SAMPLED ANALOG CCD RECURSIVE COMB FILTERS

T. F. Tao, V. Iamsaad, S. Holmes, B. Freund, L. Saetre
Naval Postgraduate School
Monterey, California

T. A. Zimmerman
TRW Systems
Redondo Beach, California

ABSTRACT - Sampled analog comb filters using a recursive implementation with CCD's as the delay elements are studied. A theoretical analysis based essentially on digital recursive filters is developed with modifications taking into account that a CCD delay generally consists of more than one delay stage. Using the standard Z and the bilinear Z transforms, transfer functions of a lowpass and a highpass continuous filter are transformed into H(Z) forms suitable for CCD implementations. Design formulas for both the canceller type and integrator type comb filters using one CCD delay are obtained. Six different comb filters are implemented using an 8-bit two-phase P surface channel CCD clocked at 20 kHz. The agreement between measurements and theoretical calculations is encouraging. Possible causes for the deviation from theory are discussed. The bilinear Z transform design procedure is shown to be superior because it provides a zero in the transfer function which leads to theoretically infinite attenuation at a series of periodically separated null frequencies. Applications of this type of comb filter to either suppress or to enhance a signal having periodic spectrum are suggested. The work performed was partially supported by the Naval Electronics Systems Command and the Naval Postgraduate School Research Foundation.

1. INTRODUCTION

Designers of sampled analog CCD signal processors have concentrated mainly on transversal filters,[1,2,3] correlators,[4] chirp Z transformers,[5,6] and two-dimensional transforms.[7,8] This paper discusses sampled analog recursive filters, which have not received as much prior attention. They were first studied using a BBD as the delay elements [9,10] and, recently, a three-pole[11] and a two-pole/one zero[12] CCD recursive filter and a one-pole CCD recursive integrator [13] have been reported.

This paper addresses three aspects of sampled analog CCD recursive comb filters: theory, experimental results, and possible applications. The general characteristics of comb filters are highlighted in Section 2, and their recursive implementations are described. In Section 3, a theoretical analysis based essentially on digital recursive filter theory is presented. Modifications of the digital theory to fit the sampled analog recursive filter case are provided. Measured results of recursive comb filters using an 8-bit, two-phase, P surface channel CCD delay line are presented and compared with theoretical calculations in Section 4. The agreement is generally close at low frequencies but deteriorates as the frequency is increased or as the order of comb teeth is increased. However, the feasibility of using a CCD to implement a recursive comb filter is confirmed. Possible applications using this type of comb filter are discussed in Section 5.

2. COMB FILTER AND RECURSIVE IMPLEMENTATION

Comb filters are characterized by their periodic transfer characteristics in the frequency domain. They can be classified into two general types. The first type is the bandstop or canceller type shown in Figure 1a. It has strong attenuation in a narrow neighborhood of a series of periodically separated frequencies and good transmission in between. The second type is the bandpass or integrator type shown in Figure 1b. It has good transmission in a narrow neighborhood of a series of periodically separated frequencies and strong attenuation in between.

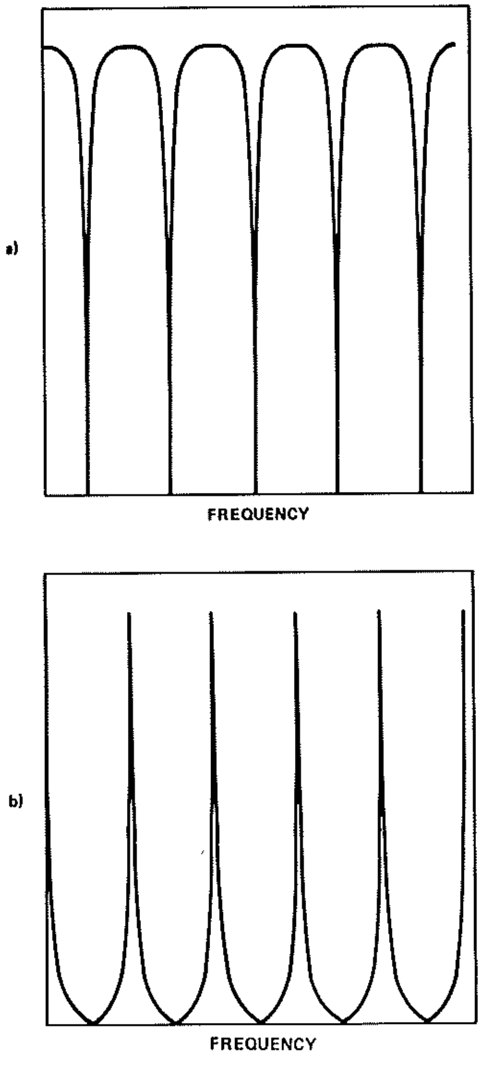

Figure 1. Frequency Characteristics of Two General Types of Comb Filters — Bandstop, Canceller Type, and Bandpass, Integrator Type

