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"ATTENUATION OF ENVIRONMENTAL NOISE USING DIGITAL FILTERING"

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ABSTRACT

Environmental noise is a pervasive environmental pollution that leads to annoyance and can be perceived as stressful. There is some evidence that environmental noise is associated with higher rate of minor psychiatric disorder. The problem of controlling the noise has been a topic of research over the years. The target is to implement an environmental noise cancellation system using least mean square algorithm and normalized least mean square algorithm with noise amplitude modulation using adaptive filters. This deals with implementation and evaluation of performance of adaptive filtering algorithms for noise cancellation without reference signal. The evaluation of the algorithm is made using MATLAB programming. The implementation results show that the adaptive noise cancellation application benefits more from the use of the NLMS algorithm instead of the LMS algorithm.

1. INTRODUCTION

An adaptive filter is a device that is useful for processing an input signal by blocking and allowing some parts of it. It allows to guarantee that the human ear is not contaminated with the unwanted signals and only to rescue the information that needs to be received. Adaptive filters are having large range of applications such as noise cancellation, system identification, adaptive linear prediction, beamforming etc. Compared with other digital filters adaptive filter is the one whose characteristics can be adapted automatically to get the desired signal.

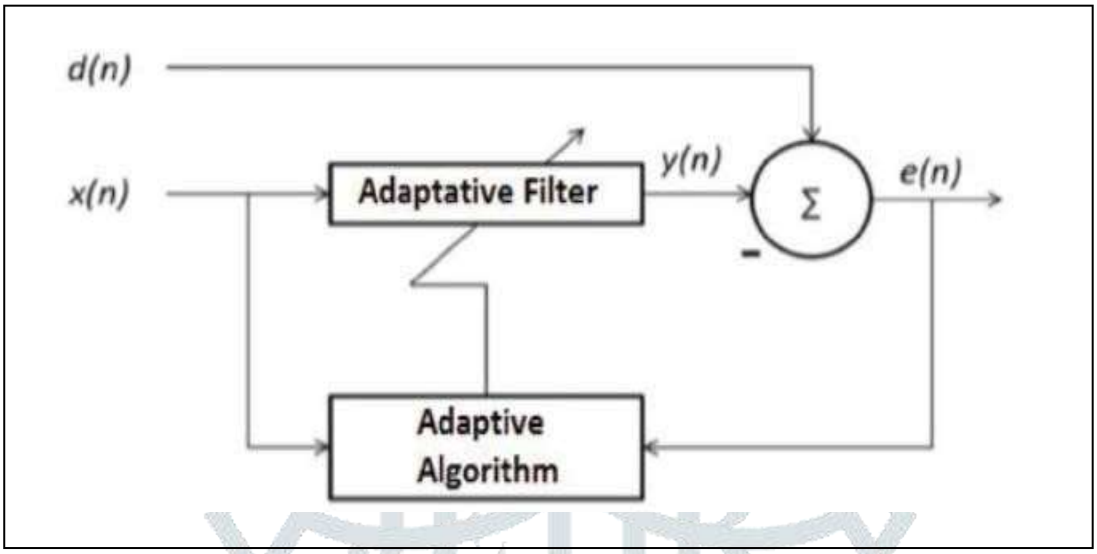

Fig 1: **Generic structure of an adaptive filter**

The filtering process involves the computation of the filter output y(n) in response to the filter input signal x(n). The filter output is compared with the desired response d(n), generating an error estimation e(n), as shown in Fig 1. The most important thing in the implementation of an adaptive filter is the requirement of the error signal e(n). Without the error signal, the filter cannot appropriate adapt to the environment, since the error sequence determines how the filter coefficients generally known as weights should be adjusted.

A common method to remove the noise from a corrupted signal is applying a linear filtering method. When the signal of interest and the noise are in separate frequency bands, a classical digital filter such as low-pass filter, high-pass filter or band-pass filter can be used to extract the desired signal. Another method is based on optimization theory and the approach is to minimize the mean-square error that is defined as the difference between the desired signal and the actual filter output. A major drawback of above two methods, is that the signal and noise characteristics must be known to determine the filter coefficients and hence there is a need of adaptive filter.

2. PROBLEM STATEMENT

In any environment there are audible signals that we need to perceive and other unwanted signals that cause the noise pollution. Noise can be generated by any source. Industrial machineries are composed of various noise sources such as rotors, electrical machines, IC engines etc. The effects of industrial noise pollution includes increase in blood pressure, headache, poor concentration, productivity losses in the workplace and sometimes it may also lead to serious issues. So a digital solution is proposed for the problem of industrial machine noise, in which desired and bearable output has to be estimated from machine noise.

