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ADAPTIVE NOISE CANCELLATION BY USING LMS METHOD

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ABSTRACT

This paper describes the concept of adaptive noise cancelling,an alternative method of estimating signals corrupted by additive noise or interference.The method uses a "primary" input containing the corrupted signal and a "reference" input containing noise correlated in some unknown way the primary noise.The reference input is adaptively filtered and subtracted from the primary input to obtain the signal estimate. A desired signal corrupted by an additive noise can often be recovered by an adaptive noise canceller using the least mean squares (LMS) algorithm. Computer simulations for all the cases are carried out using matlab software and experimental resultsare presented that illustrate the usefulness of adaptive noise cancelling technique.

KEYWORDS: Adaptive Noise Canceller, LMS Algorithm, Noise Reduction

INTODUCTION

The removal of noise from signals is an underlying problem related to several areas of research in signal processing and communications. The introduction of noise between the transmitter and receiver corrupts and distorts the input signal, thus providing an inferior signal quality on the receiving end. Processes to remove this unwanted interference are common and come in many renditions. The technique of adaptive filtering is one medium by whichsignal enhancement or noise reduction is accomplished. In a similar adaptive fashion,Systems submerged in an unknown environment can be detected with a system identification structure.

Adaptive filters are digital filters with an impulse response or transfer function that can be adjusted or changed over time to match desired system characteristics. Unlike fixed filters, which have a fixed impulse response, adaptive filters do not require complete a priori knowledge and moreover, have the capability of adaptively tracking the signal under non-stationary circumstances.

For an adaptive filter operating in a stationary environment, the error performance surface has a constant shape as well as orientation. When the adaptive filter operates in a non-stationary environment, the bottom of the error-performance surface continually moves, while the orientation and curvature of the surface may be changing. Therefore, when the inputs are non-stationary, the adaptive filters of not only seeks continuously the bottom of the error performance, but also track it.An adaptive filter is essentially digital filter with self-adjusting characteristics.

It adapts, automatically, to changes in its input signals. An adaptive filter consists of two parts: one is a digital filter with adjustable coefficients and another is an adaptive algorithm which is used to adjust or modify the coefficients of the filter.

CONCEPT OF ADAPTIVE NOISE CANCELLING

As the name implies, ANC is a technique used to remove an unwanted noise from a received signal. As shown in Figure 1, an ANC is typically a dual-input, closed-loop adaptive feedback system.

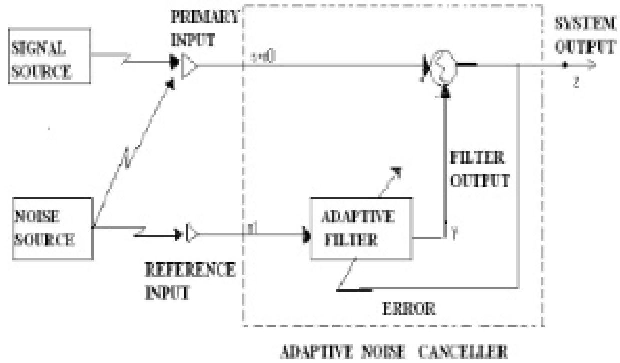

Figure 1: Adaptive Noise Cancellation

A signal *s* is transmitted over a channel to a sensor that also receives a noise no uncorrelated with the signal. Thecombined signal and noise *s +no* from the primary input the to the canceller. A second sensor receives a noise *n1* uncorrelated with the signal but correlated in some unknown way with the noise *no*. This sensor provides the reference input to the canceller. The noise *n1* is filtered to produce an output *y* that is as close a replica as possible of no. This output is subtracted from the primary input *s + no –y.*

If one known the characteristics of the channels over which the noise was transmitted to the primary and reference sensors, it would theoretically be possible to design a fixed filter capable of changing n1 into no. The filter output could then be then subtracted from the primary input, and the system output would be signal alone. Since, however, the characteristics of the transmission paths are as a rule unknown or known only approximately and are seldom of a fixed nature, the use of a fixed filter is not feasible.

Moreover, even if a fixed filter where feasible, its characteristics would have to be adjust with a precision difficult to attain, and the slightest error could result in an increase in output noise power.