Both analog comb filters (17, 18, 19) and digital comb filters (20, 21) have been developed based on several different implementation techniques: feedforward,(22) frequency sampling, fast Fourier transform, and recursion. All use delay devices in some manner.

In the analog case, a quartz delay line has been the major candidate. For digital, the delay device is the shift register. However, a new family of comb filters is being developed using a different type of integrated circuit delay device, such as the BBD (bucket brigade device), CCD (charge coupled device), and SAD (serial analog delay).(23) In these, analog signals are first sampled and then delayed. As a result of the sampling, these devices not only have standard analog properties but also some digital properties such as aliasing and stability of delay.

Using these integrated circuit delay devices, sampled analog comb filters are being developed based on the feedforward circuit,(11, 14, 15) chirp Z transform, and recursive filter implementations. (9, 10, 11, 12)

We are studying the recursive implementation using the canonical circuit shown in Figure 2. The results presented use only one CCD delay device, i.e., $b_2 = 0$, $a_2 = 0$. However, the delay device consists of N delay stages (in the experimental study, N = 8.) As shown in later sections, the presence of N stages of delay in

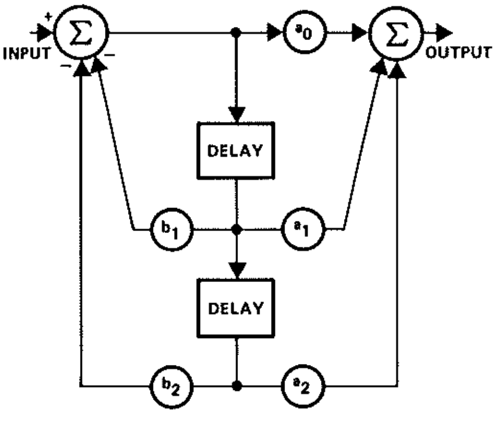

Figure 2. Canonical Form of a Second Order Recursive Filter

one delay device can be used to advantage in providing the comb feature of the frequency characteristics and also very large attenuation between these comb teeth.

The reason for this is that, since the signal is delayed N clock periods before it is fed back to the input, the frequency of recursive operation, f_r, is only one N^{th} of the clock frequency, f_c (or sampling frequency). Therefore, the frequency characteristics of the recursive filter are periodic with the frequency f_r. Since $f_r = f_c/N$, there are N/2 teeth within the Nyquist frequency range from 0 to $f_c/2$. This is in contrast to the usual digital recursive filters where the recursions take place after each delay stage, resulting in only one comb tooth within the Nyquist range. Since there are now N/2 teeth, a proper design procedure which introduces a zero at the recursion frequency f_r will provide infinite attenuation, in principle, at each of these teeth even when only one delay element is used. The procedure is known as the bilinear Z transform.

3. THEORY

Since the sampled analog recursive filters are implemented by the same circuit configurations used for digital recursive filters, we know intuitively that the theory of digital recursive filters should apply. However, proper modifications must be made to account for the fact that the signal is now sampled-analog and that the delay device now consists of N delay stages instead of one.

It is known that two general design approaches have been developed for the IIR (infinite impulse response) type digital recursive filter. The first approach is indirect. It starts with the transfer function in the Laplace transform variable S of a continuous filter and follows by digitizing the transfer function H(S) into a transfer function in the discrete time variable Z. Many different transforms are employed such as mapping by differential transform, impulse invariant (standard Z) transform, bilinear Z transform, and matched Z transform. The second approach is a direct digital design in either the frequency domain or the time domain using some type of computer aided design procedure.

We feel that the direct approach using the frequency domain is probably more instructive, and we will present it in this paper using two of the four transforms listed earlier (standard Z transform and bilinear Z transform).