3. DESIGN & METHODOLOGY

In many non-stationary applications, it is difficult to obtain the reference signal. For example, when a broadband signal is corrupted by periodic interference, no reference signal is available. In that case, it is possible to derive a reference signal by delaying the noisy process. The delayed signal x(n-n0) is used as the reference signal to eliminate the interference. If precise amount of delay n0 is chosen good noise reduction is achieved. Further the design of the proposal is partitioned in 4 stages.

3.1 STAGE A

It consists of the acquisition of the audible signal that is recovered from the industry. The proposal to be designed will minimize the sound intensity of these noises but not completely mitigate them. The proposal will capture all the random noises in the industry and process them.

3.2 STAGE B

The signal captured in stage A is in analog format since it was captured in natural form. The original signal has to be subjected to 2 basic operations Sampling and quantization.

3.2.1 SAMPLING

The first step to digitize the original signal consists of sampling operation. The sampling of the sound consists of taking small representative pieces of the signal so that they are encoded in binary digits to digitize them. Figure 5 shows the plot of sampled machine noise signal. The condition that the signal has to remain representative of the original must be considered. To cover the previous condition, the following equation known as Nyquist's theorem must be followed:

$$f_s >= 2f_o$$

3.2.2 QUANTIZATION

Quantization is the process of mapping continuous infinite values to a smaller set of discrete finite values. In the context of simulation and embedded computing, it is about approximating real-world values with a digital representation that introduces limits on the precision and range of a value. Quantization introduces various sources of error in your algorithm, such as rounding errors, underflow or overflow, computational noise, and limit cycles. Figure 6 represents quantized noise signal. Quantization is taken into account as its errors affect signal processing, wireless, control systems, FPGA, ASIC, deep learning, and other applications.

3.3 STAGE C

In this step the adaptive filter algorithm is to be implemented. Adaptive algorithms are used to adjust the coefficients of the digital filter, such that the error signal is minimized according to some criterion. Some of the most used algorithms are LMS and NLMS. These algorithms have been successfully implemented in different systems.

3.4 STAGE D

It consists of validating the filter response and also verifying that it inhibits unwanted frequencies. It is important to consider that the frequencies that will correspond to the desired signal must be the sounds that are desired and everything else must be blocked.

4. LMS ALGORITHM

The Least Mean Square (LMS) algorithm is one of the simplest and most widely used algorithms for adaptive filtering. It is based on the stochastic gradient descent method to find a coefficient vector which minimizes a cost function. The weight-vector update equation is given by

$$W+1 = W + e() * ()$$

where W is the estimate of the weight value vector at time n, x(n) is the input signal vector, e(n) is the filter error vector and μ is the step-size, which determines the filter convergence rate and overall behavior.

One of the difficulties in the design and implementation of the LMS adaptive filter is the selection of the step-size μ. This parameter must lie in a specific range, so that the LMS algorithm converges:

$$0 \quad < < 2 / \square$$

where □ is the largest eigenvalue of the autocorrelation matrix Rx.

5. NLMS ALGORITHM

Normalized Least Mean Square (NLMS) is actually derived from Least Mean Square (LMS) algorithm. The need to derive this NLMS algorithm is that the input signal power changes in time and due to this change the step-size between two adjacent coefficients of the filter will also change and also affect the convergence rate. Due to small signals this convergence rate will slow down and due to loud signals this convergence rate will increase and give an error. So, to overcome this problem, try to adjust the step-size parameter with respect to the input signal power. Therefore, the step-size parameter is said to be normalized.

The step size for computing the update weight vector is,

$$\mu(n) = \beta/c+|x(n)|^2$$

Where,

μ(n) is step-size parameter at sample n, β is normalized step-size (0 < β < 2), c is safety factor (small positive constant)

The weight vector update now is given by,

$$W(n+1) = W(n) + \mu(n)\ e(n)\ x^*(n)$$

or

$$W(n+1) = W(n) + [\ \beta/|x(n)|^2]\ e(n)\ x^*(n)$$

6. FLOWCHART

Below flowchart depicts the flow of implimentation of LMS algorithm.

