An Adapting Delay Comb Filter for the Restoration of Audio Signals Badly Corrupted with a Periodic Signal of Slowly Changing Frequency*

BERNARD A. HUTCHINS, JR., AND WALTER H. KU

Cornell University, School of Electrical Engineering, Ithaca, NY 14853, USA

The adapting delay comb filter is a device for acquiring, nulling, and tracking an unwanted periodic interfering signal that is corrupting a desired program signal. A feedback structure is used to make a standard comb filter self-adjusting so that it will attempt to null out a strong periodic signal, passing the desired signal by default.

0 INTRODUCTION

The audio engineer may at times be called on to attempt to restore a recording that has been badly corrupted with a periodic interference (high-level hum) and thereby made useless. Simple approaches to the problem are often ineffective because of the presence of numerous harmonics of the interfering signal, and because of variations in the fundamental frequency of the interference. A good example would be the case of a tape recording made with an improperly grounded signal line. Such poor recording practices are frequently found in conjunction with recorders of inferior quality. The result can be a recording with ac hum of a high level, and due to poor recorder quality, the tape will not yield exactly 60-Hz hum, or even a steady hum frequency, on playback (due to warped tape drive parts, weak batteries, etc.).

1 NONADAPTIVE APPROACHES

Attempts to use low-pass filters or 60-Hz notch filters will usually be disappointing in this case. The main reason is that the fundamental component, the 60-Hz sine wave, may not be the major culprit here. Due to the relative insensitivity of the ear to frequencies as low as 60 Hz, a mixture of 90% 60-Hz sine wave and 10% speech, for example, will still be of good intelligibility, unless, of course, the 60-Hz sine wave saturates the electronics. Most so-called 60-Hz hum is usually due in large part to harmonics of 60 Hz. This can be easily demonstrated by grasping an oscilloscope probe wire with the hand and observing the waveform picked up by the human "antenna" as it is displayed on the scope face. The hum is seen to be composed of a major 60-Hz component, but it is not a perfect sine wave, indicating the presence of harmonics which are due to machinery and other devices on the ac lines. While the harmonics may appear minor, the ear is much more sensitive to them.

The delay line comb filter depicted in Fig. 1 is a powerful tool for removing periodic interference. If the delay time T of the line is equal to the period of an interfering waveform, it is clear that that particular waveform will be canceled by the differencing summer. A more complete analysis [1] shows that the frequency response of the comb filter consists of a series of half sinusoidal lobes with nulls spaced at multiples of $1/T$, as shown. Thus it is also possible to understand the cancellation of the periodic interference as a notching out of each of its component frequencies.

Comb filters of the type shown in Fig. 1 are easily implemented with analog delay lines of the bucket-brigade or of the charge-coupled-device type. Suitable temporary setups may also be implemented with digital delay lines and studio mixers, or with "flanging" units. Yet these simple fixed delay comb filters suffer when the input interference is not of a relatively fixed frequency, since it must then be manually adjusted on a

* Manuscript received 1981 May 4.

continual basis. It should be pointed out, however, that the sinusoidal shape of the passbands is not usually a problem, even though they do result in the removal of some of the desired program material. The principal reasons for this are that the program material is generally rapidly changing and moves through the weak regions rapidly, and that the degree of improvement is from a virtually useless state to a reasonably useful state, even with the apparently poor passbands.

2 THE ADAPTING DELAY COMB FILTER (ADCF)

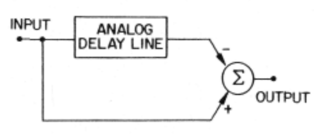
2.1 Basic Theory of the ADCF

The ADCF described here is a device that will acquire, null, and track a strong periodic interference. In the process it tends to ignore the weaker and rapidly changing program material, and most of this is passed. The principal problem to be solved in such an acquisition and tracking system can be understood by considering that while proper adjustment of the delay results in a minimum level of interference at the output, a nonminimal level indicates only that the delay should be adjusted. It does not indicate whether the delay time should be shorter or longer. To solve this problem, the ADCF employs three comb filters as shown in Fig. 2, the center comb being the one that does the actual processing, while the late and early combs examine the signal level that is achieved with longer and shorter delay times. A balance of these two side filters is used to keep the main filter on track. The overall feedback system resembles somewhat a phase-locked loop.

