A MICROCOMPUTER-CONTROLLED ADAPTIVE CCD TRANSVERSAL FILTER

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ABSTRACT

Adaptive filters for such tasks as swept bandpass filtering, variable matched filtering, multi-channel filtering and so forth are generally difficult and expensive to construct. A simple transversal filter structure employing a microcomputer for generation and adaption of the filter weighting coefficients is described here. The microcomputer receives requests for adaption either by means of operator commands or from transducers monitoring the signal environment, computes the required FIR (Finite-duration Impulse Response) coefficients and outputs the coefficients' digital representation to a data bus. The coefficient information is then routed to independently addressable data buffers for weighting, by means of an array of multiplying Digital-to-Analogue Converters, of the signal samples coming from a tapped CCD.

Feasibility of the system structure has been established by an operating prototype system which utilizes eleven filter taps under control of a four-bit microcomputer which is, consequently, of moderate price and performance; expansion now underway to a thirty-two tap filter driven by a sixteen-bit microcomputer will provide considerably enhanced speed and filtering effectiveness. Measured performance of the prototype filter operating in a switching mode between several minimax FIR realizations is presented.

INTRODUCTION

There are many filter application areas where variable filtering characteristics can be of great benefit. Simple tasks such as swept bandpass filtering, tracking notch filters, and so forth have been successfully achieved by a variety of means. More complicated jobs like adaptive matched filtering and frequency-agile jamming avoidance require some sort of supplementary intelligence (an operator or a computer) to guide the adaption process. Even when proper guidance is available the problem of physically implementing changes to filter parameters is generally difficult and expensive to accomplish.

The hybrid (time-discrete/amplitude-continuous) nature of CCD (Charge Coupled Device) filters is particularly attractive for adaptive filtering, since the filtering action can be so easily manipulated by varying the values of feedback and feedforward weighting coefficients in the filter structure. In this paper we describe a transversal (non-recursive) CCD filter which employs a microcomputer to supply the filter coefficients and to update, or totally modify, the filter behaviour as conditions demand.

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The potential of such an "intelligent filter", comprising both the filtering subsection and the computing subsection, is of course enormous. The computer could scan transducers monitoring the signalling environment, deduce the currently desired filter characteristic, design the appropriate tap weights, and then re-configure the old filter to the new desired filter by transferring the new coefficients in to the filtering subsection. If the computer and the CCD were fast, powerful, cheap and small, then a very versatile and compact adaptive filtering package could be produced. Fixed filters (which constitute the vast majority of filters) are clearly just degenerate adaptive filters, so their function would be comfortably covered (though not necessarily cost-effectively) by such a unit.

Conceptually the intelligent adaptive transversal filter we are considering appears as shown in Figure 1. The filtering action takes place at a "high" rate (hundreds of kilohertz upwards with current CCD technology) while relatively slow-speed adapting modifications are enacted through the computer. Since a microcomputer would seem to offer the best hope for physically small and cheap filter packages, the adaptation speed would, today, be many milliseconds or perhaps many seconds, depending upon the algorithm complexity demanded in computing tasks; nevertheless, a large number of present-day applications would find this adequate and the expectation of ever-increasing effectiveness in both CCD and microprocessor technology would suggest that the structure holds promise for an even broader range of future applications.

SYSTEM HARDWARE

A variable transversal filter is in essence no more than a tapped delay line (with the delays being clock-controllable in the case of CCD's) with adjustable tap weightings (which can be visualized as, say, variable potentiometers) feeding a summing amplifier. Ideally the filter's transfer function is determined solely by the values of the tap weightings, which are the coefficients defining the Finite-duration Impulse Response (FIR) sequence. Abundant coverage of the topic of design of FIR coefficients is contained in (ref. 1) and elsewhere in the digital signal processing literature.

The prototype filter we have constructed employs eleven of the 32 available taps of a Bell Northern CCTF02 CCD. Tap weighting in our system is done by means of eleven multiplying digital-to-analogue converters (DAC's) which each receive the analogue CCD tap voltage as one input and an eight-bit digital coefficient as the other input.

Figure 2 shows details of the filter, including decoding arrangements for two 4-bit coefficient words passed in from the Intel MCS-4 microcomputer employed in the prototype system. The CMOS multiplying DAC's are comparatively cheap, have DTL/TTL/CMOS compatible digital inputs, and have an analogue impedance (typically 10 kΩ) suitable for direct connection to the CCD tap outputs. It was decided that costs could be reduced by serving groups of four (with one group of three) DAC's by a pair of LM 318N operational amplifiers - rather than by having two amplifiers per DAC for performing four quadrant multiplication. A single operational amplifier then completes the overall summing operation.