The modification used to take into account the fact that one CCD delay device consists of N delay stages will be described. In conventional digital filter theory, Z^{-1} is used to represent a delay of one clock period. For a recursive filter using only one CCD device of N delay stages, the transfer function is

$$H(z) = \frac{a_N z^{-N} + a_0}{b_N z^{-N} + 1} \qquad (1)$$

In other words, it is actually an N^{th} order recursive filter but with many of its coefficients, $a_{N-1}, \ldots a_1, b_{N-1}, \ldots b_1$, equal to zero.

The analysis can be simplified greatly if we consider the N clock delay stages as "one recursion" delay stage with its corresponding recursion frequency $f_r = f_c/N$, then the transfer function is simplified to

$$H(z) = \frac{a_1 z^{-1} + a_0}{b_1 z^{-1} + 1} \qquad (2)$$

with the understanding that Z^{-1} actually consists of N clock periods of delay and that its frequency characteristics are periodic with respect to f_r and have N/2 comb teeth in the frequency range from 0 to $f_c/2$, the Nyquist limit. Because there is now more than one period in the frequency characteristics (or more than one tooth), the effects due to sampling and/or the sample and hold circuit should be taken into account in the theory. Only the effect due to the sample and hold circuit is considered in this paper, and it is discussed in Section 4.

Using this straightforward modification and the well developed digital recursive filter theory, a set of design formulas was derived for a first order lowpass filter

$$H(S) = \frac{\omega_x}{S + \omega_x}$$

and a highpass filter

$$H(S) = \frac{S}{S+\omega_x}$$

where ω_x = 3 dB corner frequency. The following two transforms are used:

Standard Z transform $Z = e^{ST}$

Bilinear Z transform $S = \frac{2}{T}\,\frac{Z-1}{Z+1}$

where $T = N\,\frac{1}{f_c}$, the CCD delay.

Table 1. Summary of Design Formula of Sampled Analog Recursive Comb Filter Using One CCD Delay

Design Method	Filter Type	Coefficients		
		a_0	a_1	b_1
Standard Z	Lowpass, integrator $\frac{\omega_x}{\omega_c} < \frac{1}{4}$	ω_x	0	$-e^{-\omega_x T}$
	Lowpass, canceller $\frac{\omega_x}{\omega_c} > \frac{1}{4}$	$1 - \omega_x$	$-e^{-\omega_x T}$	$e^{-\omega_x T}$
	Highpass, canceller $\frac{\omega_x}{\omega_c} < \frac{1}{4}$	$1 - \omega_x$	$-e^{-\omega_x T}$	$-e^{-\omega_x T}$
	Highpass, integrator $\frac{\omega_x}{\omega_c} > \frac{1}{4}$	ω_x	0	$e^{-\omega_x T}$
Bilinear Z	Lowpass, integrator $\frac{\omega_x}{\omega_c} < \frac{1}{4}$	$\frac{\omega_x T}{\omega_x T + 2}$	$\frac{\omega_x T}{\omega_x T + 2}$	$-\left\lvert\frac{\omega_x T - 2}{\omega_x T + 2}\right\rvert$
	Lowpass, canceller $\frac{\omega_x}{\omega_c}\ \frac{1}{4}$	$\frac{\omega_x T}{\omega_x T + 2}$	$\frac{\omega_x T}{\omega_x T + 2}$	$\left\lvert\frac{\omega_x T - 2}{\omega_x T + 2}\right\rvert$
	Highpass, canceller $\frac{\omega_x}{\omega_c} < \frac{1}{4}$	$\frac{2}{\omega_x T + 2}$	$\frac{-2}{\omega_x T + 2}$	$-\left\lvert\frac{\omega_x T - 2}{\omega_x T + 2}\right\rvert$
	Highpass, integrator $\frac{\omega_x}{\omega_c} > \frac{1}{4}$	$\frac{2}{\omega_x T + 2}$	$\frac{-2}{\omega_x T + 2}$	$\left\lvert\frac{\omega_x T - 2}{\omega_x T + 2}\right\rvert$

Without presenting the derivations in detail, the results are summarized in Table 1. It should be pointed out that an approximation, $\tan \omega/\omega_c \cong \omega/\omega_c$, was used. The cases of integrator type comb filters and cancellor type comb filters are listed separately. They are determined by the relative signs of the coefficients a_0, a_1, and b_1. The theoretical frequency characteristics of six special cases are presented as solid curves in Figures 3 through 7. The cases that have been implemented and measured experimentally are summarized in Table 2. The coefficients have all been normalized to unity. Their comb filter characteristics are also plotted in relative amplitudes in Figures 3 through 8. In Figure 8, the theoretical comb filter characteristics are modified by taking into account the effect of the sample and hold circuit. This introduces a factor of $\sin(\pi k/N)/(\pi k/N)$, where k = the order of the comb tooth. Discussions relating to this correction are given in Section 4.

