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Real time speech noise filtering by UNLMS

Suman Rani[1], Parveen Khanchi[2]

[1, 2]Department of ECE, RIMT, Chidana, India,

Abstract

Signal processing has become an important tool in almost all fields of science and engineering. The characteristics of signals/systems are either not known or change with time. Processing techniques/algorithms should adapt to the unknown characteristics which may be time invariant or variant. ANC is a technique used to remove a unwanted noise from received signal. ANC is typically a dual-input, closed-loop adaptive feedback system. The Least Mean Square adaptive algorithm is the most widely used real time filtering algorithm due to its computing requirements which is further updated with normalized mean square algorithm (NLMS). In our work we presented a new updated NLMS algorithm based on change in weights and biases of filter.

Keywords: Signal processing; ANC; NLMS

1. Introduction

The area of Real-Time Adaptive Signal Processing involves the use of optimum statistical signal processing techniques to design signal processing systems that can modify their characteristics, during normal operation(usually in real time), to achieve a clearly predefined application dependent objective. Signal processing has become an important tool in almost all fields of science and engineering. Since the characteristics of signals/systems are either not known or change with time. In some case, like speech their very changing characteristic nature is of utility and importance. In cases like noise control, echo cancellation and long distance communication, the time variant behavior of the system/signal involved is very undesirable. Therefore, processing techniques/algorithms should adapt to the unknown characteristics which may be time invariant or variant. Hence to extract valid information in a changing scenario, algorithms suitable for time invariant case have to be made adaptive to preserve their performance. The adaptive algorithms should be: simple, computationally efficient, implementable on the existing hardware platform and cost effective, thus any real time processing has to be adaptive. A block diagram of adaptive noise canceller is shown in figure 1.

In this setup, the signal path from the noise source is passed to the primary sensor as an unknown FIR channel H. The adaptive filter to the noise recorded at the reference sensor, and then an adaptive algorithm is used to train the adaptive filter to match or estimate the characteristics of the unknown channel H.

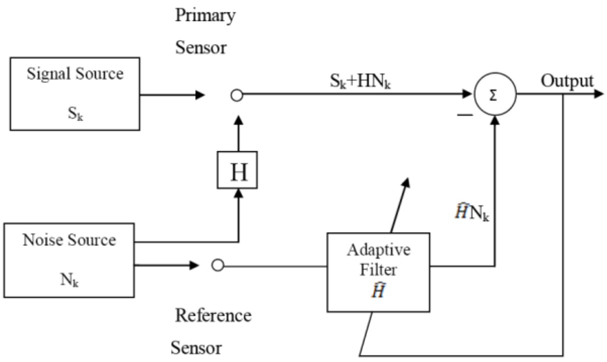

Figure 1: Adaptive Noise Canceller

If the estimated characteristics of the unknown channel have negligible differences as compared to the actual characteristics, the noise components in the corrupted signal can be cancelled to obtain the desired signal. There are various application areas fo adaptive noise cancellation work like Adaptive Cancelling ECG in Heart-Electrocardiography, Adaptive Acoustic Echo Cancellation, Noise Cancelling in Cell Phones, Adaptive Noise Control in Jet Aircraft etc.

There is always scope in the improvement of noise cancelling filter which is our motivation to develop a more adaptive filter. In next section we discussed the proposed methodology to develop our adaptive filter and section 3 discusses its outcome followed by conclusion section.

2. Proposed methodology

Efficiency of an adaptive filter algorithm depends upon the error size which is calculated by subtracting the desired signal with filter output. We know lesser the error size better the cancellation. As in normalized least mean square algorithm the cancellation of noise signal is good but error size is not much small. In proposed algorithm we have further reduced the error size.

