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ABSTRACT

This report describes a method to eliminate periodic noise signals from a disturbed signal. Starting with the signal description of the periodic noise it is shown that it is possible to implement a filter with zeroes at the frequencies of the noise. Finally some areas of application are discussed.

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1 INTRODUCTION.

When speech is disturbed by noise a person with a normal hearing will normally catch the speech much better than a person with a hearing damage when listening with one ear. This is due to the fact that the normal ear has a higher channel capacity and thereby can use the redundancy in the speech signal better. Therefore even a minor improvement in the signal-to-noise ratio should be experienced very positively by a hearing impaired listener.

If the useful signal and the noise signal are covering different frequency areas the noise can be cancelled by a conventional filter.If they do cover the same frequency area, and the noise is random, the noise can not be eliminated without loss of the useful signal.

If however the noise has some deterministic property as when the noise is periodic there are possibilities of a selective elimination of the noise.
For periodic noise a method has been worked out on a computer and also been realized in a specially built hardware network.

Examples of periodic noise which influence the perception of speech and music are engine sounds from machines and vehicles and noise emanating from the AC mains. Especially in places where thyristors are used to control light or machines one gets a broadband noise spectrum which can be very annoying.

This report is organized as follows:
The mathematical model for the periodic noise signal and the filter which eliminates the noise are treated in chapter 2.
The hardware realization of the method is shown in chapter 3.
Finally some applications are discussed in chapter 4.

2 SUPPRESSION OF PERIODIC NOISE.

A periodic signal can be described mathematically through

$$f(t) = \sum_{n=-\infty}^{\infty} c_n \exp(j2\pi nt/T) \qquad (1)$$

i.e. it just contains frequencies which are multiples of 1/T where T is the period of the signal and $j=\sqrt{-1}$.
Another description is

$$\begin{cases} f(t) = g(t) & 0 \leq t < T \\ f(t) = f(t-T) & \text{otherwise} \end{cases} \qquad (2)$$

If the sum of a useful signal x(t) and a disturbing periodic signal f(t) is passed through a filter with an impulse response

$$h(t) = \delta(t) - \delta(t-T) \qquad (3)$$

where $\delta(\ )$ is the Dirac delta function, the output signal will be

$$y(t) = x(t) + f(t) - x(t-T) - f(t-T) \qquad (4)$$

But according to (2) f(t) is equal to f(t-T) so the output signal is reduced to

$$y(t) = x(t) - x(t-T) \qquad (5)$$

i.e. the signal contains the useful signal and a delayed version of it, and the periodic noise is eliminated.
If the delay is more than approximately 30 ms, the delayed signal will be percieved as an echo. Speech intelligibility may then be increased if the delay is in the order of 50 ms (the so-called Haas-effect) but the echo can be very disturbing for longer delay times. If the delay is less than a few milliseconds the lower frequency bands will be considerably depressed. If the delay, however, is more than some 5 ms such negative effects will be avoided.

2.1 Periodic filter.

Formula (3) above shows the impulse response of the filter that eliminates a periodic noise signal.
If (3) is Fourier transformed one gets the transfer function of the filter

$$H(s) = 1 - \exp(-sT) \qquad (6)$$

where s is complex frequency. If in this expression $j\omega$ is substituted for s one gets

