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Research Article

Speech Processing using Normalized Least Mean Square

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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 primary sensor not only records the signal from the desired signal source, it also picks up a delayed and/or filtered version of noise signals originating from the noise source. The purpose of this dissertation work deals with the various aspects of design & implementation of Real-Time Noise Cancellation and its usage in different applications.

Keywords: *Signal processing, Real-Time Noise Cancellation etc*

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. One of the main objectives within adaptive signal processing is noise suppression, i.e., the detection of an information-bearing signal in noise, for example and inside aircraft cabin and automobiles, industrial noise etc.

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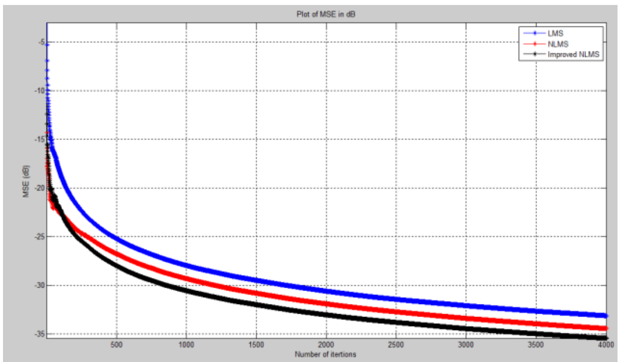
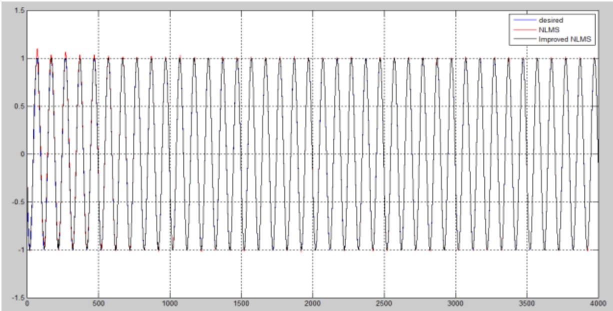
Thermal analysis can be used to determine inoculants performance, apart from the traditional usage of thermal analysis to determine the percentage of carbon equivalent liquidus, carbon and silicon levels, it can also be used to monitor metallurgical processes and identify potential problems areas such as low nodule count, under-cooled graphite and carbide/chill propensity (Udroiu, 2002), (Corneli, *et al*, 2004), (Seidu, 2008). It can be used to predict iron shrinkage tendency and help the foundry to control scrap.

Error Scale Graph



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.

3. Output signal for LMS, NLMS & Improved NLMS



Recorded signal at primary sensor = S(n) + HN(n)

(i) Let V(n) = HN(n) represent the noise signals at the primary sensor and assume that desired signal and the noise signal are uncorrelated with each other such that

E[S(n)V(n-m)] = 0 for all m

The noise signal N(n) recorded at the reference sensor is uncorrelated with the signal S(n) i.e.

E[S(n)N(n-m)] = 0 for all m

But it is correlated with the delayed and filtered version noise V(n) or HN(n) at the primary sensor output in an unknown way such that,

E[V(n)N(n-m)] = P(m) for all m

Implementation of the NLMS algorithm

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. NLMS updates the coefficient of adaptive filter by using following equation

$$w(n+1) = w(n) + \mu.\frac{e(n)(x(n))}{|x(n)|2}$$

We can also write the above equation as

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

Where $\mu(n) = \mu\frac{e(n)x(n)}{|x(n)|2}$

Now NLMS becomes the same as standard LMS except that NLMS has time varying step size which improves the convergence rate of adaptive filter.

Uncorrelated noise in primary input

A single channel adaptive noise canceller with an uncorrelated noise m_0 present in the primary input is shown in fig 3.3. The primary input thus consists of a signal and two noises m_0 and n. The figure shows a single channel adaptive noise canceller with an uncorrelated noise m_0 present in the primary input.

The desired response d is thus $s+m_0+n$. Assuming that the adaptive process has converged to the minimum mean square solution, the adaptive filter is now equivalent to a Wiener filter. The optimal unconstrained transfer function of the adaptive filter is given by

$$W^*(z) = \frac{\delta_{xd}(z)}{\delta_{xx}(z)}$$

The spectrum of the filters input δ_{xx} (z) can be expressed as$\delta_{xx}(z) = \delta_{nn}(z)|H(z)|^2$

Where $\delta_{nn}(z)$ is the power spectrum of the noise n. The cross power spectrum between filter's input and the desired response depends only on the mutually correlated primary and reference components and is given as

$\delta_{xx}(z) = \delta_{nn}(z)H(z^{-1})$(1.1)

The Wiener function is thus

$$W^*(z) = \frac{\delta_{nn}(z)H(z^{-1})}{\delta_{nn}(z)H(z)^2} = \frac{1}{H(z)} \quad (1.2)$$

Note that W*(z) is independent of the primary signal spectrum $\delta_{xx}(z)$ and the primary uncorrelated noise spectrum $\delta_{momo}(z)$.

Conclusion

In this dissertation, we have studied 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.

Future Scope

In this dissertation, we have studied the least-mean-squares adaption Algorithm, NLMS & Improved NLMS. Our scope for the future work includes:-

- The study of other adaptive algorithms and their stability for application to Adaptive Noise Cancellation compared.

- These algorithms that can include study of Recursive Least Squares with the Normalized LMS, and Variable Step-Size algorithms Symmetrical Adaptive Decorrelation Algorithm.
- Moreover, this Project does not consider the effect of finite-length filters and the casual approximation, optimization of the delay parameter to influence on the phase adjustment characteristics, and to study in detail the various other improvised implementation of the ANC in various applications.

Result and Discussion

The name MATLAB stands for Matrix Laboratory is a high performance programming language for technical computing. This software is used for a wide variety of scientific and engineering calculations, especially for automatic control and signal, image processing, it has extensive graphical capabilities, and algorithm development. Matlab allows easy matrix manipulation, plotting of functions and data, implementation of algorithms, creation of user interfaces, and interfacing with programs in other languages. Matlab is built around the Matlab language, sometimes called M-code or simply M.

In this dissertation, all the algorithms and possible implementations of the adaptive noise canceller as discussed in the previous chapters are simulated using MATLAB and results are discussed as

- Improved NLMS of adaptive noise canceller result and improvement of NLMS over simple LMS.
- Adaptive noise canceller with uncorrelated noise in primary and reference inputs result.
- Adaptive noise canceller of bias/drift removal result.
- Adaptive Noise Canceller as adaptive line enhancer result.
- Comparison of LMS, NLMS & improved NLMS error scale.
- Output signal for LMS, NLMS & Improved NLMS.

NLMS of adaptive noise canceller result and improvement of NLMS over simple LMS

The simulated models are set to the following parameters:

Filter Parameters

1) Number of epochs = 4000
2) Number of taps =60
3) Adaptation Step Size Parameter

   LMS Step Size: m = 0.01
4) Signal frequency 0.01, noise frequency 0.05
5) 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.

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