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A Novel Algorithm for the Reduction of Irregular Noise in Corrupted Speech Signals

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Abstract: One of the main problems associated with the use of two-sensor noise cancellation systems is the nature of the noise signals. This problem imposes the use of high complexity algorithms to reduce the noise in useful signals. This can be impractical for many real time applications, where computational power is a critical issue. Most of existing literature approaches is based on a single and usually complex adaptation algorithm to do the job. In this paper, a new mechanism is devised to eliminate background noise from speech communications. The procedure is based on a two-sensor adaptive noise canceller that is capable of assigning a suitable algorithm according to properties of the noise. The criterion used here is based on calculating the eigenvalue spread of the autocorrelation of the input noise. The new smart noise canceller (SNC) applies a suitable adaptive algorithm according to the eigenvalue characteristics of the input signal. This approach showed its capability in executing noise cancellation under different types of environmental noise. Fast convergence rates, improvement in signal-to-noise ratio and substantial reduction in computational power are obtained using this SNC technique. Experiments are conducted using real life signals to demonstrate the success of the method.

Keywords: Adaptive filtering; Noise cancellation; Eigenvalue spread; Environmental noise.

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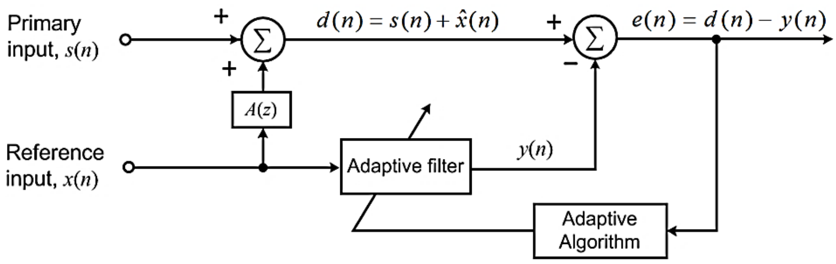
1. Introduction

Passive techniques of noise cancellation are expensive and have proven to be effective only for certain levels of noise within a limited range of frequencies [1]. In this context, noise cancellation using adaptive filters has become a viable option for many noise cancellation applications used in modern technologies. This technique offers a lower costs and more practical means of achieving the required suppression of noise than the passive methods. An adaptive algorithm is used to control the coefficients of a digital filter. The aim here is to remove or suppress noise from a target signal. The concept of adaptive noise canceller (ANC) relies on using an auxiliary or reference signal acquired from one or more sensors located at a vicinity of a noisy area where the signal of interest is weak. Then, the adaptive filter is used to filter out the noise and hence improving the quality of the target signal [2]. A basic model of adaptive noise canceller is shown in Figure.1.

Figure 1. The basic diagram of two-sensor noise cancellation system.



The choice of a particular adaptive algorithm is usually based on its convergence speed as well as its computational power. The convergence speed of the classical least mean square LMS adaptive filter becomes very slow for ill conditioned input signals [3]. Alternative algorithms such as the recursive least square (RLS) and the affine projection (AP) algorithms are proposed in literature to attain the best performance of adaptive filters [4-6]. However, this comes at the expense of increasing the computational burden.

This paper is related to application of physical sensors; make use of sound sensors (microphones) to detect acoustics noise in speech signals and attempts to reduce it. In this paper, the development of a novel, smart noise cancellation system for eliminating irregular background noise from speech signals is presented. The proposed method possesses a great advantage in real-time audio applications; no other literature has discussed the implementation of such a technique so far. Most existing literature works went for too complex, merely theoretical and non-applicable in real-time due to complexity. The method presented here is based on using a selective mechanism that can be switched to apply several adaptive filters. This is based on measuring the characteristics of the noise. The main task of this research deals with the identification of the changes in the noise signals, and applying an appropriate adaptive algorithm accordingly.

2. Experimental Section

2.1. Measurement of Eigenvalue Spread

The proposed selection mechanism is based on the calculation of eigenvalue spread in order to select an adaptive algorithm intelligently for eliminating regular and irregular types of noise from noisy signals. The target application here is speech communications, where the useful signal is corrupted with irregular types of noise that are hard to eliminate using conventional methods.