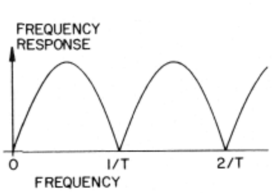
In the actual ADCF the delay time T is realized as a main delay line and half a shorter tapped delay line, such that t is small compared to T. Thus the nulls of the side filters are relatively close to the center null, and Fig. 3 shows the details of the bottoms of these three closely spaced nulls. When the filter is properly tracking, the frequency component to be nulled is at $1/T$, and the signal levels in the two side filters are approximately equal, as shown in the upper portion of Fig. 3. The lower portion of Fig. 3 shows the case where the frequency component has moved up slightly. Now the signal getting through the $1/(T + t)$ notch is greater than that in the $1/(T - t)$ notch. This out-of-balance condition indicates both improper tuning of the center notch and the proper direction for correction. As seen in Fig. 2, the signal levels of the side combs are first detected for amplitude with the absolute value detectors, and then applied to a differential integrator. With the signal imbalance, the integrator ramps in response to a differential input, and the output is fed through a summer to a voltage-controlled oscillator, which in turn determines the clocking rate on the delay line, and thereby determines the delay time T. The polarity is such that the adjustment restores the balanced condition.

2.2 Circuit Description

Fig. 4 shows a working version of the ADCF based on the Reticon type SAD-1024 and TAD-32 BBD delay lines. IC-1 through IC-5 form a two-phased voltage-controlled oscillator which clocks both delay lines. The main delay line is IC-7 (256 sample stages used), with IC-6 buffering and level shifting the input, while IC-8 buffers and reshifts the output, thus feeding the shorter delay times $T - t$, T, and $T + t$. These tapped signals, as decoupled and buffered (IC-11 typical), are mixed with the input as inverted by IC-9, and thus form the three needed comb filters (IC-12 typical). The ampli-



(a)



(b)

Fig. 1. (a) First-order nonrecursive comb filter. (b) Its frequency response.

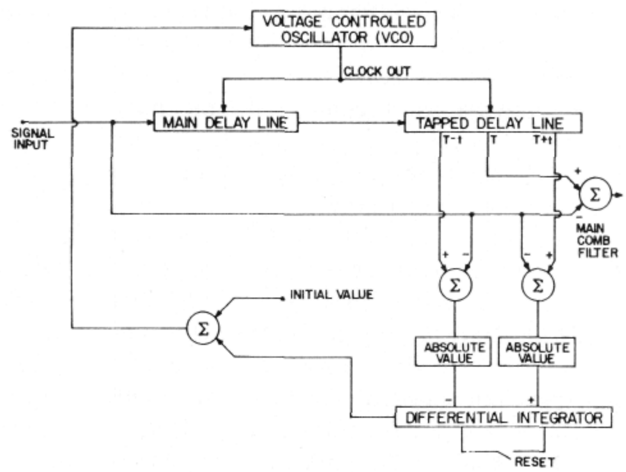


Fig. 2. Block diagram of adapting delay comb filter (ADCF).

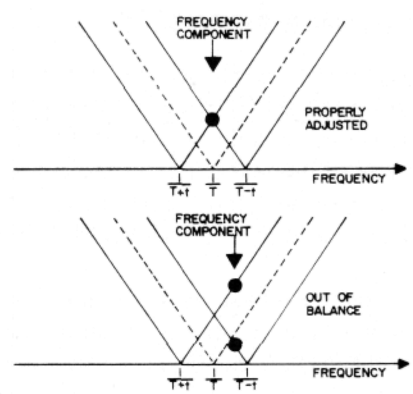


Fig. 3. Conditions at bottom of nulls of ADCF.

tudes of the side comb filters are detected by full-wave rectifiers (IC-17 and IC-18 typical) and fed to the differential integrator (IC-21 and IC-22), the output of which is fed back to control the voltage-controlled oscillator, thus completing the feedback loop.

Some adjustment of the dc level at the input of IC-7 may be necessary, and both delay lines are trimmed for dc level and signal level so as to avoid clipping and distortion. The comb filters are trimmed using a sine wave input with the reset switch closed so that each notch can be adjusted in turn.

3 USING THE ADCF

To use the ADCF, the signal to be processed is applied with the reset switch closed (output of IC-20 at zero). The initial frequency control is then set manually, if possible, to the highest frequency that gives rejection of the interfering signal, or to the approximate frequency of the interfering signal if known, or at random if nothing about the interference is known or if conditions are rapidly varying. When the reset switch is opened, the ADCF will properly capture the interference if it is within a ratio of 0.5–1.5 of this initial frequency (the spacing between the peaks of the response), and track over a much wider range if necessary. As a final step, the 50-kΩ balance control can be adjusted to fine tune the rejection.

Nulling of the interference will in general occur even if the interference frequency appears above 1.5 times the initial setting. However, in such a case an upper notch captures the fundamental, and there are additional unnecessary notches between the needed ones; this is obviously less desirable than having the fundamental in the lowest notch.

The response rate of the ADCF is determined mainly by the time constant of the differential integrator, which is 100 kΩ times the capacitor in the feedback loop of IC-22 (0.82 mF, giving an RC time constant of 82 ms, as shown). This time constant is adequate for many purposes, but a switch making available different time constants is suggested for a general ADCF. The time constant should be fast enough for satisfactory capture time and tracking response, but not so fast that the filter might try to null portions of the program material.