LMS algorithm has a drawback that its step size is not time varying which makes it very hard to make it stable. The NLMS algorithm is a derivative of LMS algorithm which can solve the problem of LMS by normalizing the input power which is described in previous chapter. In NLMS the step size is varied depending the input vector only to normalize the input vector. Although this process is followed to reduce the error but it doesn't consider the previous step size and weights which are driving the input vector to less error prone results. As it is known form previous chapter filter weights are updated by formula ;

$$w(n+1) = w(n) + 2\mu e(n)x(n) \qquad (1)$$

where μ is step size, x(n) is the input vector and e(n) is the error. This updation of weighting factor is formulated in such a way so that error reduces. It's like a feedback system in which previous error and weight is considered to get the next weight value. We have borrowed the concept of step size calculation form the same philosophy. The step size is updated in each iteration depending upon the previous step size and error. The mathematical formulation for the suggested by us is not observed in any other paper in best of knowledge. The formula for improvement of step size is:

$$\mu = \left(\frac{1}{dot(\mu',\mu)}\right) * (\lambda\sqrt{\mu} + (1-\lambda^2)e^2 \qquad (2)$$

where λ is a constant whose value is kept 0.999.

In this formulation the step size is the multiplication of normalized input vector as stated in LMS and sum of previous step size with squared error. A term (1-λ^2) is multiplied with square of error so that if error is high then also step size remains at low limit. The initial value of weighting factor and step size is set at the final value achieved for both in case of NLMS. Proposed work takes place mainly in five steps which are

1. Initially, set each weight *Wn* (i), where i= 0,1,..,N-l, to final value set in NLMS.

2. Compute filter output

$$y(n) = \sum_{i=0}^{N-1} w(n)xx(n-i) = x(n)$$

3. Compute the error estimate

$$e(n) = d(n) – y(n)$$

4. Update the step size as

$$\mu = \left(\frac{1}{dot\,(\mu',\mu)}\right) * (\lambda\sqrt{\mu} + (1-\lambda^2)e^2)$$

5. Update the next filter weights

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

The designed MATLAB code for above process is shown below

```
lambda=0.999;

for n=M:N

uvec2=u(n:-1:n-M+1);

out2(n)=(w2'*uvec2);

e2(n)=d(n)-out2(n);

mu2=(1/dot(uvec2',uvec2)).*(lambda*sqrt(mu2)+(1-
lambda^2)*e2(n));
```

```
w2=w2+sqrt(mu2).*uvec2.*conj(e2(n));

err2(n)=mse(e2);

end
```

Flow Chart of improved NLMS. The flow chart is shown in figure 2.

