CCD BASED ANALOGUE ADAPTIVE PROCESSING

C.F.N. COWAN*, J.W. Arthur†, J. Mavor*, and P.B. Denyer†

ABSTRACT

The paper presents an implementation of an adaptive filter based on the Widrow least-mean-square adaptation algorithm using a monolithic 64-point CCD programmable transversal filter as its central processing element. Significant results are presented for the prototype system operated as a noise canceller, a self-tuning filter and as an adaptive line equaliser. The technique described has potential for full monolithic integration having up to 256 filtering stages on a single integrated circuit and the results show the possible high performance characteristics of this implementation.

INTRODUCTION

Many signal processing tasks are difficult or impossible using conventional filtering techniques due to the unknown nature of signal or transmission characteristics. One example of this is the reliable detection of digital data transmitted at high bit rates over the switched telephone network. Using a dedicated transmission line a fixed equaliser may be used to compensate for known phase and amplitude distortions. However, on a switched network these distortion characteristics will be different for each connection made. It is therefore necessary to use some sort of learning machine which is capable of adaptively learning the transmission line characteristics by examining the statistics of the line output signal.

Such systems[1], known as adaptive filters, have been extensively studied in theory and many of their operational characteristics modelled[2] on computer. However, until the inception of charge-coupled devices[2] and later the monolithic progammable transversal filter[3](PTF) practical realisation of these filters has been limited to complex digital systems usually of low bandwidth and incurring heavy penalties in cost, size and power consumption. It is the purpose of this paper to present an alternative adaptive filter implementation based on a monolithic, 64-point CCD programmable transversal filter. The unit to be described[4] could, potentially, be integrated monolithically and offers a high bandwidth, compact, low-power, high-performance solution to the problem of adaptive filter implementation with potential in such diverse areas as adaptive line equalisation, speech processing and noise cancellation in medical electronics.

SYSTEM IMPLEMENTATION

In general the task of an adaptive filter is to manipulate an input signal vector $\underline{S}$ (t) in such a way as to approximate, as closely as possible, a given output d(t). This is done by operating on an error signal e(t) which is the difference between the filter output c(t) and the desired output d(t). In the system described here e(t) is minimised

* Dept. of Electrical Engineering, University of Edinburgh, Scotland.
† Wolfson Microelectronics Institute, University of Edinburgh, Scotland.

in the least-mean-square sense according to the Widrow LMS algorithm.[1]

$$\underline{H}(t+1) \;=\; \underline{H}(t) \;+\; 2\mu\,e(t)\,\underline{S}(t)$$

where $\underline{H}(t)$ is the vector of filter weight values at time t. Literally this means that each weight value is updated once for every output sample period. However, with the PTF[5] used in our prototype system this would cause a severe degradation in signal bandwidth. Therefore only one weight is updated at each output sample period. A block diagram of the proto-type system[4] is shown in Fig. 1, the error is sampled once in every 65 output sample periods and is then multiplied by the output of the last tap in the CCD tapped delay line and attenuated to yield the product $2\mu\,e(t)\,\underline{S}(t)$. This value is then added to the previous value of the filter tap weight h(t) to yield the updated weight value h(t+1). An external storage system was used in this case because weight values once stored in the reference section of the prototype PTF could not be read back, as accessing of the weight storage cell automatically destroys the stored weight value. In the prototype described here the external storage used was implemented using digitial circuitry since discrete analogue storage cells proved to be excessively leaky, resulting in severe biasing of the converged pulse response. This would not be the case for fully integrated analogue storage where decay values in the region of 0.1V/sec are achievable. The external digital store does therefore represent a realistic model of a fully integrated analogue adaptive filter.

Three basic system configurations were used to evaluate the performance characteristics of the prototype system and these are illustrated in Fig. 2.

