Audio Quality Enhancement Using Adaptive Filters

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Abstract- Audio quality enhancing plays a vital role in the field of speech recognition, communication, medical etc. The most widely used method is an optimal linear filtering that can reduce the noise level present in an audio signal and improve its signal to noise ratio (SNR). Here, we propose Adaptive LMS filtering method that can improve SNR and enhance audio quality in a very fruitful manner. The signal is filtered at once and the filter coefficients are computed adaptively in an exponential algorithm. The simulation results show a higher audio quality than the raw noise signal. Almost all practical signalling applications are difficult to implement. In this article, we propose a method to reduce noise in audio or speech signals using LMS adaptive filtering algorithms. The signal is filtered at once and the filter coefficients are computed adaptively in an exponential algorithm. The simulation results show a higher quality than the raw noise signal.

Indexed Terms- ADAPTIVE NOISE FILTER, LMS ALGORITHM, FIR FILTER

I. INTRODUCTION

Noise is a common noise as the number of industrial facilities such as motors, transformers, compressors, and blowers increase. The traditional approach to acoustic noise cancellation is to manually remove unwanted noise such as housings, barriers, and noise. Removal of noise over a wide range of frequencies is important, but this is inefficient and expensive in the low-frequency region. Mechanical vibration is a kind of noise that causes problems in all areas of communications and electronic devices. Signals are carriers of useful or unwanted information. Extracting or improving useful information from a mixture of conflicting information is the simplest form of signal processing. Signal processing is an operation designed to extract, enhance, store, and communicate useful information. So signal processing is usually application-specific. Adaptive filters are not known and do not have constant filter coefficients, unlike traditional filter design methods. A filter with tunable parameters is called an adaptive filter. Adaptive filters can be implemented as finite impulse response grids (spruce), infinite impulse response (IIR), grid and domain filters. The most common types of adaptive filters are the LMS (Least Rectangular Section) algorithm and the cross filter using the NLMS algorithm. This report defines noise as an unwanted signal, whether it is electrical, sound, vibration, or any other type of environment. In this report, we applied an adaptive algorithm to different types of noise. The main idea of an adaptive noise reduction algorithm is that the corrupted signal passes through a filter that tries to suppress the noise, and the signal itself remains unchanged. It is an adaptive process and does not require significant knowledge of signal or noise characteristics. Adaptive Noise Cancellation (ANC) effectively absorbs ineffective low-frequency noise.

II. LITERATURE REVIEW

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For adaptive filtering, spruce filters and IIR filters can be used, but FIR filters are the most practical and widely used. The reason is that since the spruce has an adjustable zero point, there are no stability issues and it is stable with an adjustable polar adaptive IIR filter. Literature review of this article, Basic Noise Removal Algorithm, Oldest Average Square (LMS) Algorithm, Basic Noise Removal Algorithm with Adaptive Filter. A noise reduction simulation was developed using the LMS adaptive filter algorithm. The corrupted speech signal and motor noise signal are used as inputs to the LMS adaptive filter algorithm. Compares the filtered signal to the original speech signal to extract the attenuation level of the noise signal. As a result, the noise signal is successfully suppressed by the adaptive filter. The sound dimension of a sound is diffuse and has a filtered signal and a filtered signal, and consequently, the filtered signal approaches the speech noise signal when adaptive filtering is applied. Successfully suppressed noise for the LMS adaptive filter algorithm is determined by Quick-fire signal transform (FFT).