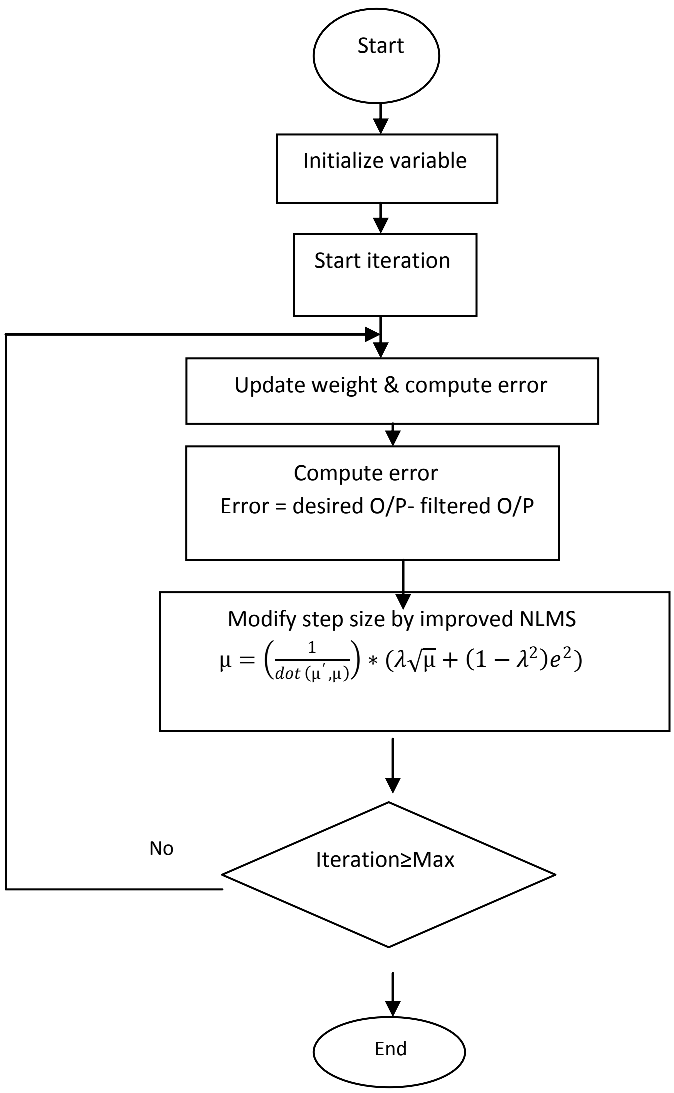

Figure 2: Flow Chart of improved NLMS

3. Results and analysis

We have written a MATLAB script for our proposed work and developed an user interface which records real time sound and filter it with proposed filter. To validate results proposed filter results are compared with NLMS and LMS filters to. The graphical user interface is shown in figure 3 below.

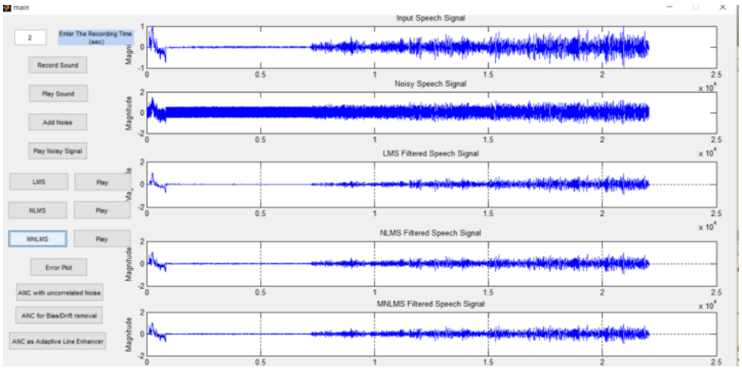

Figure 3: GUI interface of proposed work

The main validation of proposed work is done by comparing the results in terms of mean square error. Filtered output is compared with the recorded signal to get the MSE value. A graph in figure 4 shows the comparative plot for this work.

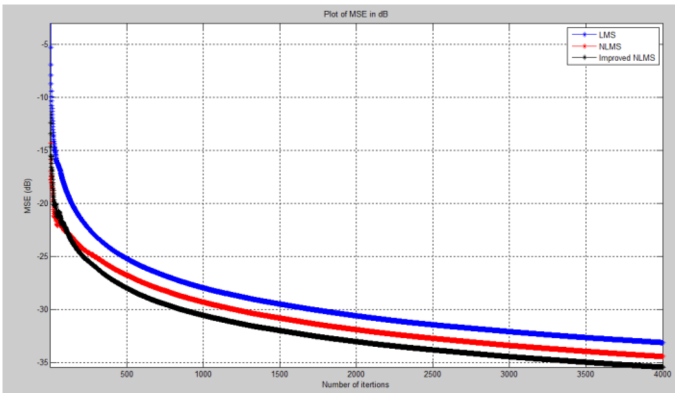

Figure 4: Error scale for LMS, NLMS, & updated NLMS

From the above graph which shows the comparison between error scale for LMS, NLMS, improved NLMS we can conclude that the MSE is very low with the algorithm improved normalized mean square error. In the above graph we can see that the Error is reduced to -36dB.

