HYBRID REALISATION OF AN ADAPTIVE FILTER FOR REAL-TIME NOISE-CANCELLING APPLICATIONS

Indexing terms: Adaptive filters, Signal processing

An efficient hybrid realisation of an adaptive filter for real-time noise-cancelling applications is described. The filter is based on an analogue tapped delay line but its coefficients are digital and are updated according to a simplified l.m.s. algorithm. A single multiplying d.a. device is used for performing the multiplications. The structure of the filter, its performance and processed examples are presented.

Introduction: An attractive method of filtering signals corrupted by additive noise or interference is the adaptive noise-cancelling technique proposed by Widrow.[1] This technique can be applied when a suitable signal-free noise source is available as a reference input. The principle of its operation is shown in Fig. 1.

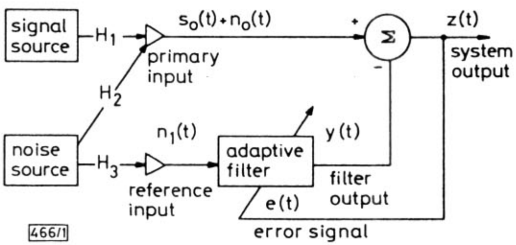

Fig. 1 *Principle of the adaptive noise cancelling technique*

If the reference input contains noise correlated in some unknown way with the additive noise at the primary input, but not with the desired signal, minimisation of the mean square error means that the filter output is an estimate of the primary input noise. Following the subtraction this results in an estimate of the desired signal at the residual error output.

Several implementations of this type of filter have been previously proposed.[2–4] However, the filter described here is especially attractive because of its simplicity and its relatively good performance in noise-cancelling applications. This is achieved by means of a simplified adaptation process and by using combined analogue and digital circuits, avoiding the need for fast digital multipliers. The filter can be easily applied to other signal-processing or data-communication applications, such as system modelling, adaptive prediction, adaptive equalisation etc.[5]

Implementation: The conventional l.m.s. algorithm proposed by Widrow[1] is a variation of the steepest-descent adaptation method using the gradient estimate instead of the true gradient, and is given by the recursive relation:

$$\boldsymbol{W}(j+1) = \boldsymbol{W}(j) + 2\mu e(j)\boldsymbol{X}(j)$$

$\boldsymbol{X}(j)$ is the input vector, $\boldsymbol{W}(j)$ is the weight vector, $e(j)$ is the residual error and the parameter μ is the step-size factor which determines the rate of convergence and the stability of the adaptation process.

A great simplification in the algorithm implementation is obtained by modifying this l.m.s. algorithm as has been done in Reference 6. With this modification the coefficients are adjusted by a fixed amount in the negative direction of the m.s.e. gradient estimate, as given by:

$$\boldsymbol{W}(j+1) = \boldsymbol{W}(j) + 2\mu \operatorname{sign}[e(j)] \cdot \operatorname{sign}[\boldsymbol{X}(j)]$$

Results from simulations on a digital computer show only little degradation of the system performance when using this sign algorithm instead of the l.m.s. algorithm in noise-cancelling applications. The convergence of the mean square error is even faster when using the sign algorithm instead of the l.m.s. algorithm, as can be seen from Fig. 2.

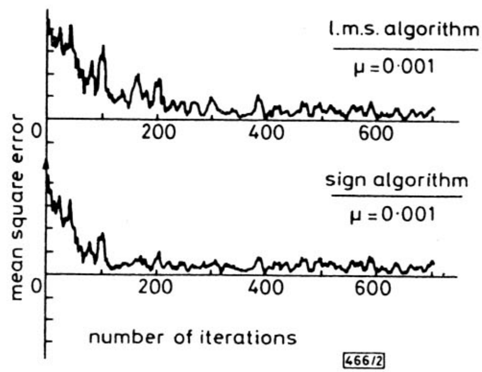

Fig. 2 *Learning curves when using the l.m.s. algorithm and the sign algorithm*

A simplified block diagram of the constructed adaptive filter which uses the sign algorithm is shown in Fig. 3. An analogue clock-controlled tapped delay line with 12 taps, built in a single i.c., is used for implementing the transversal filter. The filter coefficients, however, are in binary form and are adjusted digitally to obtain sufficient accuracy. The multiplication of the coefficients with the signal samples in the delay line is performed by a single multiplying d.a. convertor which operates in a time-division multiplexed mode, thus greatly reducing the amount of hardware required. The weighted sum of the tapped delay line outputs is simply performed by an integrator and a sample and hold circuit, which are controlled from the main control unit. The filter 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 at the output of the XOR gate. The coefficients are represented as 12 bit binary words, allowing sufficient accuracy and a step size as small as 2^{-12}. Each iteration step consists of two phases. During the first phase the filter output $y(j)$ and the error sample $e(j)$ are computed and during the second phase the filter coefficients are readjusted.