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The LMS adaptive filter algorithm shows significant changes in noise at low frequencies. 2 This project involves the study of Adaptive Noise Reduction (ANC) and its application principles. ANSIS de-adaptation is an alternative method for evaluating signals damaged by noise or interference. The advantage is that the signal or noise estimate is not evaluated and noise cannot be achieved and cannot be achieved with other denoising signal processing methods. Cost is not necessarily unavoidable, it is a reference input with noise that contains the noise in an unknown way along with key noise and noise that requires two inputs from the main input, including the distorted signal. The cold input is filtered with an adapter to get a signal estimate from the main input. Adaptive filtering for subtraction allows you to process critical or probabilistic, probability, fixed, or qualified input data. In the primary and reference inputs, the impact of the signal components of the reference item has investigated the performance of non-corrosion noise and ANC. If the NECROTISING Noise standard does not have a signal, you can significantly remove the noise of the main input without distorting the signal. The configuration of the adaptive noise of the citrus that does not require the reference entry is not required and it appears in many applications. When using a notch filter, it contains the quantitative analysis depth, exceeds the offset, or when you remove the offset or remove it with the adaptive line, other applications cancel noise, such as regular interference, violence, antenna, an adaptive filter of antennas, cancelled interference, and adaptive filters of remote ANCs, to cancel noise such as voice signals. Computer simulations for all cases are performed using MATLAB software and experimental results demonstrate the usefulness of adaptive noise reduction techniques. The (LMS) named algorithm methodology is popular and widely used because of its reliability and simplicity. It is known as a simple context is a fixed context and is known for good performance [12].

III. METHODOLOGY

a. LMS ALGORITHM

The least-mean-square (LMS) algorithm is widely used in adaptive signal processing for its robustness and simplicity. It is known for its simplicity and its good steady-state performance in stationary context. Consider in general an N-tap filter, with the weight vector w(n) at time instant n denoted by,

$w(n) = [w_1(n)w_2(n) \dots w_N(n)]$ T

Let {x(n)} be the input sequence x(n) = [x(n) x(n-1) ……… x(n-N+1)] be its vector representation containing the immediate past N samples of {x(n)}.

The filter output $y(n) = W^T(n)x(n)$aims to follow a desired signal and the estimation error e(n) is defined

by

$e(n) = d(n) - y(n)$

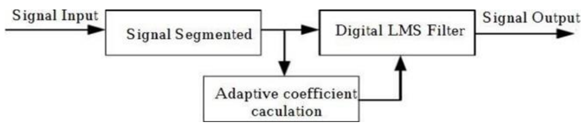
An adaptive filtering algorithm adjusts the filter tap weight w(n) at each time in- stand according to the measured value of e(n). The standard LMS algorithm up- dates as

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

where mu(u) is defined as the step-size parameter which affects the convergence behaviour of the filter weights.

b. NOISE REDUCTION ALGORITHM BASED ON LMS FILTER:

We propose a noise cancelling scheme based on LMS filtering algorithm of its optimum performance. The block diagram of the noise reduction method is shown in figure. Most audio signals are time varying signals, in order to achieve effective noise reduction with LMS filtering method, the input signal must be segmented. The unprocessed noisy signal is segmented every 40ms. Let y= {y: t=1,2,…..T} be a noisy test signal with T frames and being the frame at time t



In the proposed method, the problem of noise cancellation can be stated as identifying for each noisy frame y(t) a matching weight vector w(t). Since a whole unit with several frames, when treated as a segment of consecutive speech frame, can be identified more accurately from noise than the whole frames. In the LMS filter we employ the normalized least mean square (LMS) algorithm. The LMS is a version of the well-known LMS algorithm that normalizes the weight vector updates with respect to the squared norm of the regressor. This normalization makes the LMS algorithm less sensitive to variations in the input power of the adaptive filter. Therefore, the LMS algorithm is a good candidate for applications with a high degree of uncertainty about the filter input power. The statistical analysis of the LMS algorithm is complicated by the normalized weight update.

The LMS algorithm can be reviewed as a special case of the LMS algorithm with a time-varying step size, in which the step size varies with the input signal strength. The tap-weight adaptation equation of the LMS algorithm is given by

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

Where μ is a parameter to be chosen and $\frac{\mu}{x^T(n)x(n)}$ is the actual step size.