Each DAC is buffered by two quad-SR flipflops which are separately addressable from the microprocessor. The microprocessor outputs an address in 2 x 4-bit bytes, followed by 4 bits of data, and, finally, a peripheral write pulse. A total of 22 separate addresses are required, and so address decoding is achieved by two 4-line to 16-line decoders, these being simultaneously strobed by the write pulse.

Since the microprocessor works using 2's complement arithmetic, it is convenient for the data output to be in 2's complement form. The multiplying DAC's require "offset binary" data coding (modified 2's complement). This may be achieved very simply by inversion of the most significant data bit.

SYSTEM SOFTWARE

Microcomputer software was produced to enable operator loading of any 11-coefficient filter impulse response or selection from among four different sets of pre-specified filter coefficients. The four "standard" filters

Lowpass
Bandpass
Differentiator
Hilbert filter (allpass 90° phase-shifter)

were designed on an off-line computer as linear-phase minimax filters, using the Remez Exchange Algorithm (ref. 1). The binary representations of the coefficient sets are stored in eighty-eight adjacent four-bit RAM locations for loading as DAC weighting inputs upon operator command.

Rapid (i.e. pushbutton speed) reconfiguration of the transversal filter is easily effected. It was desired to demonstrate some further degree of variability in this initial prototype, without incurring excessive programming difficulty due to the very restricted software features of the four-bit microcomputer. To this end, subroutines were generated which:

Swap any two coefficients
Rotate all eleven coefficients (single-step)
Rotate all eleven coefficients (continuously).

These three features, though not particularly useful in practical applications, do nonetheless provide effective demonstration that filter variation can be carried out at a rapid rate. For example, the continuous coefficient rotation feature has the effect of periodically repeating filtering at rates controllable to periods as short as about 1 millisecond. It can be shown that eleven different filter characteristics occur per period - their amplitude and phase curves being different at all frequencies, except those corresponding to DFT (Discrete Fourier Transform) sample frequencies. Test trials on audio test signals abundantly exhibit the variable nature of the filtering in progress.

SYSTEM PERFORMANCE

Theoretical and measured magnitude transfer functions for the four selectable filters are shown in Figures 3-6. The sampling frequency was chosen at 40 kHz for ease of measurement.

It was found that considerable departure from the theoretical gain curves was obtained unless special care was taken in grouping the AD7520 DAC's with closely matched multiplier gains. The results depicted in Figures 3-6 were achieved with two groups of four and one group of three DAC's, which had gain variations not exceeding 12% within each group; various arrangements of mis-matched DAC's gave rise to transfer functions displaced 6dB or more from the theoretical curves.

The prototype filter system was tested and found to perform satisfactorily with sampling rates ranging between 500 Hz and 500 kHz. The high frequency limitation is considerably less than the 5 MHz bandwidth specification of the CCTF02 CCD alone - a fact which derives from circuit design criteria relating to minimum DAC settling time.

The dynamic range of the CCTF02 is specified as 60 dB, with a maximum input signal swing of 750 mV peak-to-peak (265 mV r.m.s.). Tests on the operating filter indicated a very high level of digital data noise. The r.m.s. noise measured in the absence of input signal was 5.9 mV, while the output for a 265 mV r.m.s. sinusoidal signal was 120 mV, giving a system dynamic range of only 26 dB. It is believed that significant improvement in the filter dynamic range can be effected by careful circuitboard layout of components in future system versions.

FUTURE WORK

The prototype intelligent filter was limited in two main respects: a short CCD length and a small microcomputer subsystem. These factors limited the precision of filtering attainable and the ease (and speed) of software implementation, respectively. The prototype has adequately established the feasibility of the system structure and efforts are now underway to expand the system to incorporate 32 CCD taps, under control of a 16-bit microcomputer. One of the prime potential applications for this system would appear to be in the field of adaptive equalization of data channels, which we intend to explore.

It is recognized that one of the principal problems with a more advanced system will remain that of fast calculation of good quality filter coefficients under conditions of rapid filter variation. Optimization techniques which have reached a high degree of sophistication usually require large-scale computers and are not likely to be usable in a small microcomputer. Considerable effort is currently being devoted to investigation of suboptimal design techniques which yield reasonably efficient filter designs with algorithms especially suitable for microcomputer use.

ACKNOWLEDGEMENTS

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(2) Cain, G.D. and A.H. Abed, "Mildly suboptimal digital filters using a host windowing approach", *Electronics Letters,* vol.11, no. 20, 2nd October 1975.