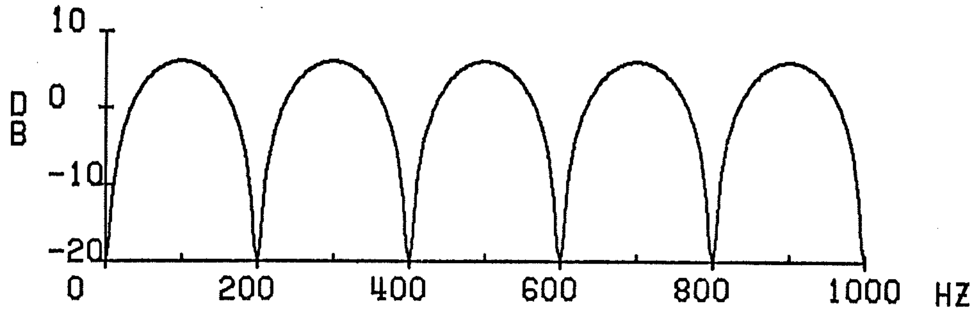
$$H(j\omega) = 1 - \exp(-j\omega T) =$$

$$= 1 - \cos\omega T + j\sin\omega T \qquad (7)$$

This expression is identically equal to zero when $\omega = 2\pi n/T$ where n is an arbitrary positive or negative integer or zero. The filter therefore has zeroes along the $j\omega$-axis in the complex frequency plane and is periodic.

The frequency curve for the filter with impulse response h(t) will consequently have a strongly varying amplitude characteristic. Figure 1 shows the frequency curves for two different delays. Listening tests show that these strongly varying frequency curves do not appreciably influence the impression of sounds and therefore it is in practice of no importance.

a)



b)

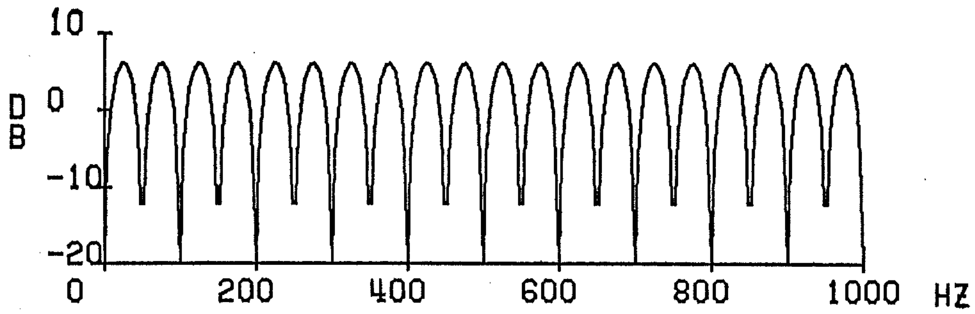


FIGURE 1. Frequency curves when the delay is
a) 5 ms and b) 20 ms

3 HARDWARE REALIZATION.

The filter that eliminates a periodic noise signal is according to (3) composed of a direct channel and a delay channel. As the preceeding chapter showed the delay time should be chosen to be exactly one period of the noise signal. If the delay is 20 ms (figure 1b), the signal components with frequencies which are multiples of 50 Hz will disappear from the output signal. A delay of 20 ms will therefore remove disturbances emanating from the AC mains. (For 50 Hz power systems).

Figure 2 shows the block diagram for the filter circuit.

To make the system adapt to arbitrary noise frequencies (and even to variations in AC main frequency) it has a syncronizing circuit which detects the noise frequency and adapts the delay time to this.

3.1 Delay circuit.

An integrated circuit from Reticon, SAD 1024, has been used to realize the delay channel. The circuit is an analog shift register of CCD type (CCD, Charge Coupled Device), where the input signal is sampled syncronously with a clock signal. The signal processing is thus time discrete and to avoid aliasing effects the input signal is first lowpass filtered.