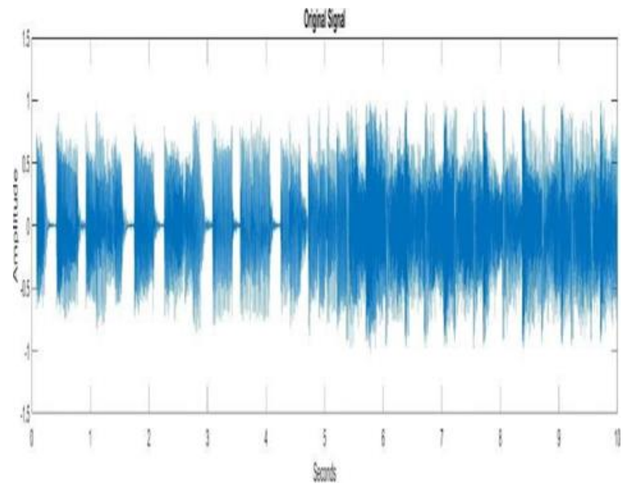
The output signal of the proposed method is.

IV. RESULTS

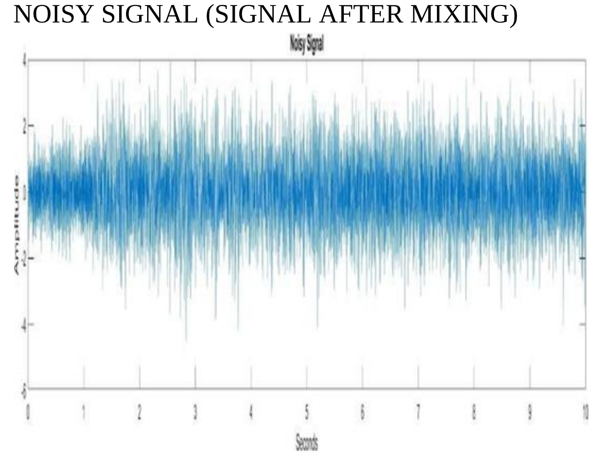
This normalization makes the NLMS algorithm less sensitive to changes in the input power of the adaptive filter. As a result, the NLM algorithm is suitable for applications with high uncertainty of the filter input power. Statistical analysis of the NLM algorithm is complicated by updating the fully qualified balance. Algorithm LMS can be seen in a special case in the chain frequency of the LMS algorithm. Here, the step size depends on the power of the input signal. LMS Algorithm Duret Adaptive Almine As a result, it is, leading to the obtained algorithm of the adaptive LMS filter, and gradually relaxes the noise and gradually reverses the unarmed engine noise. The noise is corrupted by the voice signal. The results show that clean speech signals can be obtained after applying the adaptive LMS filter.

- OUTPUT GRAPHS:

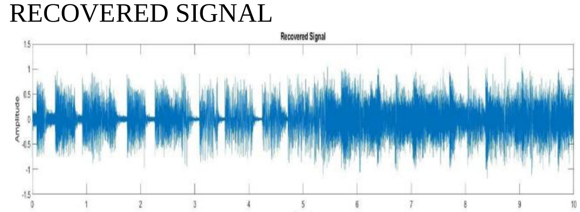
OUTPUT SIGNALS



NOISY SIGNAL (SIGNAL AFTER MIXING)

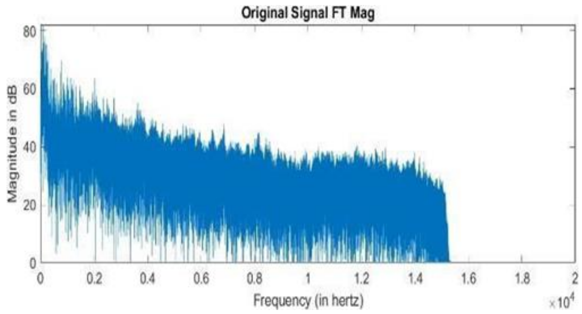


RECOVERED SIGNAL

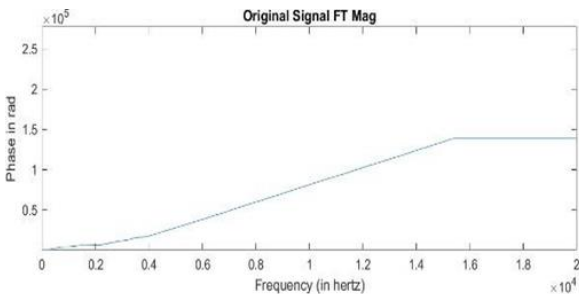


FOURIER TRANSFORMS OF SIGNAL:

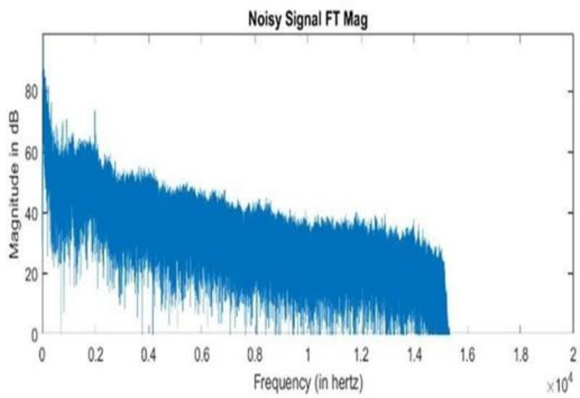
MAGNITUDE RESPONSE OF ORIGINAL SIGNAL



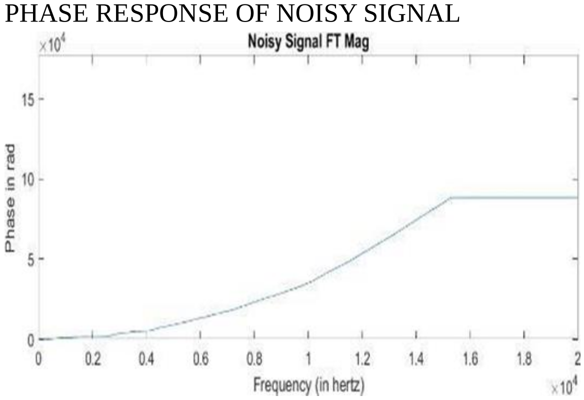
PHASE RESPONSE OF ORIGINAL SIGNAL



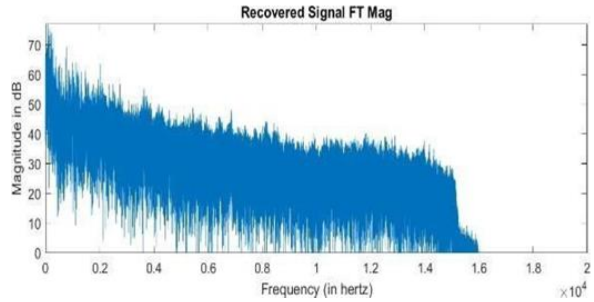
MAGNITUDE RESPONSE OF NOISY SIGNAL

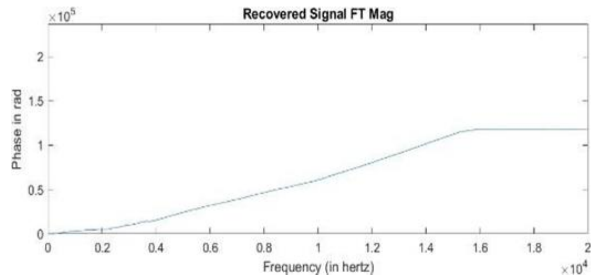


PHASE RESPONSE OF NOISY SIGNAL



MAGNITUDE RESPONSE OF RECOVERED SIGNAL PHASE RESPONSE OF RECOVERED SIGNAL





V. DISCUSSIONS

In this project, we have done the literature survey on audio enhancement using adaptive filter, studied different research paper on this topic, studied about the LMS algorithm the mathematical equation used in MATLAB software filters the noise signal and convert it into pure output without any noise.

CONCLUSION

In this report, we have presented a noise reduction method for audio and speech signals by applying adaptive linear filtering technique. The noise reduction problem has been formulated as a filtering problem which is efficiently solved by using the LMS method. In addition, the method pays attention to the nonstationary 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, we have demonstrated that the proposed method is quite effective in noise reduction, especially in the case of stationary white Gaussian noise

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