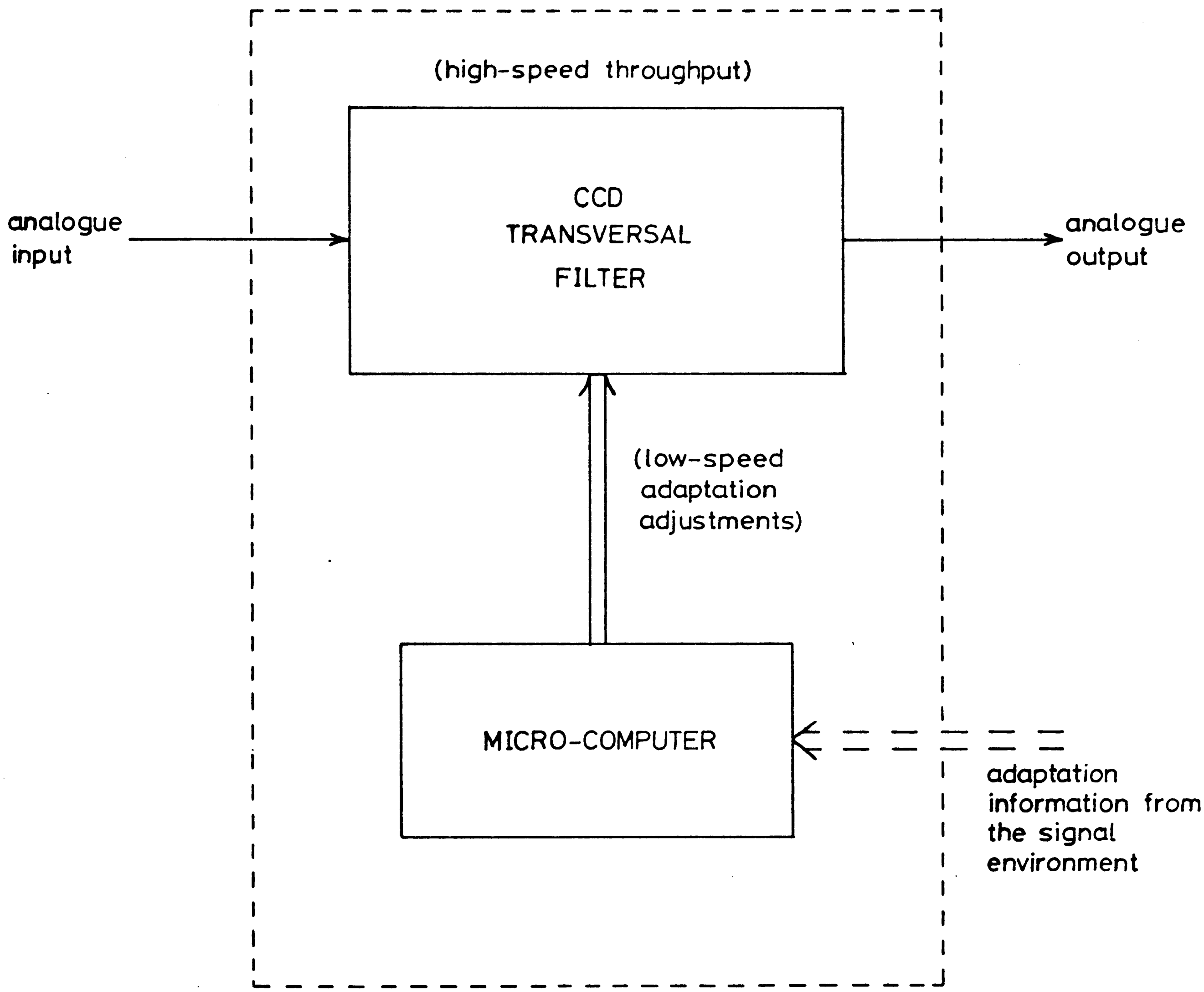