Table 2. Summary of Six Experimental Sampled Analog Recursive Filter Cases

Design Method	Filter Type	Coefficients			Figure
		a_0	a_1	b_1	
Standard Z	Highpass, integrator	1	0	0.7	3
	Lowpass, integrator	1	0	-0.3	4
	Lowpass, canceller	1	1	0.7	5
	Lowpass, canceller	1	1	0.3	6
Bilinear Z	Lowpass, integrator	1	1	-0.9	7
	Highpass, integrator	1	-1	0.7	8

NOTE: 8-bit, 2 phase, surface channel CCD (Figure 9)
f_c = 20 kHz
Circuit schematic (Figure 11)

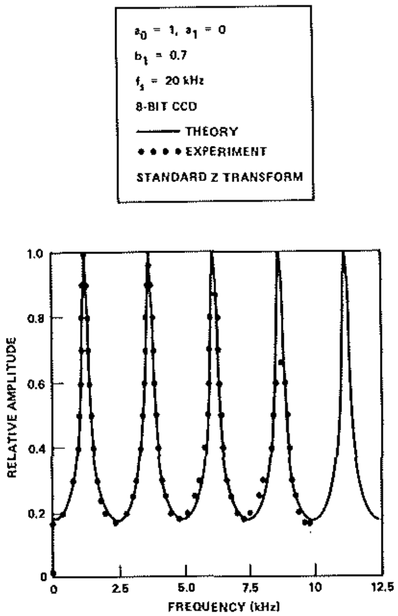

Figure 3. Highpass Integrator Filter — Frequency Characteristics

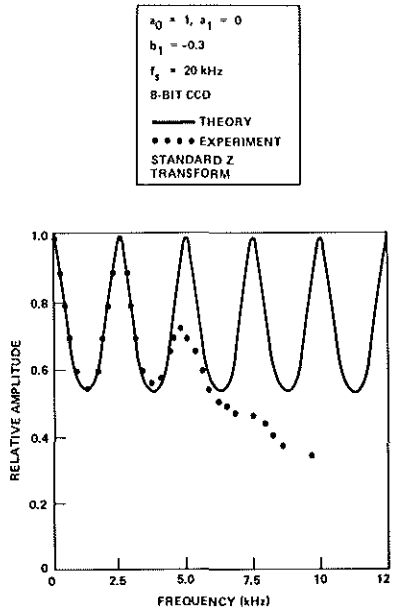

Figure 4. Lowpass Integrator Filter — Frequency Characteristics

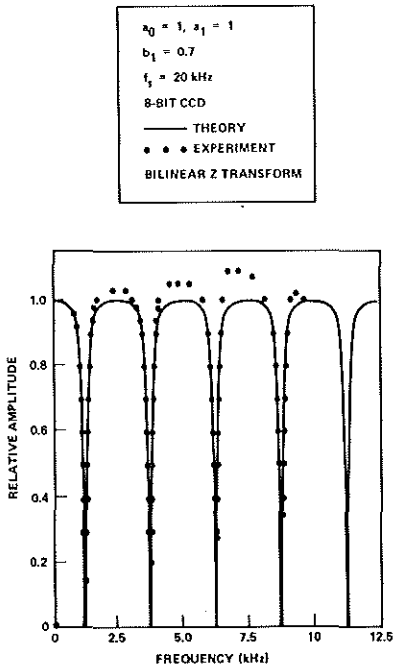

Figure 5. Lowpass Canceller Filter — Frequency Characteristics

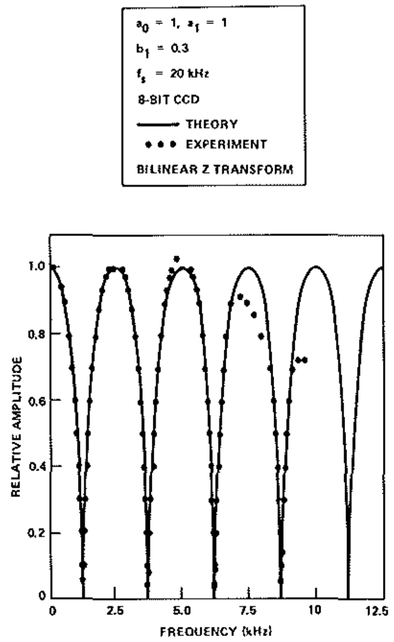

Figure 6. Lowpass Canceller Filter — Frequency Characteristics

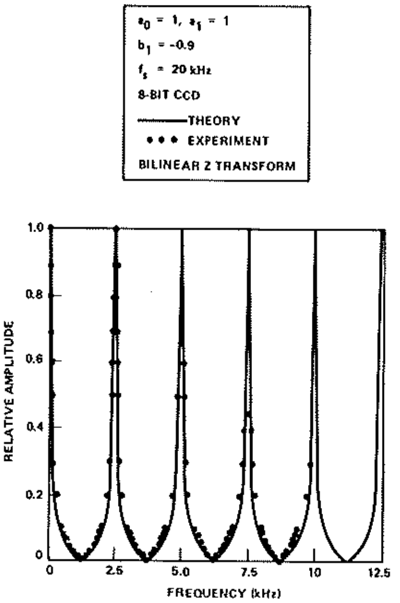

Figure 7. Lowpass Integrator Filter – Frequency Characteristics

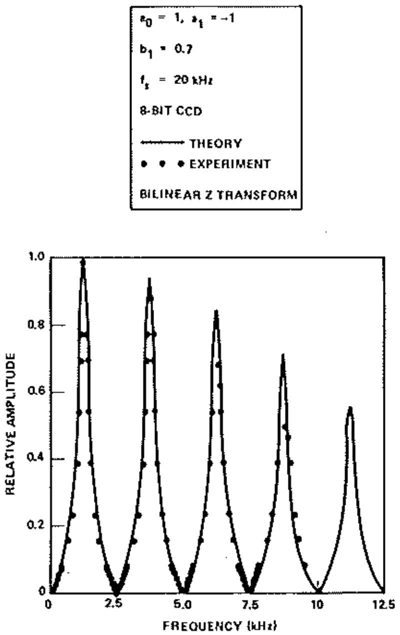

Figure 8. Highpass Integrator Filter – Frequency Characteristics

4. EXPERIMENTS, COMPARISONS, AND DISCUSSIONS

To demonstrate the feasibility of using CCD as delay elements for the implementation of sampled analog recursive comb filters and also to verify the validity of the theoretical analysis presented in Section 3, six cases of comb filters are implemented. They are listed in Table 2. The table includes both canceller type and integrator type comb filters whose Z transform transfer functions were obtained from both lowpass and highpass continuous filters. The CCD used is a two-phase, overlapping gate, P surface channel, 8-bit