(1) Basic noise canceller: this system is shown in Fig. 2(b). In this case it is desired to subtract or cancel an unwanted noise or interference from a required signal. Therefore the incoming signal, contaminated by some interfering signal, it supplied to the d(t) input of the adaptive filter. A correlated, but not identical, version of the interfering signal is then applied to the filter input s(t). After convergence the filter output should be an accurate representation of the interfering signal source tracking in phase and amplitude the interference on the d(t) signal so that it subtracts coherently from d(t) leaving the desired signal at the e(t) output.

(2) Self-tuning filter: here the system is supplied with only one input (Fig. 2(d)) which will consist of a mixed narrowband periodic signal and a broadband non-periodic signal. This input is applied directly to the d(t) input and a delayed version of the signal is applied to the s(t) input. The delay T is sufficient to decorrelate the broadband, non-periodic signal component between the s(t) and d(t) inputs. However, the narrowband periodic component will still correlate and will therefore be reproduced at the filter input in phase with the periodic component of d(t). Therefore the filter output should contain only the narrowband, periodic signal component and the error output will contain only the broadband non-periodic component of the input signal. This scheme may be used to reduce unwanted hum on speech signals or to reduce unwanted noise on periodic signals.

(3) Inverse filter (adaptive equaliser): here the objective is to reproduce an accurate replica of a signal which has been transmitted through a distorting medium such as a telephone line. In order to do this the filter frequency response must be the inverse of the transmission medium response. One way of achieving this is to transmit a known

data sequence through the transmission medium and apply the output of
the transmission medium to the filter input s(t) while the known data
sequence is used as a training signal d(t). After initial training the
filter response is frozen and the filter is thereafter used as a straight-
forward transversal equaliser. This system has potential for application
in systems such as adaptive modems for fast digital transmission on long
telephone lines.

RESULTS

The results presented in this section are for the prototype 64-stage
analogue/digital hybrid system operated at a 16kHz sampling rate. The
maximum sampling frequency for the prototype system was 50kHz (which
could be extended to 100kHz with circuit modification). This bandwidth
limitation is the maximum clocking rate of the prototype 64-point
programmable filter device. The system was constructed on a double
Eurocard-sized circuit board consuming less than 10 watts of power.

Fig. 3 shows a typical result for the system operated in the basic
noise cancellation mode (Fig. 2(b)). Here the incoming signal (Fig.3(b))
was a sinusoid with a strong interfering sinusoid at twice the desired
frequency. The s(t) input (Fig.3(a)) was a version of this interfering
sinusoid which is incorrect in phase and amplitude for coherent sub-
traction to take place. Fig. 3(c) shows the filter output which is the
interfering signal corrected in phase and amplitude to match the
corresponding component in Fig. 3(b). Fig. 3(d) shows the error or
canceller output which is the desired sinusoid without the strong 1st
harmonic interference. Fig. 3(e) and (f) show frequency spectra of the
signal before and after cancellation showing that the unwanted signal
component has been attenuated by about 50 dB. This yields a general
figure of merit for the system, x, known as the adaptivity which is
defined as:

$$ x = 20 \log_{10} \frac{e(t)}{d'(t)} $$

where $d'(t)$ is the component of d(t) correlated with s(t).

Fig. 4 shows a result obtained for the filter operated in the
self-tuning filter mode where the input (Fig. 4(d)) is a sinusoid
contaminated by broadband noise. The filter output (Fig.4(b)) is the
sinusoid with the broadband component considerably attenuated (spectral
analysis shows average attenuation of the noise to be about 25 dB) and
the error or canceller output (Fig. 4(c)) is the broadband signal
component without the sinusoidal component. In this case the weight
vector is the expected matching sinusoid.

Fig. 5 shows results obtained from the system operated as an
inverse filter or adaptive equaliser. The input signal s(t) is the
output from a telephone line simulator (equivalent to about 5 miles of
telephone line) which has as its input a PN sequence of length 16.
The training signal d(t) is a delayed version of the telephone line
input and Fig. 5(c) shows the actual equaliser output demonstrating the
phase and amplitude correction applied by the adaptive filter, Fig. 5(d)
shows the eye pattern of the telephone line simulator output,
demonstrating almost total closure of the eye whilst Fig. 5(e) is the eye
pattern measured after equalisation. Here a good eye opening is clear
and no jitter is evident in the waveform zero crossings. This result
demonstrates the feasibility of the CCD adaptive filter for use in
adaptive modems.