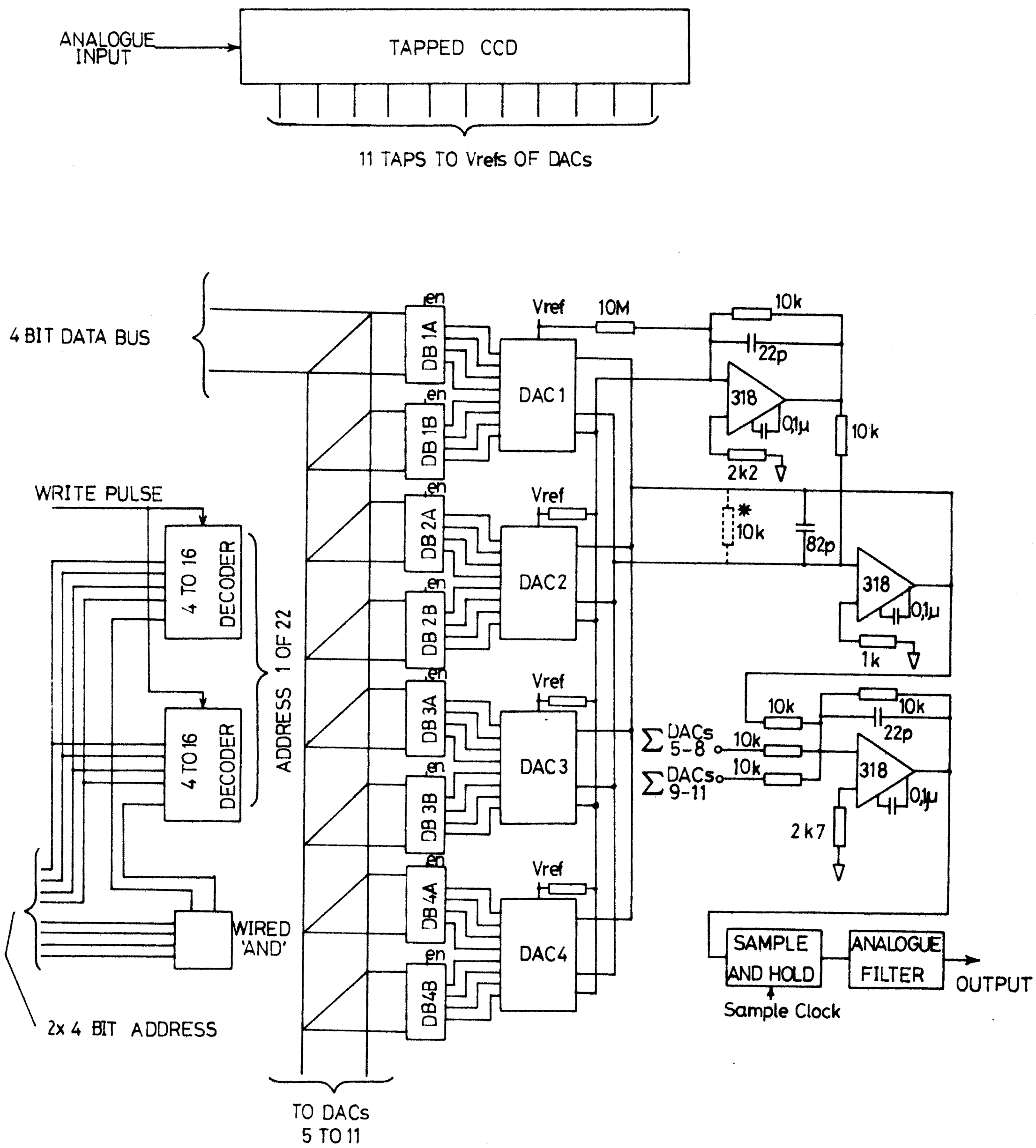