The eigenvalues spread is determined from the autocorrelation matrix **R**, denoted by

$$\mathbf{R} = E\left[\mathbf{n}(k)\mathbf{n}^H(k)\right] = \begin{bmatrix} E\left[\left|n_0(k)\right|^2\right] & E\left[n_0(k)n_1^*(k)\right] & \cdots & E\left[n_0(k)n_M^*(k)\right] \\ E\left[n_1(k)n_0^*(k)\right] & E\left[\left|n_1(k)\right|^2\right] & \cdots & E\left[n_1(k)n_M^*(k)\right] \\ \vdots & \vdots & \ddots & \vdots \\ E\left[n_M(k)n_0^*(k)\right] & E\left[n_M(k)n_1^*(k)\right] & \cdots & E\left[\left|n_M(k)\right|^2\right] \end{bmatrix}. \tag{1}$$

Here, $\mathbf{n}^H(k)$ is the Hermitian transposition of input signal $\mathbf{n}(k)$. The eigenvalues spread is determined from the ratio of the maximum to the minimum eigenvalues of the matrix **R**. The eigenvalues are denoted by λ_j. We first establish the characteristic equation of **R** as follows.

$$\det(\mathbf{R} - \lambda_j \mathbf{I}) = 0 \tag{2}$$

where **I** is the identity matrix, and λ_j is given by following diagonal matrix,

$$\lambda_j = \begin{bmatrix} \lambda_1 & & & 0 \\ & \lambda_2 & & \\ & & \ddots & \\ 0 & & & \lambda_M \end{bmatrix}. \tag{3}$$

where, $\lambda_1, \lambda_2, \ldots, \lambda_M$ are the eigenvalues of **R,** and all of which may not be distinct from each other. Then, the eigenvalue spread of **R** is calculated as follows,

$$s(\mathbf{R}) = \frac{\max(\lambda_j)}{\min(\lambda_j)} \tag{4}$$

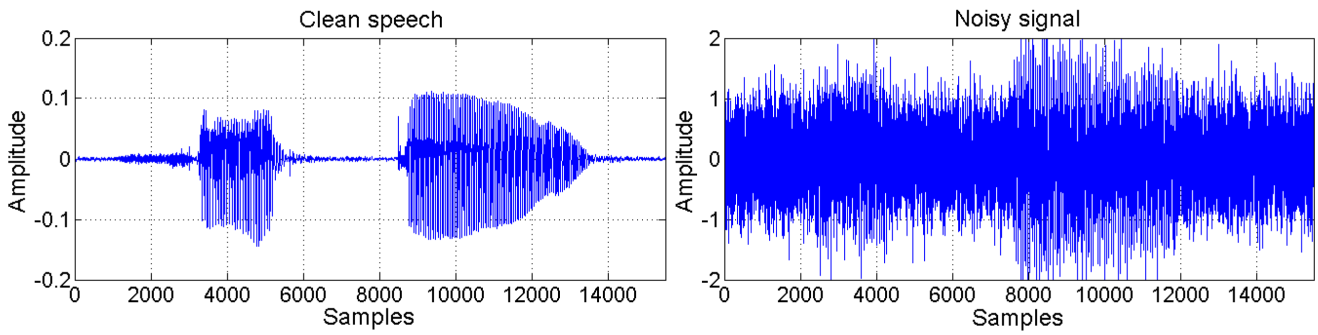
where $\max(\lambda_j)$ and $\min(\lambda_j)$ are the maximum and minimum eigenvalues of the autocorrelation matrix, respectively. Using the measurement of $s(\mathbf{R})$, the selection mechanism is set for canceling different types of noise in the noisy speech signal.

2.2. Selection Mechanism of the Proposed Algorithm

Basically, the adaptive algorithms used in this work are the LMS, RLS and the Affine Projection (AP) algorithms. This can be extended to include more advanced algorithms such as recent variants of these algorithms. The LMS algorithm is applied during intervals when the eigenvalue spread is quite low, which is considered to be the best case, this algorithm works efficiently with white noise. The RLS is applied when the eigenvalue spread is very high, which is regarded as the worst case. Between these two situations, the AP is applied such that it would not cross some predefine projection order. The benefit of applying the AP with low projection order comes about from the fact that the AP would reduce colored noise with mild computational complexity, sometimes similar to that of the normalized LMS as long as the projection order is kept as low as possible. The performance of the RLS is independent of eigenvalue [7]; therefore, it is assigned to the noise signal which has large eigenvalue spread. The discussion here is kept minima due to space constraint.

The proposed smart noise canceller abbreviated as SNC is simulated and tested to evaluate the performance of the algorithm under real-life environmental conditions. The SNC selects an adaptive algorithm intelligently based on a flag setting to apply an appropriate algorithm. To judge the performance of the proposed SNC, a comparison is made with solo algorithm systems. The solo algorithm systems used in the comparison are simulated and tested under the same conditions. These algorithms are the NLMS, the RLS and the APA. The comparison with the AP is thought to give a fair judgment with recently devised adaptive algorithms [9, 10]. The target signal used in this work is a Malay utterance “Satu”, sampled at 16 kHz. The noisy speech signal subjected to several types of background noises such as white noise, car noise, voice babble and a pink (colored) noise. These noise signals are sampled at 16 kHz and concatenated to produce a noise with variable characteristics. The waveform of clean and noisy speech signals are shown in Figure 2. The eigenvalue spread of the input noise signals used in this simulation, are calculated using 125 data per frame, with 60 frames for every noise signal. The calculation of the eigenvalue spread is repeated to observe the changes in the noise signals.