CCD. Its physical description is presented in Figure 9. Its output circuit consists of a reverse biased floating diode, gated by a reset MOSFET and followed by a MOSFET source follower. The load resistor is connected off the chip. Its input circuit consists of one diode, pulsed to a forward biased condition, and four gates with appropriate dc bias applied.

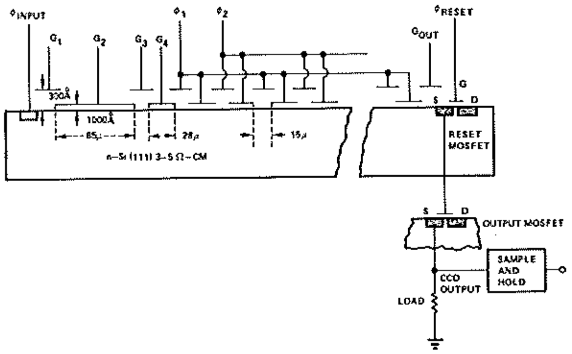

Figure 9. Cross-Sectional View of Two Phase, Overlapping Gate, P Surface Channel CCD

The electrical signal can be fed into the CCD either at the diode or at a gate. In this study, an input signal technique known variously as the surface potential equilibration method, the fill and drain method, or the scuppering method is used. To carry out this input procedure, a clock pulse is applied to the diode, and dc voltages of various levels are applied to the gates. The signal is applied to one of the gates. The combination of third and fourth gates provides one of the widest dynamic ranges, as shown in Figure 10. The figure presents the output voltage measured after the sample and hold circuit as a function of the input signal expressed by a varying dc voltage level at the third gate. The bias voltage on the fourth gate is held at selected values. The operating conditions are described as follows:

- Four clock pulses:

  ϕ_{input} = input diode pulse level from -12.8 to -2.4 V

  ϕ_{reset} = reset MOSFET gate pulse level from -37.5 to -19 V

  ϕ_1, ϕ_2 = 50% duty cycle CCD clock pulses from -22.5 to -10 V

- Four dc gate biases:

  G_1, G_2 biased at -30 V

  G_{out} and V_{DD} at -35.5 V

  Load resistor = 50K ohms

  Clock frequency = 20 kHz.