In the system shown in the Figure 1 Thereference input is processed by an adaptive filteran adaptive filter differs from a fixed filter in that it automatically adjusts its own impulse response. Adjustments are accomplished through an algorithm that responds to an error single dependent, among other things, on the filter's output. Thus with the proper algorithm, the filter can operate under changing conditions and can readjust itself continuously to minimize the error signal.

Assume that s, no, n1, and y are statically stationary and have zero means. Assume that s is uncorrelated with no and n1, and suppose that n1 is correlated with no. The output z is

$$Z= s + no - y. \quad (1)$$

Squaring, one obtains

$$Z2 = s2 +(no-y)2 + 2s\ (no-y) \quad (2)$$

Taking expectations of both sides of (2), and realizing that s is uncorrelated with no and with y, yields

$$E[z2] = E\ [s2] +E\ [(n0-y)\ 2] + 2E[s\ (no-y)]$$

$$E[s2]E\ [(n0 - y)] \quad (3)$$

The signal power E[S2] will be unaffected as the filter is adjusted to minimize E [z2]. Accordingly, the minimum output power is

Mine [z2]= E [s2]+mine[(no-y02] (4)

When the filter is adjusted so thatE (z2) is minimized, E [(no –y) 2] is, therefore, also minimized. The filter output y is then a best least squares estimate of the primary noise no.Moreover, when E [(no -2)]is minimized [(z-s)2] is also minimized, since, from (1),

(z – s) = (no –y). (5)

Adjusting or adaptingthe filter to minimize the total output power is thus total amount to causing the output z to be a best least squares estimate of the signal s for the given structure and adjustability of the adaptive filter and for the given reference input. The output z will contain the signal plus noise. Since E [z2] minimizes E [(no – y) 2] minimizing the total output power maximizes the output signal-to-noise ratio.

Some of the applications of the adaptive filters are as follows:

- Acoustic noise equalization
- Adaptive speech enhancement
- Channel equalization
- Adaptive line enchanter
- Cancelling antenna side lopeinterference

Adaptive noise cancellation is used to remove background noise useful signals. This is an extremely useful technique where a signal is submerged in a very noisy environment. A typical example is inside a jet aircraft. The engine of jet aircraft can produced noise at a level over 140decibles with normal human speech is at a level of 30-40decibles, the pilot's communication is impossible in such a enviornment if there no noise cancellation equipments inside the cockpit. Usally the background noise does not keep steady and it will change from time to time. For example, the noise from the jet engine will be different at various flight states. So the noise cancellation must be an adaptive process: it should be able to work under changing conditions, and be able to adjust itself according to the changing enviornment.

Suppose a hospital is recording a heart beat (an ECG), which is being corrupted by a 50 Hz noise (the frequency coming from the power supply in many countries).

One way to remove the noise is to filter the signal with a notch filter at 50hz.However, due to slight variations in the power supply to the hospital; the exact frequency of the power supply might (hypothetically) wander between 47 Hz and 53 Hz.

A static filter would need to remove all the frequencies between 47 and 53 Hz, which could excessively degrade the quality of the ECG since the heart beat would also likely have frequency components in the rejected range.To circumvent this potential loss of information, an adaptive filter could be used.

The adaptive filter would take input both from the patient and from the power supply directly and would thus be able to track the actual frequency of the noise as it fluctuates. Such an adaptive technique generally allows for a filter with a smaller rejection range, which means, in our case, that the quality of the output signal is more accurate for medical diagnoses.

LEAST MEAN SQUARE ALGORIHTM

The LMS algorithm is a widely used algorithm for adaptive filtering. The algorithm is described by the following equations:

- Initially, set each weight W_k (i), where i= 0, 1.., N-1, to an arbitrary fixed value such as o.
- Compute filter output

$$\widehat{n_k} = \sum_{i=0}^{N-1} W_k(i) X_{k-1} \tag{6}$$

- Compute the error estimate

$$e_k = y_k - \widehat{n_k} \tag{7}$$

- Update the next filter weights

$$W_{k+1}(i) - W_k(i) + 2\mu e_k X_{k-1} \tag{8}$$

μ is the step size which determines the convergence speed of the algorithm and W is the weight of the filter.

