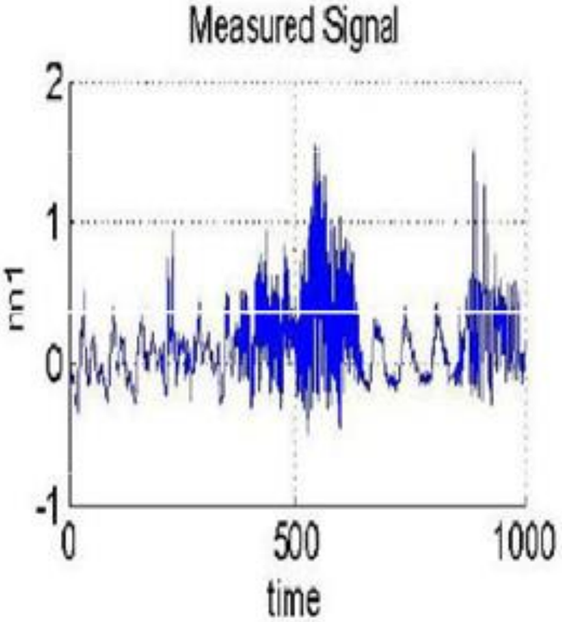


Figure10: message signal with bubble noise

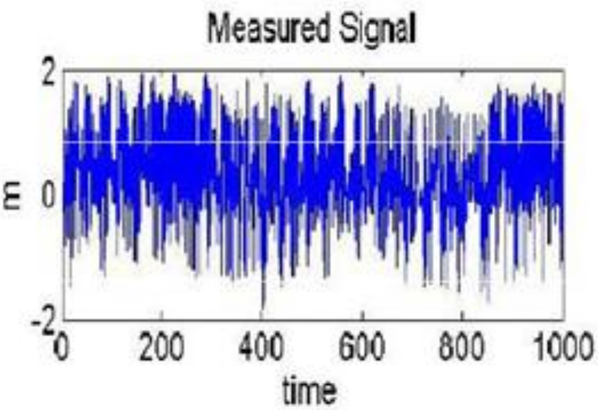


Figure 11: message signal with car horn

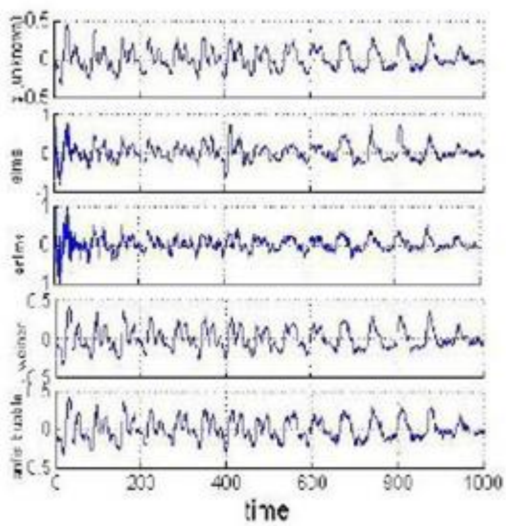


Figure12: output for car horn noise

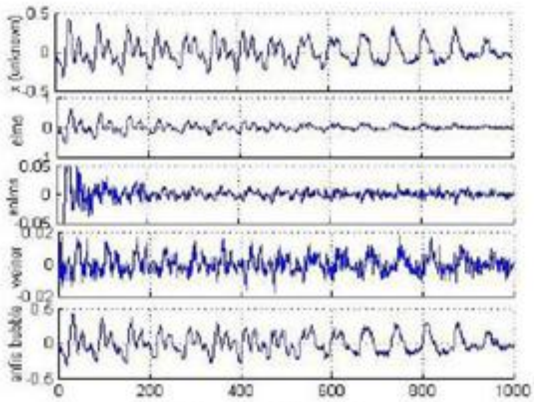


Figure13: output for truck horn noise

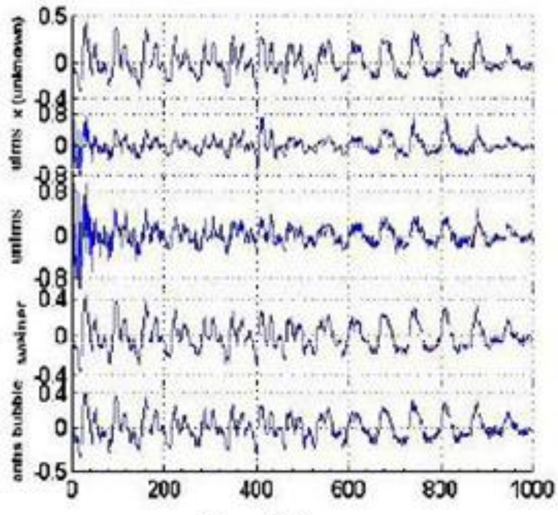


Figure14: for

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	Anfis 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

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

$\mathbf{SNR}=\mathbf{10log}\frac{d^2(k)}{n^2(K)}$ 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

CONCLUSION AND FUTURE WORK

The proposed system provides an incorporation of technique 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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