CONCLUSIONS

It has been shown that a CCD-based, programmable transversal filter may be used effectively in an adaptive filtering context in a number of application areas. One area of particular interest is that of adaptive equalisation of telephone lines for high speed digital data transmission and it has been shown that our prototype system yields considerable improvement in signal detectability in this instance. Future development will include work on digital decision-directed feedback systems aimed at high-speed modem applications.

The ultimate aim of the project is the production of a monolithic fully integrated version of the CCD adaptive filter and feasibility studies are presently under way, leading to the design of a circuit having 256 filter stages on a single chip with a bandwidth in excess of 100kHz. Such a system will clearly be a very attractive alternative to digital adaptive filters using either hard-wired logic or microprocessor-based systems.

ACKNOWLEDGEMENTS

REFERENCES

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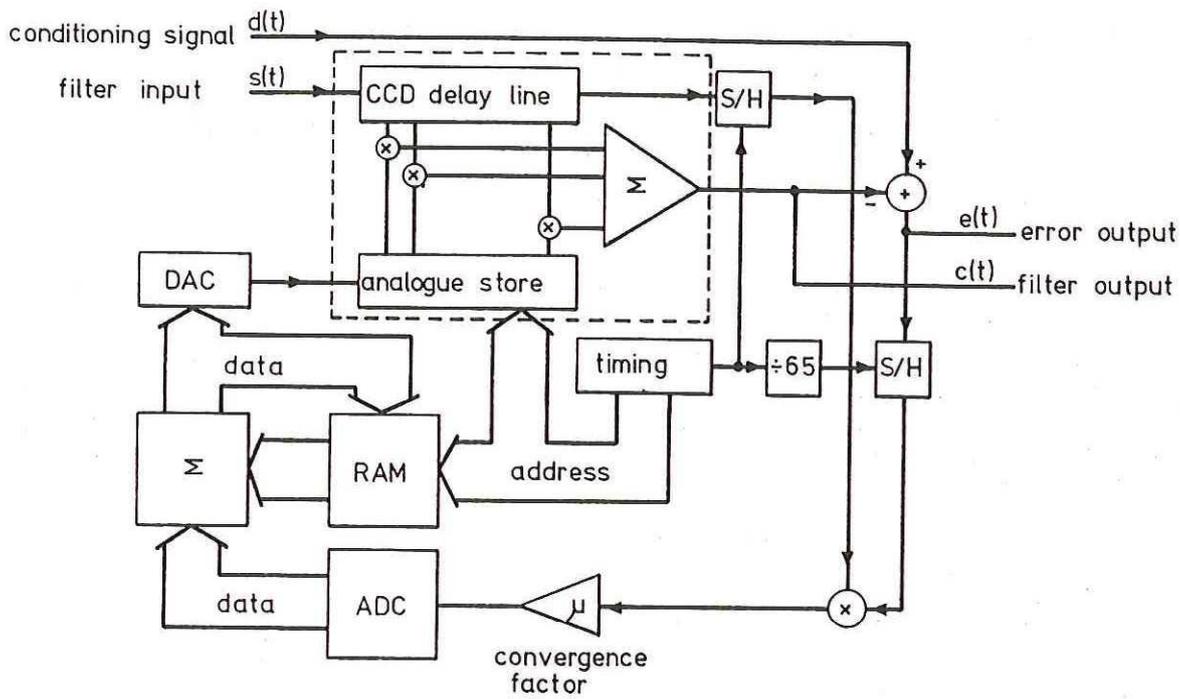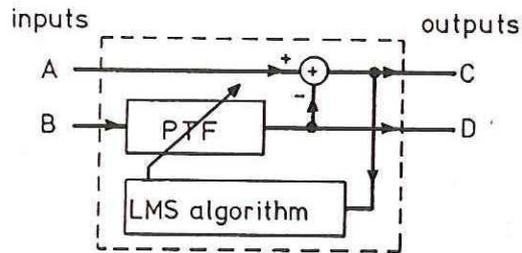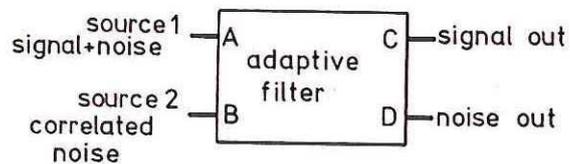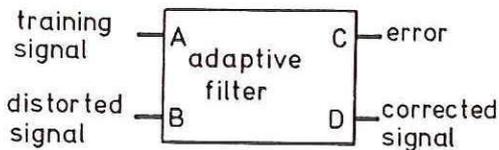
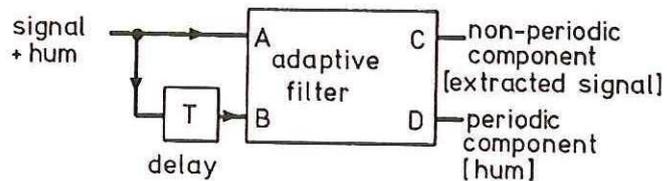Fig. 1 : Block diagram of the experimental system.