Figure 2. The waveform of clean speech signal of the word “SATU” and noisy signal.



3. Results and Discussion

The performance of SNC is evaluated using mean square error (MSE). Figure 3(a) shows the MSE performances of the proposed method compared to other single algorithms. The SNR input was fixed at −15dB. From this figure, the SNC convergence shows a similar behavior to that of the RLS at the beginning and it converges faster than other algorithms at the rest of the operation.

Figure 3(b) illustrates the processed speech using different algorithms to control the adaptation process. Based on the results of the variable noise, this approach is capable of computing different algorithm when the properties of the noise signal are changed. The little spike in the middle of the speech processed by the proposed SNC is due to transient when switching from one algorithm to the other, but this has shown negligible effect on subjective hearing tests.

At the primary input of the SNC, different levels of signal-to-noise ratios (SNR) were added to observe the performance of the proposed approach. It is evident from subjective tests that the system possesses better performance, and the noise at the output system is efficiently eliminated using the SNC. The results of SNR are omitted due to space limitations.

Computational complexities of the LMS, RLS, APA and proposed SNC algorithms are shown in Table 1. The table also shows a sample calculation of the computational complexity based on the number of multiplications per unit sample. The calculations are made using parameters used in the experiments, namely filter length N = 32 and projection order for AP, M = 4 compared with that of solo algorithm. For the worst case, we had the SNC performing approximately one third of the time as RLS. The remaining time performs approximately similar to LMS or NLMS in case of AP. It is clear that the proposed system devised in this paper has nearly 65% reduction in computational complexity compared to that of the RLS which is normally used to remove irregular noise from corrupted signals.

Figure 3. Results of proposed approach and other single algorithms for the case of variable background noise. (a) MSE performance (b) Output signals

(a) MSE performance under variable noise (b) Processed speech

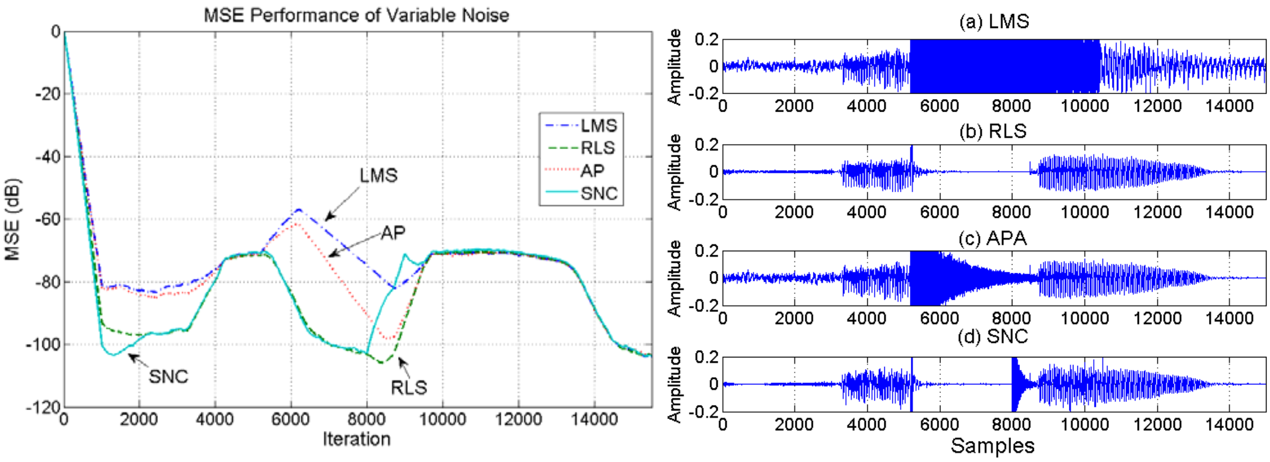


Table 1. Computational complexity of the solo algorithms, SNC algorithm and a sample calculation.

Algorithm	Computational Complexity		
	Additions	Multiplication	Sample calculation
LMS	$2N+1$	$2N+1$	65
RLS	$3N^2+11N+9$	$3N^2+7N+9$	3305
AP	$(M^2+2M)N+M^3+M^2-M$	$(M^2+2M)N+M^3+M^2$	848
SNC	$\frac{3N^2+15N+12}{3}$	$\frac{3N^2+11N+11}{3}$	1145

4. Conclusions

The paper presented novel noise canceller. The algorithm based on measurement of eigenvalue spread for eliminating theoretical as well as real life noise. Results showed capability to remove regular and irregular noise by applying an appropriate algorithm. The SNR performance showed improvements up to 10 dB at the output system. The convergence of the proposed system outperformed that of other algorithms. The advantage of the computational complexity is a reduction of almost 65% of the RLS. The idea can be extended to include more variants of existing algorithms.

Conflicts of Interest

The authors declare no conflict of interest.

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