SAD 1024 consists of two delay lines of 512 samples each. It

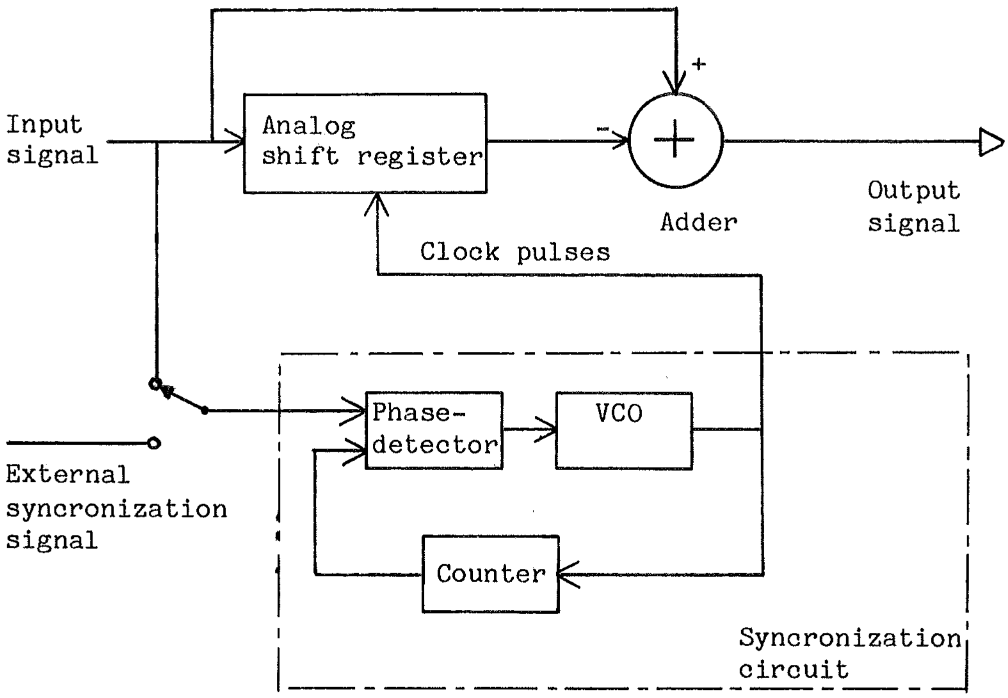


FIGURE 2. Block diagram of periodic filter

would therefore be possible to get a delay of 1024 samples but this is not a good solution as the circuit generates quadratic distortion. Instead the delay lines are used in parallel but with the signal in one of them inverted. By subtracting one of the output signals from the other the distortion is reduced.

The attenuation in the CCD circuit is temperature dependent. To compensate for this an amplifier with variable gain is inserted after the delay line. As shown by figure 1 and equation (7) the filter should have zero amplification for DC. By adding a DC level to the filter input and then changing the gain till the DC level after the adder is zero the temperature dependence can be compensated. The filter characteristics are then kept constant.

3.2 Syncronizing network.

The syncronizing network has to control the sampling rate of the analog shift register so that the delay time is equal to one period of the noise signal.
As input signal to the syncronizing circuit either the input signal to the analog shift register or an external signal can be chosen. The external signal is most conveniently used if one has the possibility to detect the noise frequency at the origin as the case is if the disturbance is caused by the AC mains.
The syncronizing network consists of a binary counter and a phase locked loop (PLL). The PLL is made up by a phase detector and a voltage controlled oscillator (VCO). The counter divides the clock frequency by a factor that equals the number of samples in the shift register. The phase detector compares the input signal and the divided signal and controls the VCO so that the frequencies of the input signal and the divided signal are equal.

4 APPLICATIONS.

4.1 Hearing aids.

When listening through a hearing aid in for instance a theatre one likes to use the telephone pickup coil to get a better S/N ratio. If however thyristors are used to control the light they can produce so much disturbance in the induction loop that listening via the loop is impossible. A solution to this problem would be to support the hearing aid with a periodic filter to reduce the disturbance.
At the moment this is unfortunaly not possible as the delay circuit needs a higher power source voltage than that which is used in hearing aids today.

4.2 Hearing protectors.

A hearing protector with good attenuation for periodic noise sounds and low attenuation for non-periodic sounds and pseudo-periodic sounds like speech will result if an ordinary hearing protector is supported with a microphone and an earphone and a periodic filter is inserted between them. Sounds from engines would be attenuated by the filter and normal speech communication could be maintained at the same time.
In industrial environments with periodic noise from rotating machines such a hearing protector would be of great value.

4.3 Recordings.

The method has been used on an important recording of a conference where a microphone contact failed, giving such a high periodic broadband noise that speech was not detectable.

The noise could not be eliminated using conventional filters but was successfully reduced by the periodic filter.