From Figure 10, it can be seen that two operating modes exist. The first is a high gain mode occurring in a third gate voltage range from approximately -11.7 volts to values from -11.9 to -12.5 volts, depending on the bias voltage on the fourth gate. This mode is quite nonlinear. The second mode is characterized by a much lower gain but occurs over a much wider bias range on G_3. The range is more than 8 V for V_{G4} = -15 V. For V_{G4} = -12 V, it can be seen that a section of the mode characteristics is linear from -16.9 to -20.5 V. Therefore, the combination of V_{G3} = -18.7 V and V_{G3} = -12 V was chosen as the electrical bias condition. The ac input signal was applied to the third gate to carry out the frequency response measurements. In this way, a linear range of approximately 3.7 V was obtained.

The circuit schematic for the comb filter measurements is shown in Figure 11. The coefficient a_0 = 1 is easily implemented by direct connection. The coefficients a_1 = ±1 and the different values of b_1 were implemented by using two potentiometers connected after the sample and hold circuit and the level shifter. Operational amplifiers with the proper feedback were used to provide the two summation operations.

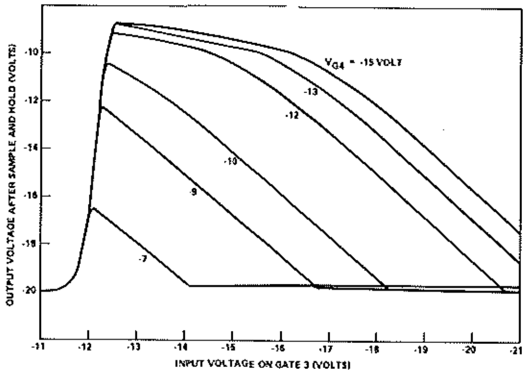

Figure 10. Input G_3 vs Output Characteristics of Two Phase Surface Channel 8 Bits CCD

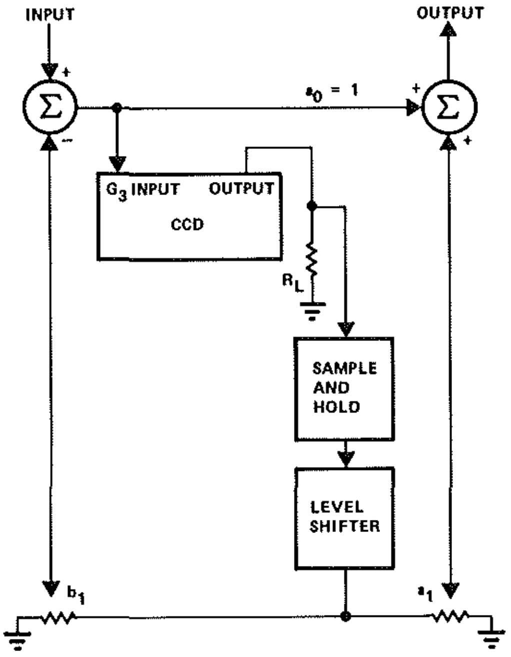

Figure 11. Circuit Schematic of CCD Recursive Filter Experiment

The measured frequency characteristics of six comb filters are presented as dots in Figures 3 through 8. Figures 3 and 4 are filters designed by the standard Z transform procedure. Figures 5 through 8 are filters designed by the bilinear Z transform procedure. The filter type and corresponding filter coefficients are shown in each figure and are also listed in Table 2. The advantage of the bilinear Z transform design procedure over the standard Z transform is clear by noticing the theoretically infinite attenuations achieved at the null frequencies separated by the recursion frequency, $f_r = f_c/N = 20$ kHz$/8 = 2.5$ kHz.

In conventional filter designs, large attenuations in the stop bands can be achieved by using higher order filters in the case of recursive filters and by using many bits of delay in the case of nonrecursive filters. In this case of sampled analog recursive comb filters, infinite attenuation is theoretically achieved even in cases where only one delay element is used. Of course, the reason for such an interesting property is that the delay element has N stages of delay; this causes its frequency characteristic to be periodic with respect to f_c/N and to have $N/2$ teeth within the Nyquist range. A zero introduced by the bilinear Z transform will force the transfer function to zero at a series of null frequencies separated by f_c/N. In the conventional digital recursive filter design where N equals 1, the zero introduced causes the transfer function to become zero at frequencies beyond the Nyquist limit and is therefore not useful.