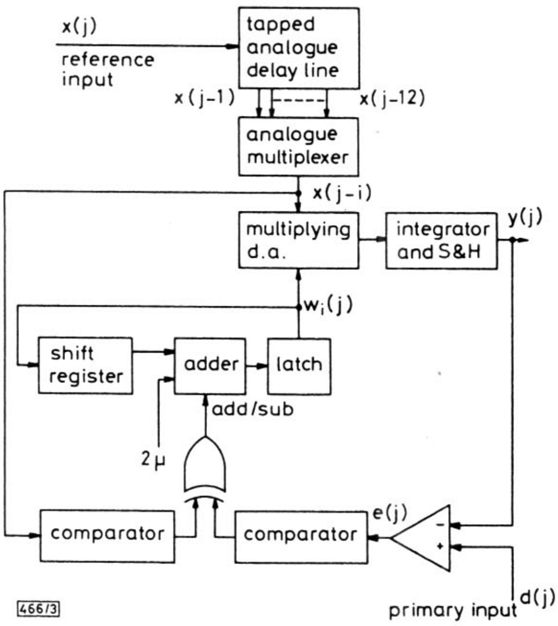

Fig. 3 *Simplified block diagram of the adaptive filter*

In its present form, the maximum sampling frequency is limited to 40 kHz. A more detailed description of the constructed adaptive filter can be found in Reference 5.

Results: This section describes some of the experimental results, demonstrating the performance of the adaptive filter in noise-cancelling applications. Two typical applications of the filter are cancelling periodic interference and cancelling broadband noise. Fig. 4 shows typical steady-state input and output signals, illustrating the usefulness of the system for cancelling periodic interference.

The primary input a consists of a triangular signal and additive sinusoidal interference. The reference input 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 residual error output d which is an estimate of the triangular signal. Typical signal/noise ratio improvement is about 23 dB when the signal and interference frequencies are between 20 Hz and 5 kHz, i.e. up to 1/8 of the sampling frequency. The best results for stationary signals are obtained if the smallest step-size factor is used, i.e. $\mu = 2^{-12}$. Fig. 5 shows typical input and output signals, after the adaptation has been completed, when the adaptive filter is used for cancelling broadband noise.

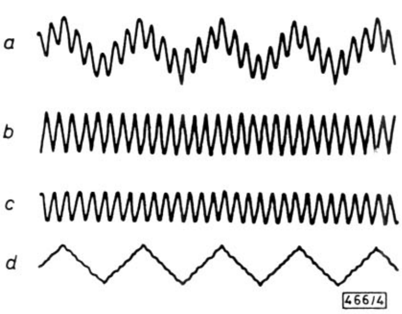

Fig. 4 *Cancelling periodic interference*

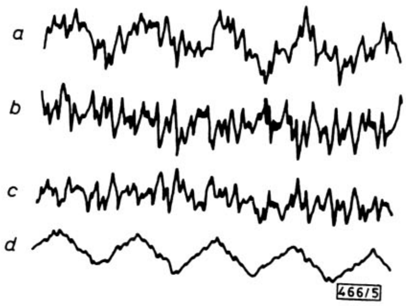

Fig. 5 *Cancelling broadband noise*

The primary input a consists of a triangular wave and additive band-limited noise (from a pseudorandom noise generator). The reference input b is taken directly from the noise source, while the noise at the primary input is a delayed and attenuated version of the reference input. The filter output c is an estimate of the additive primary noise and is subtracted from the primary input, yielding the residual error output d that is an estimate of the triangular signal. Typical signal/noise ratio improvement is about 12 dB when the input noise bandwidth is 1 kHz and about 15 dB when the noise is limited to below 500 Hz.

Conclusions: The constructed adaptive filter combines analogue and digital circuits. This results in a simple realisation 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 analogue circuits which may have a cumulative effect. The particular adaptive filter constructed has the capability of processing signals up to 5 kHz. However, with a small amount of additional hardware and utilising a comercially available tapped delay line i.c., an adaptive filter with 32 coefficients can be constructed, with a respective improvement in performance.

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