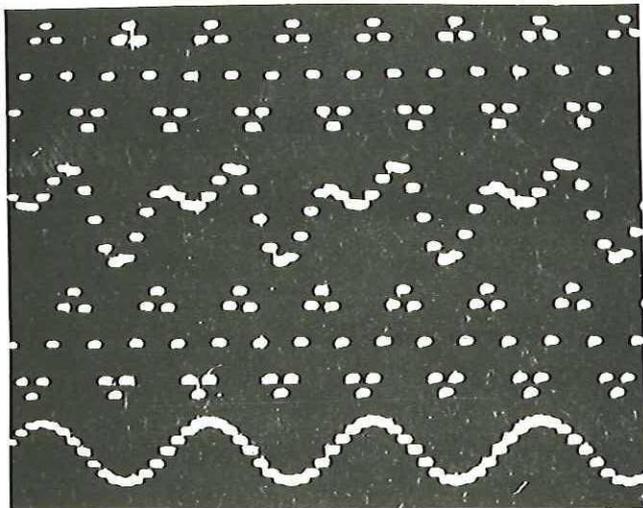(a) basic adaptive filter configuration
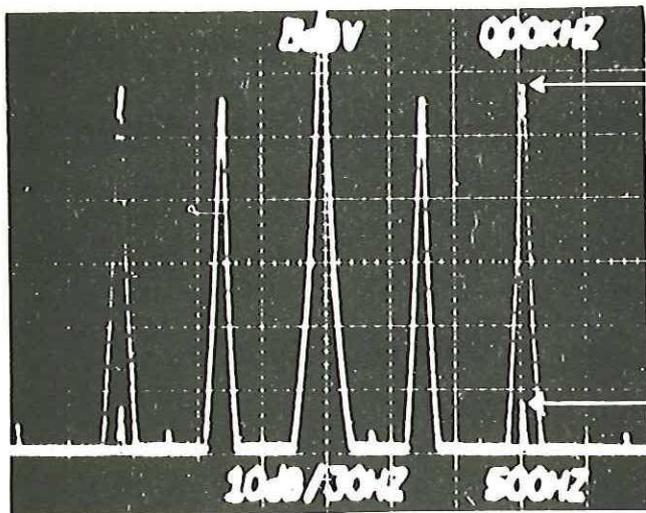
(b) noise canceller