It should be noted that, in the preliminary experimental study, infinite attenuation was not found. Instead, the measured attenuation varied from -60 dB down from the maximum to -10 dB down. In fact, this is but one of several deviations of measured results from theoretically calculated values. The agreements and disagreements are summarized as follows.

For the first tooth of the comb filter, the agreement between theory and experiment is excellent in all six cases. However, measured values and calculated values started to deviate from each other as the frequency increased or as the order of the comb tooth increased. On the one hand, the attenuation at the null frequencies for the bilinear Z cases started to deteriorate. On the other hand, the filter shape did not always stay in close agreement with the theoretical results. However, different trends of deviation were discovered.

In four cases (Figures 3 and 4 for the standard Z transforms and Figures 7 and 8 for the bilinear Z transforms), the measured frequency response steadily became lower than the calculated response. The worst case is shown in Figure 4. Such a decrease could be caused by several factors including effects due to the sample and hold, effects due to charge transfer inefficiency, and effects due to the frequency rolloff of the electrical instrumentation. In Figure 8, the effect due to the sample and hold circuit is included in the theoretical calculation. However, its rolloff effect does not adequately account for the measured decrease. This suggests that other factors must be considered. The explanation for these deviations is made more complicated by the opposing trend shown in Figures 5 and 6 for two other bilinear Z transform cases. Here the measured frequency responses are higher than the calculated values. We are only beginning to search for the factors which could cause such increases.

The general deterioration of the attenuation at null frequencies is considered to result from two effects. The first is the change of a_1 as a function of frequency; this includes the effects of frequency variations in the CCD, the sample and hold circuit, and the level shifter. Note that a_0 is directly connected to the output summing operational amplifier. The imbalance of a_0 and a_1 makes the "zero" a less than perfect zero and yields less attenuation at the null frequencies. The second effect is the general noise background which prevents the measurement of very small signals. One major contribution of such noise is the clock feedthrough.

In spite of these deviations of measured results from theoretical calculations, it is proper to conclude that the feasibility of using a CCD to implement sampled analog recursive comb filters is confirmed,[10, 11, 12] and the basic approach of a theoretical analysis based essentially on the digital recursive filter theory is correct to the first order. Obviously, refinements are needed to extend the success of the theory at the first comb teeth to all higher order teeth.

5. APPLICATIONS

The range of applications of sampled analog recursive filters is obviously very broad. However, it is important to recognize that the performance of this type of filter is somewhere between that of analog and digital filters. Compared with regular analog filters, the delays of this family of integrated circuit electronic delay devices are controlled by the main clock pulse, and these IC filters do not have nearly the narrow bandwidth and time and temperature drift problems which plague other analog filters. They are as stable as the main clock. Compared with digital filters, they process signals in sampled analog form, thus avoiding the expense of A/D converters and digital multipliers, though there are penalties of somewhat lower performance and relatively more limited programmability. Another feature of sampled analog recursive filters is the possibility of time shared filtering because the delay elements generally have more than one stage.[15]

They are particularly useful as comb filters. Their periodic transfer characteristics make them well suited to process signals which are periodic in time and in frequency such as most radar and sonar signals. Canceller type comb filters can be used to reject periodic signals by aligning the null frequencies of the comb filter with the periodic frequency spectrum of the undesired signal. This is indeed the principle of both the feedforward and recursive filters[11] being developed to cancel clutter spectrum in MTI radars. Also, because the delay can be changed rather conveniently by changing the main clock frequency, the canceller type sampled analog comb filter may simplify the signal processing in either staggered PRF or jittered PRF radars.

In a different direction, integrator type comb filters can be used to enhance periodic signals contaminated by noise and/or interferences. A video integrator is being developed[13] based on this principle.