FIGURE 1: MICROCOMPUTER CONTROLLED CCD TRANSVERSAL FILTER



Data Buffers 1A-4B: 74175

DACs 1-4 (etc): AD7520

*extra resistor on DACs 9-11

FIGURE 2: COMPUTER CONTROLLED ADJUSTABLE CCD FILTER

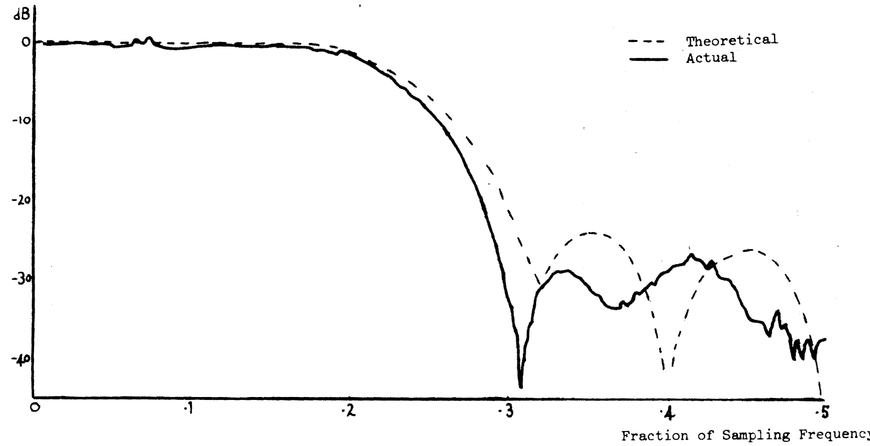


FIGURE 3: LOW-PASS FILTER

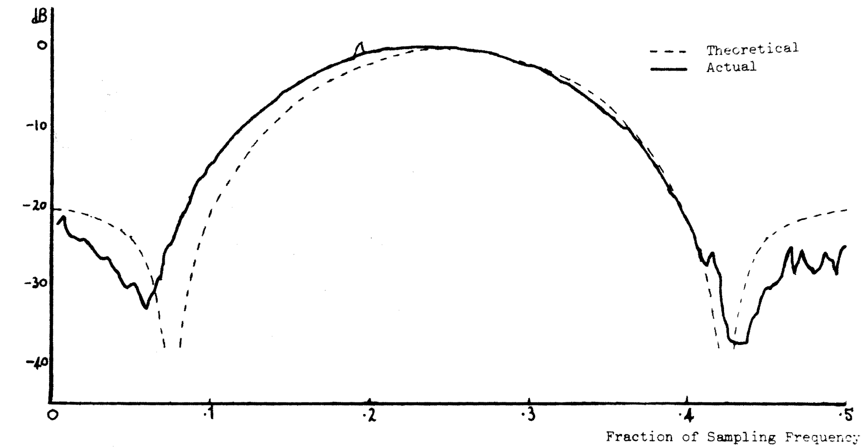


FIGURE 4: BAND-PASS FILTER

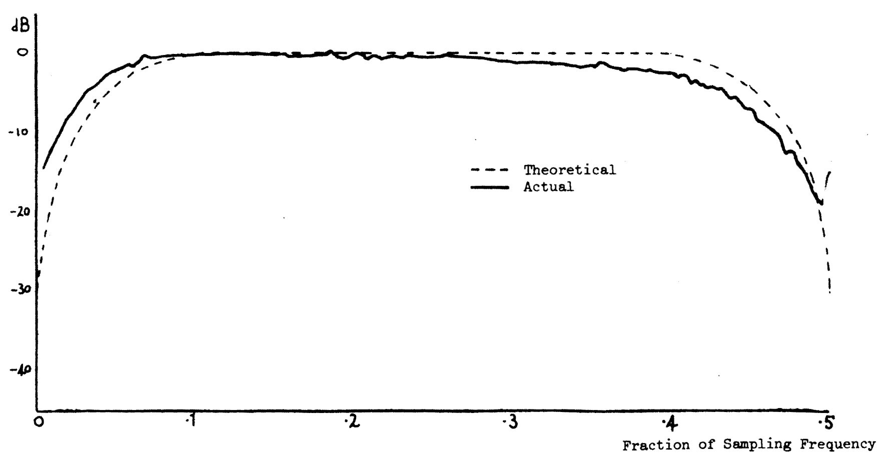


FIGURE 5: HILBERT FILTER

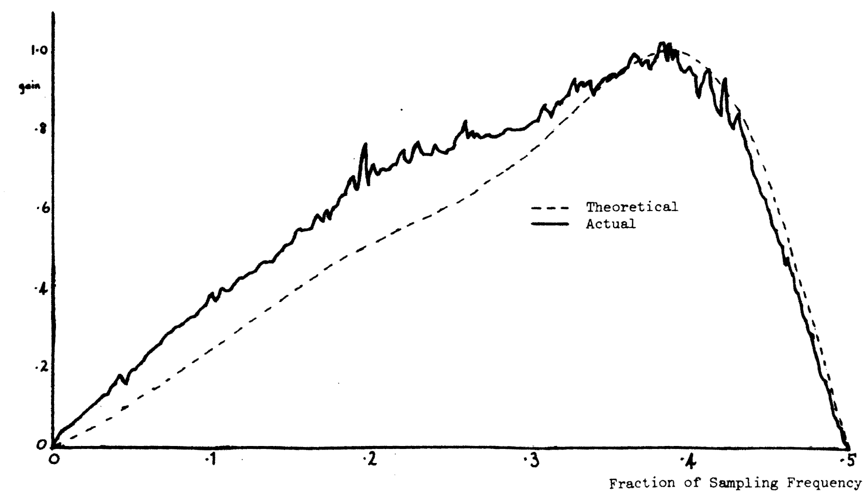


FIGURE 6: DIFFERENTIATOR