(c) inverse filter

(d) self-tuning filter

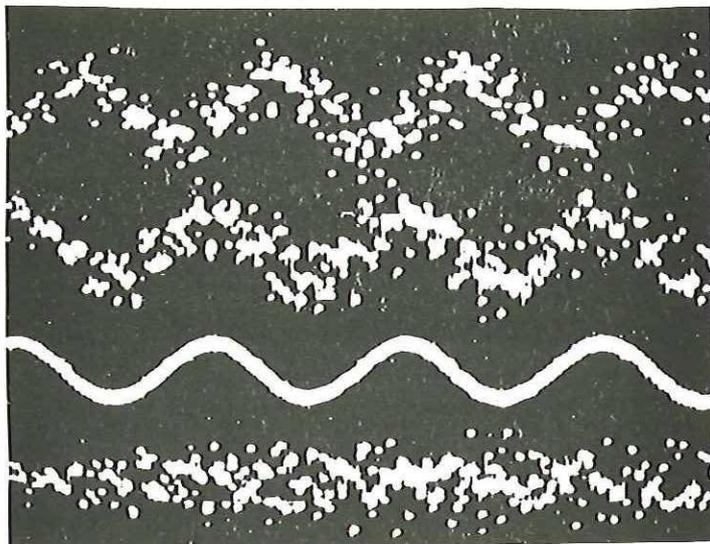Fig. 2 : Basic operational modes of the adaptive filter.

A. Signal input : distorting waveform

B. D(t) input : required signal with interfering signal (A) added

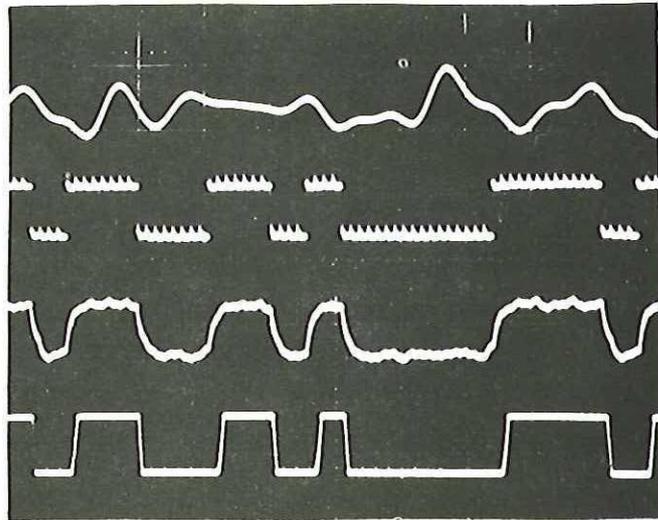C. Filter output : interfering signal corrected in phase and amplitude

D. Canceller output

E.

Spectra of D(t) and the canceller output

F.

Fig. 3 : Operation in the noise canceller mode.

A. Signal input

B. D(t) input

C. Filter output

D. Error or canceller output

Fig. 4 : Operation in the self-tuning filter mode.
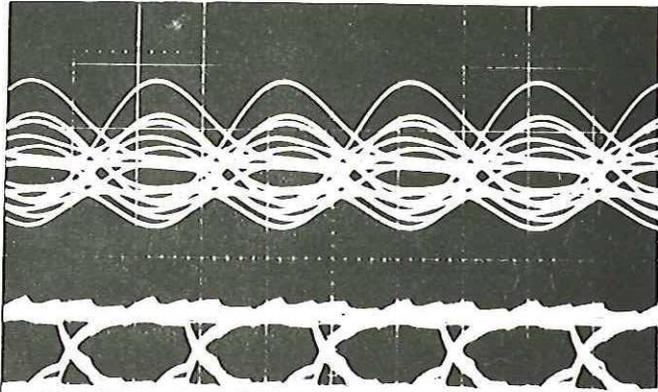
A. Simulated line output

B. Training signal

C. Filter output

D. Threshold detect output

E. Line output eye pattern

F. Equaliser output eye pattern

Fig. 5 : Adaptive equalisation of a baseband digital signal at the output of a simulated telephone line.