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A HYBRID REALIZATION OF AN ADAPTIVE FILTER FOR REAL-TIME SIGNAL PROCESSING APPLICATIONS

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Abstract

This presentation describes an efficient hybrid realization of an adaptive filter, operating in real time, and some of its applications for filtering signals corrupted by additive noise or interference. A transversal filter with 12 adaptive coefficients is used for processing a reference input signal which contains noise, correlated in some unknown way with the additive noise at the primary input. The output of the transversal filter is subtracted from the primary input signal, resulting in an error signal, which controls the filter unit sample response such that the mean square error (MSE) is minimized. A simple iterative algorithm based on the usual LMS algorithm is used for sequentially adjusting the filter coefficients.

Even if a reference source is not available it is possible, in some cases, to obtain a reference signal by sufficiently delaying the primary input signal.

The applications for which the filter performance was examined were:

(1) Cancelling periodic interference from a composite signal, when a reference source correlated with the interference is available.

(2) Filtering broadband noise using an external reference source correlated with the noise.

(3) Adaptive self-tuning filter.

(4) Cancelling periodic interference from broadband signals without an external reference source.

(5) Removing noise from speech signals when a reference source correlated with the noise is available.

Experiments show that the signal to noise ratio improvement is usually 7-22 db, depending on the application, the input signals and the filter's parameters.

Design considerations, a detailed description of the constructed filter and its performance for the above applications are presented, illustrating the usefulness and the limitations of the adaptive system constructed.

Introduction

One method of filtering signals corrupted by additive noise or interference is adaptive noise cancelling. This technique was proposed by Widrow [1] and its principle is shown in Fig. 1.

The primary input signal is composed of the desired signal $S_o(t)$ and additive noise $n_o(t)$ uncorrelated with the signal $S_o(t)$. The reference input receives a noise signal $n_1(t)$ which is correlated in some unknown way with the primary noise $n_o(t)$. The reference input $n_1(t)$ is adaptively filtered and subtracted from the primary input $(S_o(t) + n_o(t))$. The resulting error signal e(t) is used to control the adjustment of the filter coefficients (weights) such that the mean square error (MSE) is minimized. Little or no a priori knowledge of the signal and noise characteristics is necessary for the adaptation process.

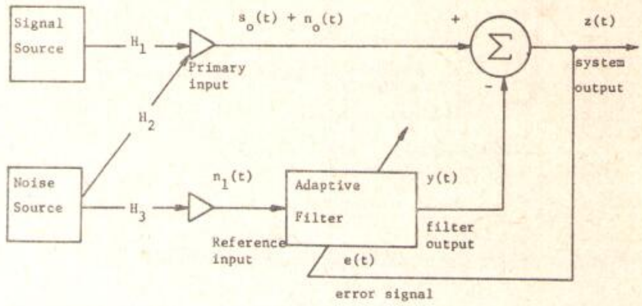

Fig. 1: The Principle of the Adaptive Noise Cancelling Technique.

With the above assumptions on the relations between $S_o(t)$, $n_o(t)$, $n_1(t)$ it can be shown [1] that the filter output y(t) is the best least squares estimate of the primary noise $n_o(t)$, and the system output z(t), which is also the error signal e(t) , is the best least squares estimate of the signal $S_o(t)$. If the channels' characteristics through which the noise signals $n_o(t)$ and $n_1(t)$ were passed to the primary and reference inputs were known, it would be possible, theoretically, to design a fixed filter capable of estimating $n_o(t)$ from $n_1(t)$. The channels' characteristics, however, are generally unknown or known only approximately, and are often variable with time, so the use of a fixed filter is not feasible. The adaptive filter can operate under nonstationary conditions and through a proper algorithm it can readjust itself continuously to minimize the mean square error.

Adaptive Algorithms

The principal component of most adaptive systems is the adaptive linear combiner with adjustable parameters shown in Fig. 2.

The conventional algorithm for adjusting the filter coefficients is a variation of the steepest descent process using a gradient estimate instead of the true gradient. The new weight vector $\underline{W}(j+1)$ is obtained from the previous weight vector $\underline{W}(j)$ by a change proportional to the negative gradient, as given by (1).

$$\underline{W}(j+1) = \underline{W}(j) - \mu\cdot\hat{\underline{\nabla}}(j) \quad ; \mu>0 \quad (1)$$

lay line receives the reference input x(j) in analog form, without A/D conversion. The transversal filter outputs x(j-1),..., x(j-12) are analog delayed samples of the reference input x(j) and are selected sequentially by an analog multiplexer/selector to provide the analog input to the multiplying D/A converter. The multiplication of the coefficients with the signal samples in the delay line is thus performed by a single multiplying D/A converter which operates in a time division multiplexed mode. The filter output y(j) is the weighted sum of the transversal filter outputs and is given by (4).

$$y(j) = \sum_{i=1}^{12} w_i(j) \cdot x(j-i) \qquad (4)$$

This sum is simply performed by an integrator and a sample & hold circuit which are controlled from the main control unit. The error signal e(j) is obtained as the output of a differential amplifier, the inputs of which are the primary input of the system d(j) and the adaptive filter output y(j).

This error signal e(j) controls in a simple way, according to the SIGN algorithm(3), the adjustment of the filter coefficients. The coefficients are stored in digital shift registers and are updated at each iteration step by a fixed amount of $+2\mu$ or -2μ, according to the sign of the gradient estimate. Instead of evaluating the gradient estimate, as in the LMS algorithm (2), only the sign of the gradient estimate is evaluated. This is done simply by two comparators and one XOR gate. The updating of the filter coefficients is performed by a binary adder.