6. CONCLUSIONS

Sampled analog comb filters by recursive implementation have been studied. A theoretical analysis based essentially on digital recursive filter theory was used to calculate the frequency characteristics of both the canceller type and the integrator type comb filters, using only one delay element. The fact that the delay element has N delay stages is used to achieve the comb feature of the transfer characteristics. Using the zero introduced by the bilinear Z transform, infinite attenuations are achieved at null frequencies, which is a very desirable feature. Six calculated cases were selected for implementation using an 8-bit CCD as the delay element. The agreement between measurements and theory is excellent at the first set of comb teeth and varying degrees of closeness are achieved as the order of comb teeth is increased. The causes for this are under study. Studies are in progress to extend this work to cases where more than one delay element is used and to examine the reasons why the experimental results at higher order comb teeth are at variance with some of the theoretical predictions.

BIBLIOGRAPHY

1. D. D. Buss, D. R. Collins, W. Bailey, and C. Reeves, IEEE J. Solid States Circuits, SC-8, 1973, p. 134.
2. M. H. White, D. R. Lampe, F. J. Fagen, Int. Electronics Device Meeting, Tech. Digest, Washington, D. C., Dec. 1973, p. 130.
3. P. J. MacLennan, J. Mavor, G. Vanstone, and D. J. Windle, Proc. Int. Conf. Tech. and Appl. of CCD, Edinburgh, Sept. 1974, p. 221.
4. J. Tiemann, W. Engeler, R. Baertsch, IEEE J. Solid State Circuits, SC-9, 1974, p. 403.
5. R. W. Means, D. D. Buss, H. J. Whitehouse, Proc. CCD Appl. Conf., San Diego, Sept. 1973, p. 95.
6. R. W. Broderson, H. S. Fu, R. C. Frye, D. D. Buss, ISSCC Tech. Paper Digest, Philadelphia, Feb. 1975, p. 144.

7. I. Lagnado and H. J. Whitehouse, Proc. Int. Conf. Tech. and Appl. of CCD, Edinburgh, Sept. 1974, p. 198.

8. H. J. Whitehouse and R. W. Means, IEEE Int. Symp. on Ckts and Systems, Publication IEEE Advanced Solid State Comp. for Signal Processing, April 1975, p. 5.

9. D. A. Smith, C. M. Puckette, and W. J. Butler, IEEE J. Solid State Circuits, SC-7, 1972, p. 421.

10. D. A. Smith, W. J. Butler, and C. M. Puckett, IEEE Trans. Communications, COM-22, 1974, p. 921.

11. J. E. Bounden, R. Eames, and J. B. Roberts, Proc. Int. Conf. Tech. and Appl. of CCD, Edinburgh, Sept. 1974, p. 206.

12. J. Mattern and D. Lampe, ISSCC Tech. Paper Digest, Philadelphia, Feb. 1975, p. 148.

13. J. B. G. Roberts, M. Chesswas, and R. Eames, Electronics Letters, 10, 1974, p. 169.

14. W. J. Butler, W. E. Engeler, H. S. Goldberg, C. M. Puckette, and H. Lobenstein, IEEE Int. Symp. Ckts and Systems, Publication IEEE Advanced Solid State Comp. for Signal Processing, April 1975, p. 30.

15. M. F. Tompsett, A. M. Moshen, D. A. Sealer, and C. H. Sequin, IEEE Int. Symp. Ckts and Systems, Publication IEEE Advanced Solid State Comp. for Signal Processing, April 1975, p. 83.

16. W. H. Bailey, R. W. Broderson, W. L. Eversole, M. M. Whatley, L. R. Hite, D. D. Buss, and R. Sproat, Digest of Papers, 1974 Government Microcircuit Applications Conference, 5, 1974, p. 76.

17. W. D. White and A. E. Ruvin, IRE Conv. Record, Pt. 2, 1957, p. 186.

18. A. G. J. MacFarlane, Proc. IEEE, Paper 3121E, 1960, p. 39.

19. G. T. Flesher, Proceedings National Electronics Conference, 14, 1958, p. 282.

20. C. M. Rader, B. Gold, Proceedings IEEE, 55, 1967, p. 149.

21. S. A. White, IEEE Tran. Automatic Control, AC-14, 1969, p. 423.

22. R. E. Bogner, Bell Sys Tech Journal, 48, 1969, p. 501.

23. Reticon Corporation, Sunnyvale, California.

24. L. Rabiner, B. Gold, Theory and Application of Digital Signal Processing, Prentiss Hall, 1975.

ACKNOWLEDGEMENTS

The authors are grateful to the following persons for their technical advice and for their assistance in instrumentation and measurements: P. Kopp, S. R. Parker, and D. Hoisington of the Naval Postgraduate School; J. L. Rogers and A. C. Schoening of TRW; and G. Temes of UCLA.