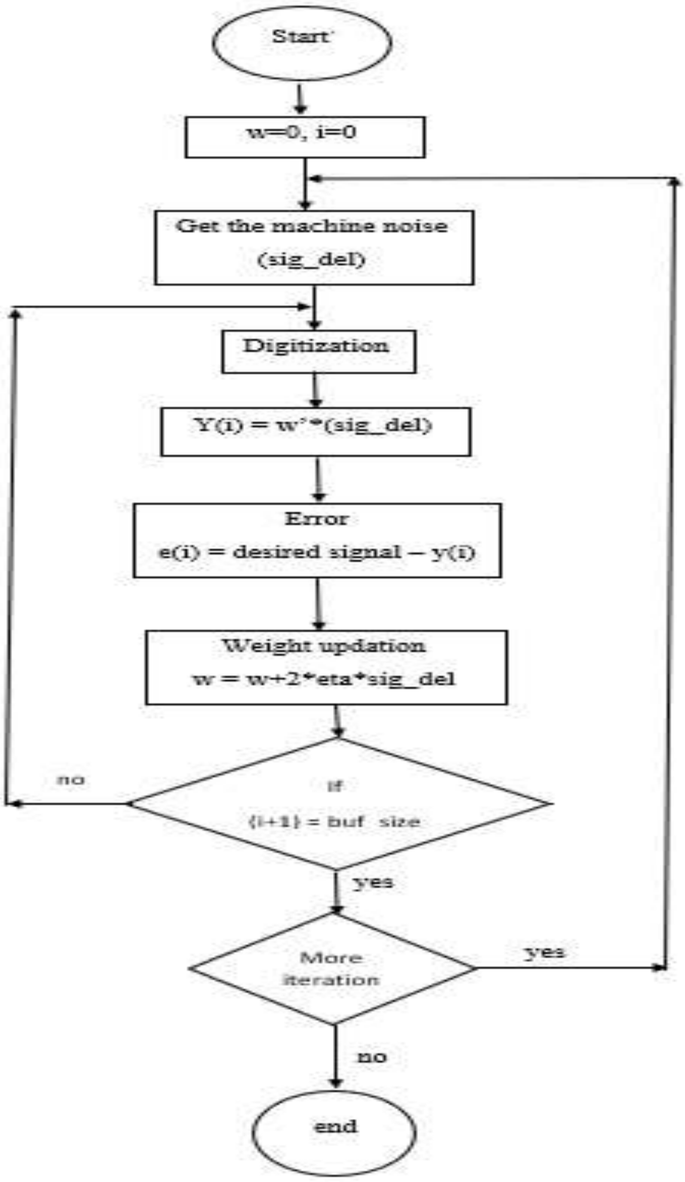

Fig 2 : flowchart of LMS

7. RESULTS

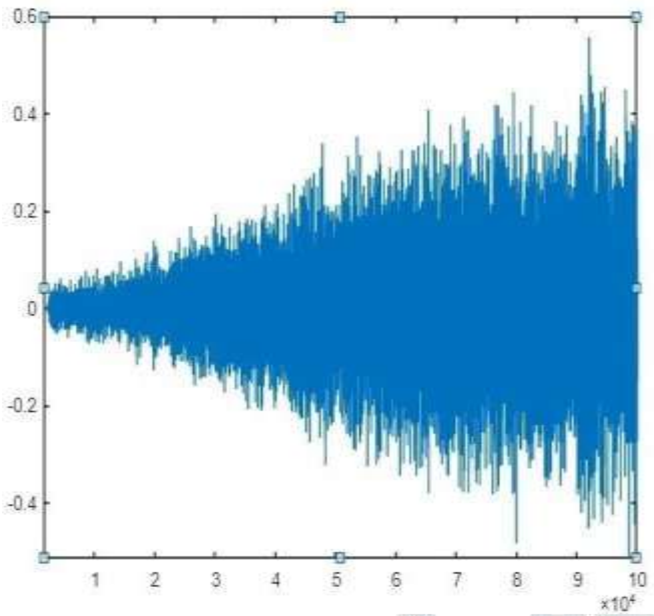

Fig 3: Machine Noise Observation

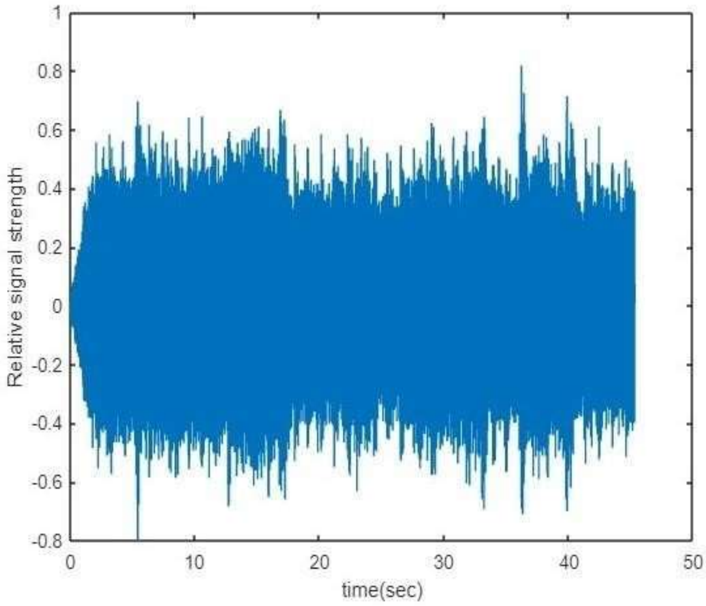

Fig 4: Relative Signal Strength Plot

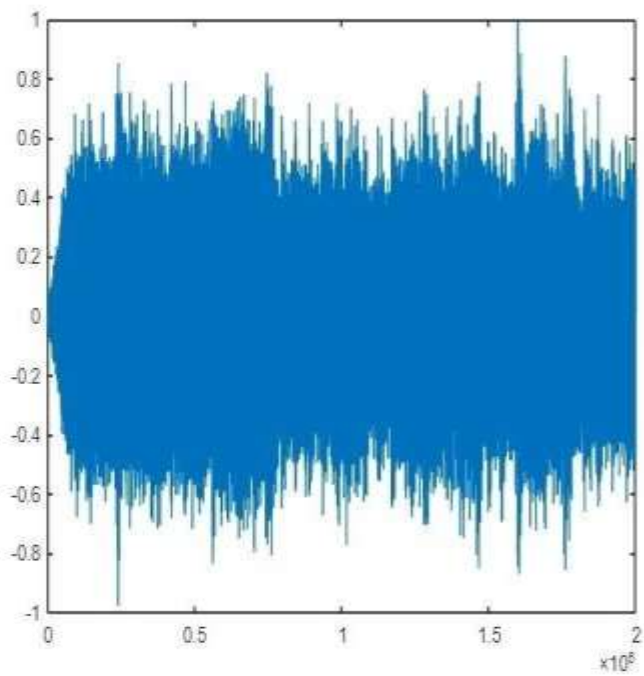

Fig 5: Sampled Noise Signal

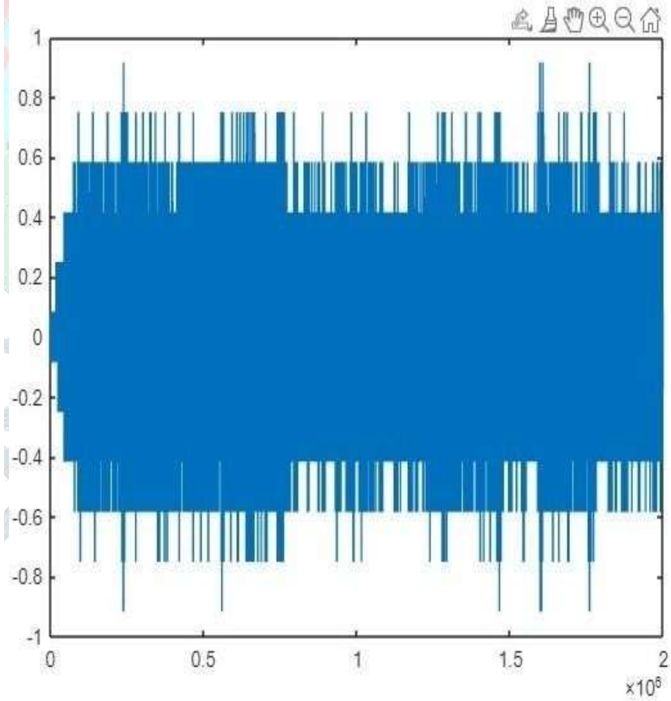

Fig 6: Quantized Noise Signal

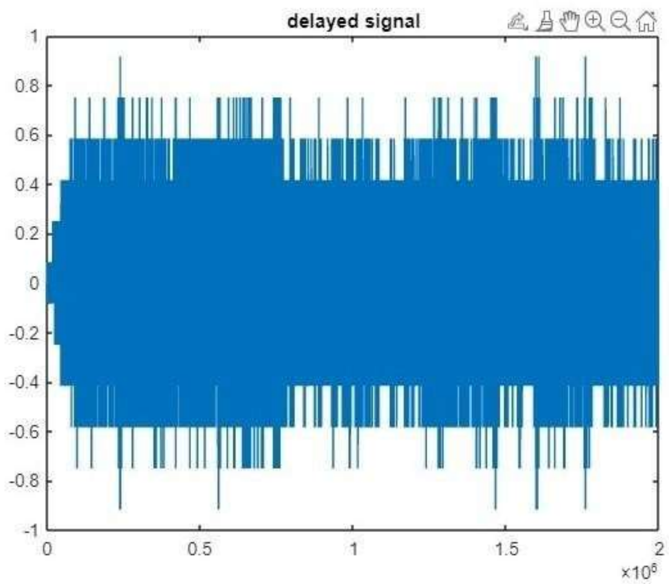

Fig 7: Delayed Signal Plot

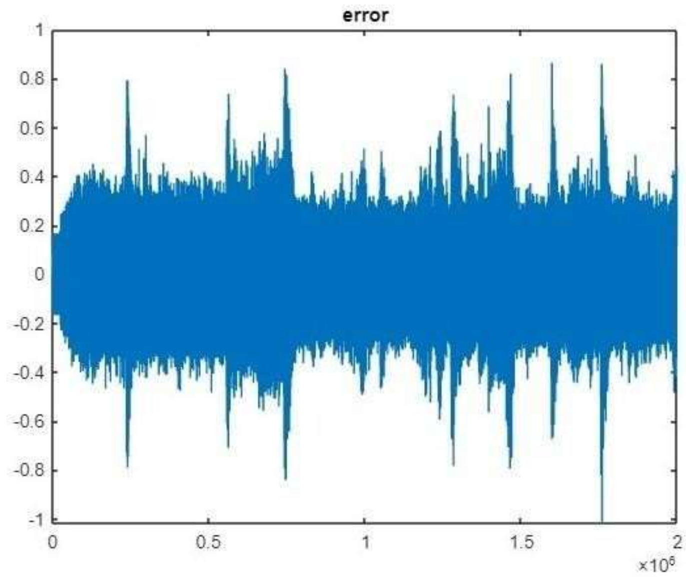

Fig 8: LMS Error Signal

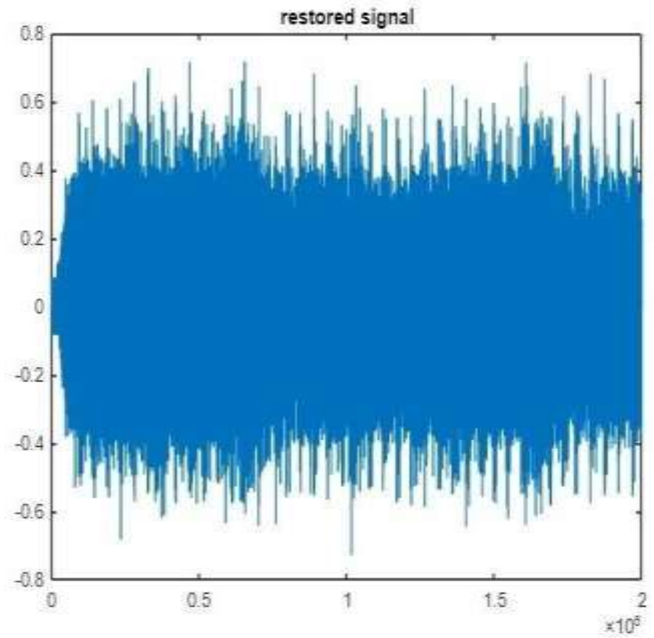

Fig 9: LMS Estimate- Restored Signal

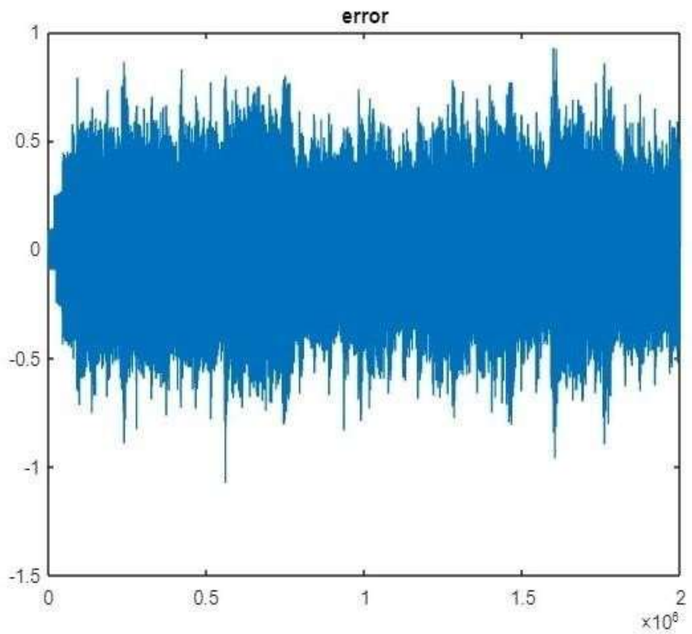

Fig 10: NLMS Error Signal

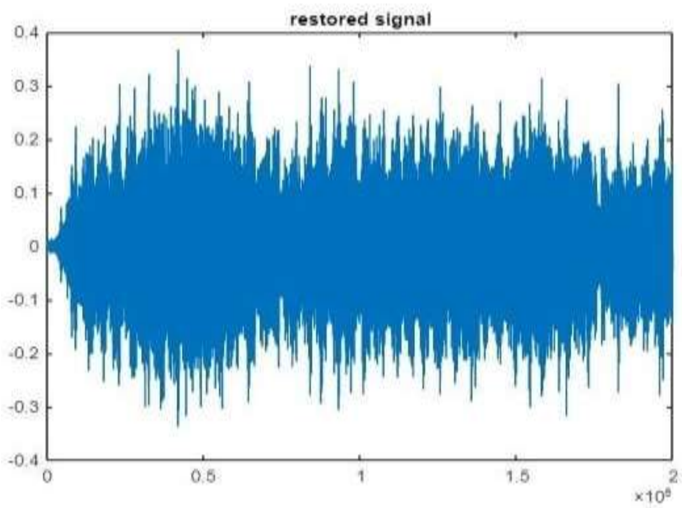

Fig 11: NLMS Estimate- Restored Signal

8. APPLICATIONS OF ADAPTIVE FILTERING

1. Adaptive filter is used in system identification which is an approach to model an unknown system.
