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Inventive Comparative Speech Analysis Using Adaptive Filter

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ABSTRACT: Current method of speech enhancement has been developed with adaptive filtering approach. The adaptive filter utilizes the least mean square algorithm for noise removal, but in LMS algorithm key parameter is step size. When step size is large speed and least mean square error is large and it is that computational cost increases to an undesirable level as the length of the impulse response increases. This is mainly due to the fact that the algorithm lies in the time domain, leaving the algorithm at an obvious disadvantage when an impulse response is very long. The computational power required simply becomes too high for efficiency of use. Computational power required is very large. In this Project an optimal estimation of adaptive filtering using Unbiased and normalized adaptation noise reduction (UNANR) algorithm will be implemented for the noisy speech. We have to prove that the Unbiased and Normalized adaptation noise reduction (UNANR) algorithm is an efficient adaptive filtering algorithm than least mean square (LMS) algorithm.

KEYWORDS: LMS, UNANR, Adaptive Filtering, Synthetic Speech

I. INTRODUCTION

The speech enhancement tells about the growth of communication system. Enhancement means improvement in the value or quality of something. When we applied it to speech this simply means the improvement in the intelligibility and quality of degraded speech signal by using signal processing tools. By speech enhancement, it refers not only to noise reduction but also to de-reverberation and separation of independent signals. This is a very difficult problem due to two reasons. First, the nature and characteristics of the noise signal can change dramatically in time and between applications. Noise signal measure can also be defined differently for each application. Two criteria are often used to measure performance: quality and intelligibility. It is very hard to satisfy both at time.

Speech enhancement is an area of speech processing where the goal is to improve the intelligibility and pleasantness of a speech signal. The most common approach in speech enhancement is noise removal; where estimation of noise characteristics can cancel noise components and retain only the clean speech signal. The basic problem with this approach is that if we remove those parts of signal that resemble noise. In other words, speech enhancement procedures, often inadvertently, also corrupt the speech signal when attempting to remove noise. Algorithms must therefore compromise between effectiveness of noise removal and level of distortion in the speech signal.

II. RELATED WORK

Previous studies explained concept of speech enhancement in a theoretical approach, using different speech enhancement algorithms. The speech enhancement methods aimed at suppressing the background noise are based on one way or the other on the estimation of the background noise. If the noise is evolving more slowly than the speech, it is easy to estimate the noise during the pauses in the speech. If the noise is varying rapidly then estimation is more difficult. This paper addresses the problem of reduction of additive background noise in speech. Here speech enhancement is implemented using three methods. They are speech enhancement using spectral subtraction, speech enhancement using Wiener filter and Kalman filter respectively.

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This concept explained the adaptive filtering constitutes one of the core technologies in digital signal processing and finds numerous application areas in science as well as in industry. Adaptive filtering techniques are used in a wide range of applications, including echo cancellation, adaptive equalization, adaptive noise cancellation, and adaptive beam forcing. Acoustic echo cancellation is a common occurrence in today's telecommunication systems. The signal interference caused by acoustic echo is distracting to users and causes a reduction in the quality of the communication. It focuses on the use of LMS and NLMS algorithms to reduce this unwanted echo, thus increasing communication quality.

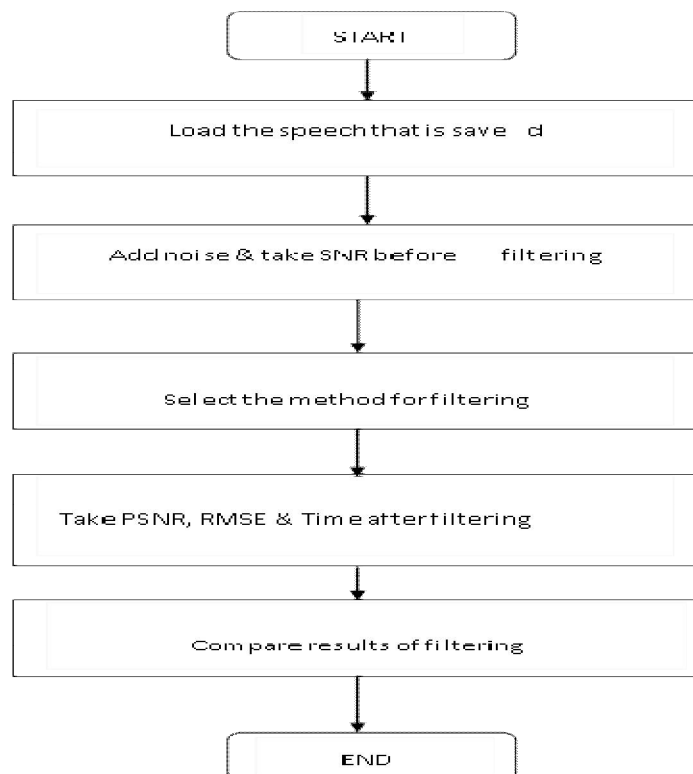
The concept described and implemented that a novel constrained-stability least-mean-squares algorithm for filtering speech sounds is proposed in the adaptive noise cancellation problem. It is based on the minimization of the squared Euclidean norm of the weight vector change under a stability constraint over the a posteriori estimation errors. To this purpose, the Lagrangian methodology has been used in order to propose a nonlinear adaptation in terms of the product of differential input and error.