4 PERFORMANCE DISCUSSION

Typical signal-to-noise improvements are in the range of 20–30 dB for the ADCF shown. Subjectively, speech signals that are impossible to understand or annoying to listen to can be made intelligible and comfortable to listen to, although the quality of the recovered audio is not totally natural due to the sinusoidal shape of the passbands.

The performance is best where the interference is on the order of 10–100 times stronger than the signal, since in this case the ADCF clearly keys to the interference rather than the desired signal. In cases where the interference is not so strong, separation can often still be achieved on the basis of the time constant. For example, if the interference is relatively steady over a period of several seconds, and the program material is speech, changing at least every 100 ms, a time constant of several seconds for the ADCF may be effective in nulling the interference, while largely ignoring (passing) the speech.

Since only half the SAD-1024 line is being used in the circuit as shown, the second half can be added to tap 27 of the TAD-32, and a second-order response can be implemented. This second-order response must be nonrecursive in order to not effect the capture and hold properties of the existing system. This allows the addition of a second zero to the response, in addition to the zero already at $z = +1$ in the z plane. Placing this zero at $z = -0.3$ is a good choice, and does result in a more natural sound to the speech due to the flatter passbands. However, due to the sharper slopes on the sides of the notches, any tuning error is somewhat more serious, and rejection is in general not quite as good as with first order.

5 ACKNOWLEDGMENT

This work was supported by Rome Air Development Center (RADC), Deputy for Electronic Technology, Hanscom Air Force Base, MA, under Contract F49620-77-C-0069 (Dr. William Ewing, technical monitor). The authors wish to acknowledge the assistance of Edward Kellett who studied the capture properties of the ADCF by computer simulation, and the independent suggestion of a portion of the ADCF principle in a private communication from Nicholas Bodley.

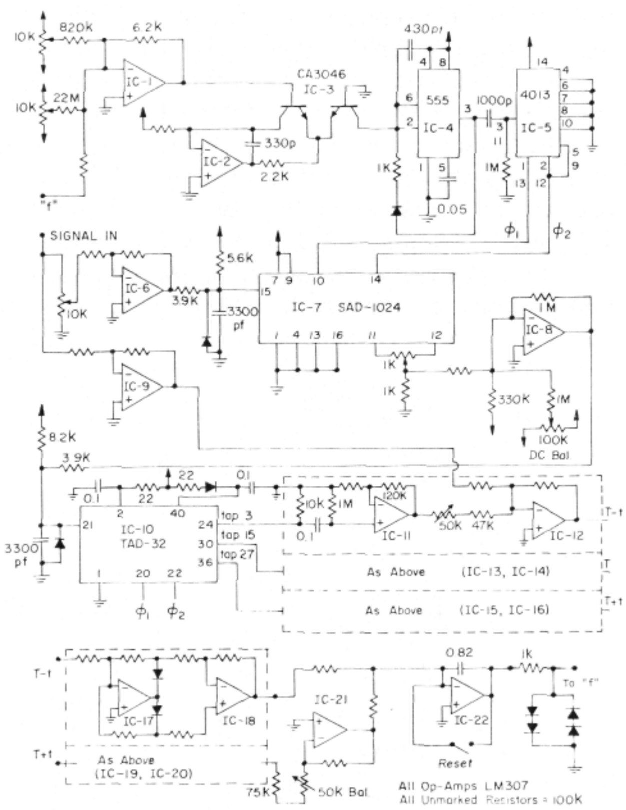


Fig. 4. Complete circuit diagram of ADCF.

6 REFERENCES

[1] B. A. Hutchins, Jr., and W. H. Ku, "Audio Frequency Applications of Integrated Circuit Analog Delay Lines," presented at the 51st Convention of the Audio Engineering Society, *J. Audio Eng. Soc. (Abstracts)*, vol. 23, p. 486 (1975 July/Aug.), preprint no. 1032.

THE AUTHORS

Bernard A. Hutchins, Jr. received a B.S. degree in engineering physics from Cornell University in 1967. He has been associated with the School of Electrical Engineering, Cornell University, since 1974 with both research and teaching responsibilities. Mr. Hutchins is also the editor and publisher of *Electronotes*: *Newsletter of the Musical Engineering Group*. In this connection, he has pursued his interest in the analysis, synthesis, and processing of audio and musical sounds. He is a member of the Audio Engineering Society.

Walter H. Ku received a B.S. degree (with honors) from the University of Pennsylvania, and an M.S. and Ph.D. from the Polytechnic Institute of Brooklyn, Brooklyn, NY. In 1969 he joined the faculty of the School of Electrical Engineering, Cornell University, where he is professor of electrical engineering. His current research interests are in the areas of active and microwave circuit design, digital filter structures, signal processing, and digital communication systems. Dr. Ku is a member of the IEEE, Sigma Xi, and Tau Beta Pi.