2. It is used in linear prediction which estimates the value of a signal at a future time.
3. It is used in inverse modelling which is used in the area of channel equalization for example, it is applied in modems for channel distortion.
4. It can be a powerful tool for the rejection of narrowband interference in a direct sequence spread spectrum receiver.
5. It is used in telecommunication in echo cancellation.
6. Adaptive filtering can be powerful tool for the rejection of narrowband interference in a direct sequence spread spectrum receiver.
7. The adaptive noise cancellers are used to eliminate intense background noise. This configuration is applied in mobile phones and radio communications, because in some situations these devices are used in high-noise environment.

9. CONCLUSION & FUTURE SCOPE

From this analysis it is concluded that a noise reduction method for audible signal by applying adaptive filter technique. The noise reduction problem has been done by a filtering problem which is efficiently solved by above method. The Least mean square LMS, and the Normalized Least mean square NLMS algorithm are implemented, analysed and compared against each other. Based on the implementation results, it concludes that the adaptive noise cancellation application without reference signal, benefits more from the use of the NLMS algorithm instead of the LMS algorithm.

In addition, the method pays attenuation to the non-stationary nature of some audio signal. Simulation results indicate that the proposed method can improve the performance the quality of noisy audio signal. Through computer simulations, it is demonstrated that the proposed method is quite effective in noise reduction, especially in the case of stationary while Gaussian noise.

An inaccuracy in noise assessment may result in the instalment of acoustic meta-structures with an incorrect design. For proper functioning changes can be made in the filter itself, by choosing the different type of filter and the order of the filter. As the noise cancellation is done on non-vocal noise, it can also be achieved in various applications where the pure form of signals are required. ECG, EEG are such examples.

In this report the value of $\mu = 0.02$, which can also changed but the result be different from the nearest value of original signal. Changes can also be made in order of step size rather than making changes in step size which can reduce the steady-state error as it is always observed while changing the step size. Apart from this different algorithms like RLS, FTRLS, TVLMS, VVNLMS can be applied for better results.

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