Each iteration step consists of two phases. During the first phase the filter output y(j) and the error sample e(j) are evaluated and during the second phase the filter coefficients are readjusted. Let us see what happens during one iteration step.

Phase 1: The integrator is first reset, then the analog multiplexer sequentially selects the transversal filter outputs x(j-i), (i = 1,...,12), and synchronously the digital coefficients $w_i(j)$ appear at the digital input of the multiplying D/A converter. At the end of the first phase the S & H circuit samples the output of the integrator which gives the filter output y(j). The value of y(j) is held fixed until the next period.

Phase 2: The analog multiplexer, again, sequentially selects the same transversal filter outputs x(j-i), (i = 1,...,12), this time for the adaptive algorithm used for adjusting the filter coefficients. Synchroniously, the digital coefficients $w_i(j)$ which appear at the binary adder input are updated by $+2\mu$ or -2μ and are stored back in the shift registers. At the end of phase 2 the analog signal samples are shifted by one delay unit and the system returns to phase 1.

A more detailed description of the constructed adaptive filter can be found in [4].

Applications

This section describes some noise cancelling applications of the constructed adaptive filter. Experimental results are presented which demonstrate the performance of the adaptive filter in these applications. A more detailed study of the effects of the different parameters on the constructed filter performance is reported in [4]. Other possible applications, including adaptive equalization, can be found in [5] - [7], as well as in [4].

1. Cancelling periodic interference using an external reference source.

A major problem in many precise electronic measurements of low-level signals obtained from various gauges is the appearance of unwanted 50Hz power-line interference in the output. Some examples for it could be found in electronic instrumentation, electro-medical equipment, electrocardiography, etc. One method of reducing this kind of interference is adaptive noise cancelling, using an external reference souce taken directly from a wall outlet. Other types of periodic interference can be also reduced by adaptive noise cancelling, if the source of interference is available for use as the reference input. The primary input consists of the desired signal and the interference.

This method of reducing periodic interference is more advantageous than conventional notch filtering, because it results in less distortion and there is no need for exact adjustment to the interference frequency.

Fig. 5 shows typical steady-state input and output signals, demonstrating the usefullness of the system for cancelling periodic interference.

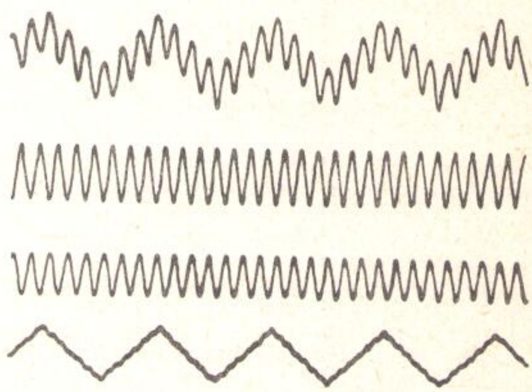

Fig. 5: Cancelling periodic interference

The primary input signal (a) consists of an 85 Hz triangular signal and additive 550 Hz sinusoidal interference at 0db signal to noise ratio. The reference input signal (b) is a sinusoidal signal different in amplitude and phase from the additive interference. The filter output (c) is an estimate of the interference and is subtracted from the primary input, yielding the system output (d) which is an estimate of the triangular signal. Typical signal to noise ratio improvement is about 22db when the signal and the interference frequency is between 20 Hz and 2.5 KHz. The maximum signal or interference frequency is about 1/8 of the sampling frequency. The best results were obtained when using a small step-size, i.e. $\mu = 2^{-12}$.

2. Cancelling broadband noise using an external reference source.

If the desired signal $S_o(t)$ and the noise $n_o(t)$ occupy the same frequency spectrum, the use of conventional filters is not feasible. However, the adaptive noise cancelling technique can be applied to reduce the broadband noise at the output, if a reference signal-free source correlated with the noise $n_o(t)$ is available

Typical results are similar to those obtained in the previous application.

5. Cancelling noise in speech signals.

One practical application of real-time adaptive filtering is cancelling acoustic interference from speech signals. Acoustic interference is commonly picked up by the microphone into which a person speaks, thus severely interfering with the intelligibility of the speech. By placing a second microphone at a suitable location in the acoustic noise field, a reference signal correlated with the interference and free of the speech signal could be obtained. The primary input signal is the speech corrupted by the interference. If the reference signal is adaptively filtered and subtracted from the primary input signal, significant reduction of the acoustic interference at the system output can be achieved. A set of experiments were performed with the constructed adaptive filter, demonstrating the feasibility of cancelling noise in speech signals. When the acoustic interference was a 1 kc band-limited pseudo-random noise, the adaptive filter was able to reduce the interference by about 10 db, which otherwise made the speech barely intelligible. The best results were obtained when the step-size μ was 2^{-7} allowing fast adaptation.

Conclusion

The constructed adaptive filter combines analog and digital circuits. This allows simple realization and medium speed of operation. The adjustment of the filter coefficients is controlled by a simple iterative algorithm which is easily implemented in hardware. Some loss in performance is caused by the inaccuracies of the analog circuits which may have a cumulative effect. In its current form the adaptive filter has the capability of processing signals up to 5 Kc. When a suitable reference input is available, the adaptive filter can be used in various configurations for real-time noise cancelling applications. The experimental data presented in the paper demonstrate the ability of the adaptive filter to reduce additive periodic interference or broad-band noise.

References

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