RESULTS AND DISCUSSIONS

Simulation result based on the LMS algorithm.

Model 1 Adaptive Noise Canceller

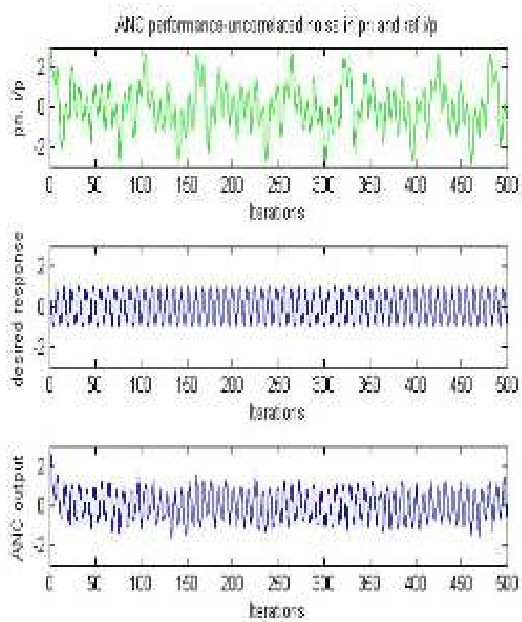

Figure 2: Adaptive Noise Canceller

It has been assumed these additive noises are uncorrelated with the other signals in the adaptive noise cancellation.

As observed from the above graph, the distortions in the output waveform get reduced to a large extent as compared to as the desired waveform.

The simulation models are set to the following parameters:

- Input signal: Sine wave
- Number of iterations = 200
- Order of filter = 18
- Adaption Step Size parameter

  LMS Step Size m = 0.004
- Cut of frequency Wn = [0.1 0.5]

  Zero mean, unit variance
- Additive Noises: White, Normally distributed, Zero mean, Unit variance.
- Additive noises: Input sensors noises are simulated as white signals.

Model 2 Addaptive Noise Canceller for Bias/Drift Removal

- Input siganl Sine wave
- Additive noise Random noise
- Number of iterations = 1024
- Order of filter = 18
- Adaption Step Size parameter

  LMS Step Size m = 0.01

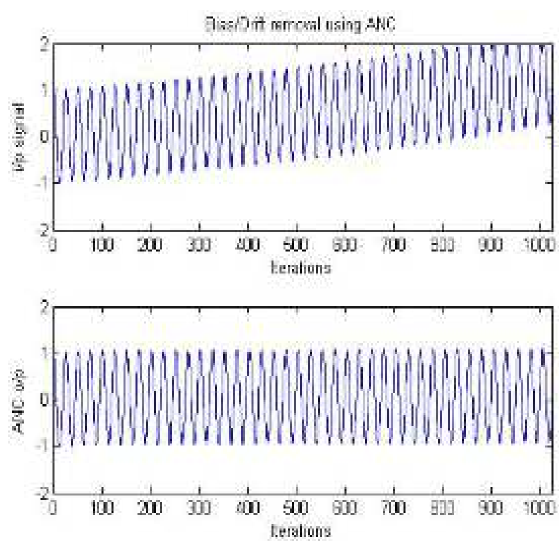

Figure 3: Bias/Drift Removal Using ANC

It is clear from the above graph the distortions in the output waveform get reductd to a large extent. The single-weight noise canceller acting as a HPF is capable removing not only a constant bias but also the slowly varying drift in the primary input.If the Bias level drift and this drift is slow enough, the bias weight adjust adaptively to track and cancel the drift using a bias weight along with the normal weights in an Addaptive Noise Cancellation can accomplish bias or drift removal simultaneously with cancellation of periodic or stochastic interference.

CONCLUSIONS

In this study, we presented an adaptive noise cancellation is an alternative way of cancelling noise in a corrupted signal presented. The principle advantage of the method is its adaptive capability, its low output noise, and its low signal distortion. The adaptive capability allows the processing of inputs whose properties are unknown and in some cases non-stationary. Output noise and signal distortion are generally lower than can be achieved with conventional optimal filter configurations. The simulation results verify the advantages of adaptive noise cancellation.

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