Researcher studied a blind dereverberation method based on spectral subtraction using a multi-channel least mean squares (MCLMS) algorithm for distant-talking speech recognition. In a distant-talking environment, the channel impulse response is longer than the short-term spectral analysis window. By treating the late reverberation as additive noise, a noise reduction technique based on spectral subtraction was proposed to estimate the power spectrum of the clean speech using power spectra of the distorted speech and the unknown responses.

III. METHODOLOGY/ALGORITHM

The aim of this system is to implement various adaptive noise cancellers for speech enhancement based on gradient steepest descent approach using Unbiased and Normalized adaptive noise reduction. The speech enhancement method will be aimed at suppressing the background noise is evolving more slowly than the speech.

System Flow



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UNANR- Unbiased and adaptive noise reduction

The UNANR model of the system performs the function of adaptive noise estimation. The UNANR model of order M, as shown in Fig 5.4, is a transversal, linear, finite impulse response (FIR) filter. The response of the filter $f(n)$ at each time instant (sample) n can be expressed as,

$$f(n) = \sum_{m=1}^M w_m(n) r(n - m + 1) \text{----- (1)}$$

Where $w_m(n)$ represents the UNANR coefficients, and $r(n - m + 1)$ denotes the reference input noise at the present ($m = 1$) and preceding $m - 1$, ($1 < m = M$), input samples. In order to provide unit gain at DC, the UNANR coefficients should be normalized such that

$$\sum_{m=1}^M w_m(n) = 1 \text{----- (2)}$$

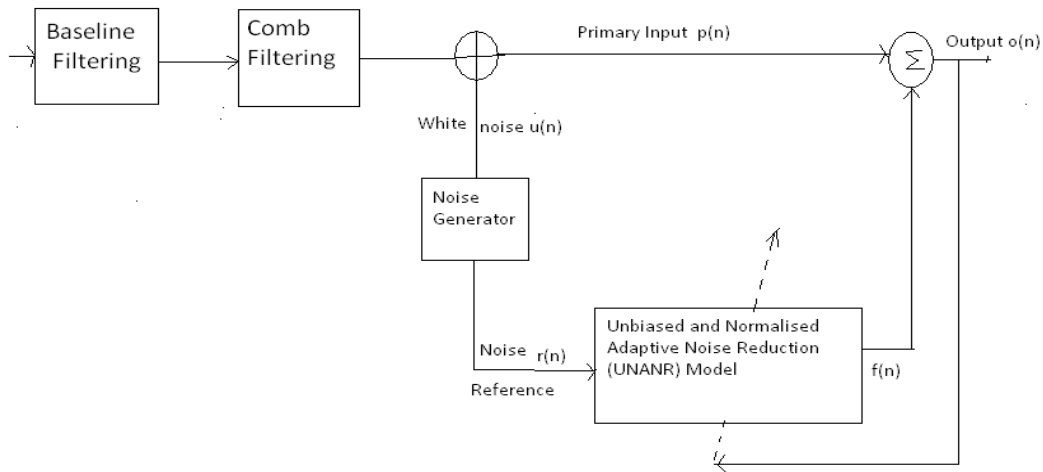


Fig: - Overview of Adaptive noise-reduction system

The adaptation process of the UNANR model is designed to modify the coefficients that get convolved with the reference input in order to estimate the noise present in the given speech signal. To provide the estimated speech signal component $s(n)$, at the time instant 'n', the output of the adaptive noise-reduction system subtracts the response of the UNANR model $f(n)$ from the primary input $p(n)$, i.e.,

$$\hat{s}(n) = o(n) = p(n) - f(n) \text{----- (3)}$$

IV. CONCLUSION

InSpeech enhancement the LMS filter algorithm and UNANR filter algorithm is introduced as the main adaptive algorithm in the time domain and its operation has been examined. There are mainly three parameters PSNR, RMSE and Time are considered for performance analysis .The comparison of these different parameters considered for filtering speech signal using LMS filter and UNANR Filter is done.This comparison shows that the signal to noise improvement in the input signal after UNANR filtering is much higher than that LMS filter algorithm. The adaptation process of the UNANR filter is designed to modify coefficients by using sample to sample analysis. The UNANR

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model is also having simple implementation compared to that of LMS filter algorithm. The convergence rate of LMS algorithm compared to UNANR algorithm is also high. UNANR filter model gives better performance as compared to LMS filter model.

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