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The simulated models are set to the following parameters:

Filter Parameters

- Number of epochs = 4000
- Number of taps =60

- Adaptation Step Size Parameter
  LMS Step Size: m = 0.01
- Signal frequency 0.01, noise frequency 0.05
- Input Signals: White, Normally distributed, zero mean, unit variance

Additive Noises: Input sensors noises are simulated as white signals. It has been assumed these additive noises are uncorrelated with the other signals in the adaptive noise canceller.

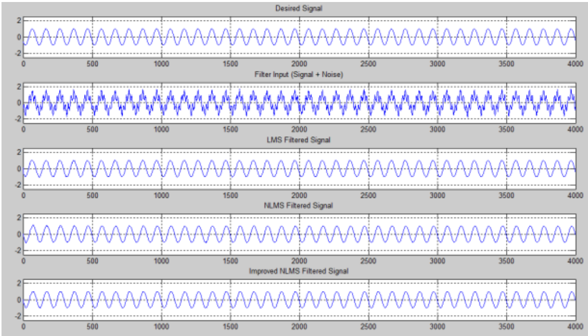

Figure 5: NLMS of Adaptive Noise Canceller

As observed from the above graph, the distortions in the output waveform get reduced to a large extent as compared to the desired waveform. So wiener filter proves to be as an efficient filter. The performance criterion for filtered signal is mean square error (MSE). To achieve desired results MSE should be minimize.

Adaptive noise canceller with uncorrelated noise in primary and reference inputs result

The simulated models are set to the following parameters:

- Number of Iterations = 500
- Order of filter =16
- Adaptation Step Size Parameter LMS Step Size: m = 0.01
- Cut of frequency Wn = [0.1 0.5]

An adaptive noise canceller performance in the presence of correlated and uncorrelated noises is shown in fig 6.

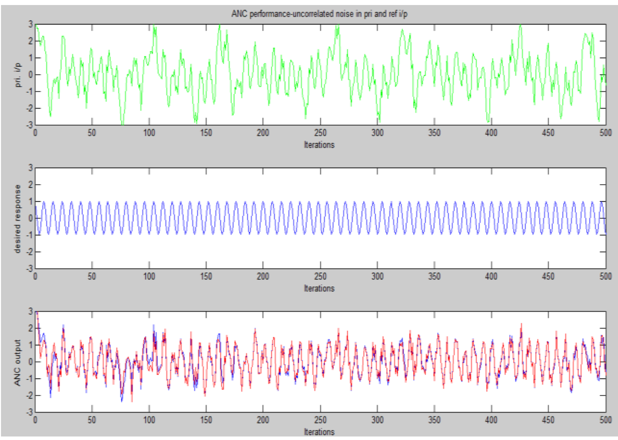

Figure 6: ANC with Uncorrelated Noise in Primary and Reference Input

As discussed in the previous chapter, the ability of a noise canceling system to reduce the noise is limited by the uncorrelated-to-correlated noise density ratios at the primary and reference inputs. Smaller the ratios of the spectra of the uncorrelated to the spectra of the correlated noises at the primary and reference, the greater the ratio of signal-to-noise density at the output and the primary input and the more effective action of the canceller. The desirability of low levels of uncorrelated noise in both primary and reference inputs is thus emphasized.

4. Conclusions

This paper described the design and implementation of Real-Time Noise Cancelling System. LMS Adaptive Filtering is an important basis for signal processing; Adaptive Noise Cancelling is a method of optimal filtering that can be applied whenever a suitable reference input is available. The advantages of this method are its adaptive capability, its low output noise, and its low signal distortion. The adaptive capability allows the adaptive filter are used for estimation of non-stationary signals and systems, or in application where a sample-by sample adaptation of a process and for a low processing delay is required. Output noise and signal distortion are generally lower than can be achieved with conventional optimal filter configurations. In each instance cancelling was accomplished with little signal distortion even though the frequencies of the signal and interference overlapped. Thus Noise Cancelling Technology establishes the usefulness of Adaptive Noise Cancellation in techniques and its diverse application for the development.

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