The practical value of an ALE as a preprocessor to a conventional matched filter has been demonstrated by Nielson and Thomas (1988) as a means of improving the performance of the detector in the presence of Arctic ocean noise. This type of noise is known to have highly non-Gaussian and nonstationary characteristics; hence the benefit to be gained from the use of an ALE.

Adaptive Noise Canceling

As the name implies, adaptive noise canceling relies on the use of *noise canceling* by subtracting noise from a received signal, an operation controlled in an *adaptive* manner for the purpose of improved signal-to-noise ratio. Ordinarily, it is inadvisable to subtract noise from a received signal, because such an operation could produce disastrous results by causing an increase in the average power of the output noise. However, when proper provisions are made, and filtering and subtraction are controlled by an adaptive process, it is possible to achieve a superior system performance compared to direct filtering of the received signal (Widrow et al., 1975b; Widrow and Stearns, 1985).

Basically, an adaptive noise canceler is a *dual-input, closed-loop adaptive feedback system* as illustrated in Fig. 24. The two inputs of the system are derived from a pair of sensors: a *primary sensor* and a *reference (auxiliary) sensor*. Specifically, we have the following:

1. The primary sensor receives an *information-bearing signal* $s(n)$ corrupted by *additive noise* $v_0(n)$, as shown by

$$d(n) = s(n) + v_0(n) \tag{35}$$

   The signal $s(n)$ and the noise $v_0(n)$ are uncorrelated with each other; that is,

$$E[s(n)v_0(n-k)] = 0 \qquad \text{for all } k \tag{36}$$

   where $s(n)$ and $v_0(n)$ are assumed to be real valued.

2. The reference sensor receives a noise $v_1(n)$ that is *uncorrelated* with the signal $s(n)$ but *correlated* with the noise $v_0(n)$ in the primary sensor output in an *unknown* way; that is,

$$E[s(n)v_1(n-k)] = 0 \qquad \text{for all } k \tag{37}$$

   and

$$E[v_0(n)v_1(n-k)] = p(k) \tag{38}$$

   where, as before, the signals are real valued and $p(k)$ is an unknown cross-correlation for lag k.

The reference signal $v_1(n)$ is processed by an adaptive filter to produce the output signal:

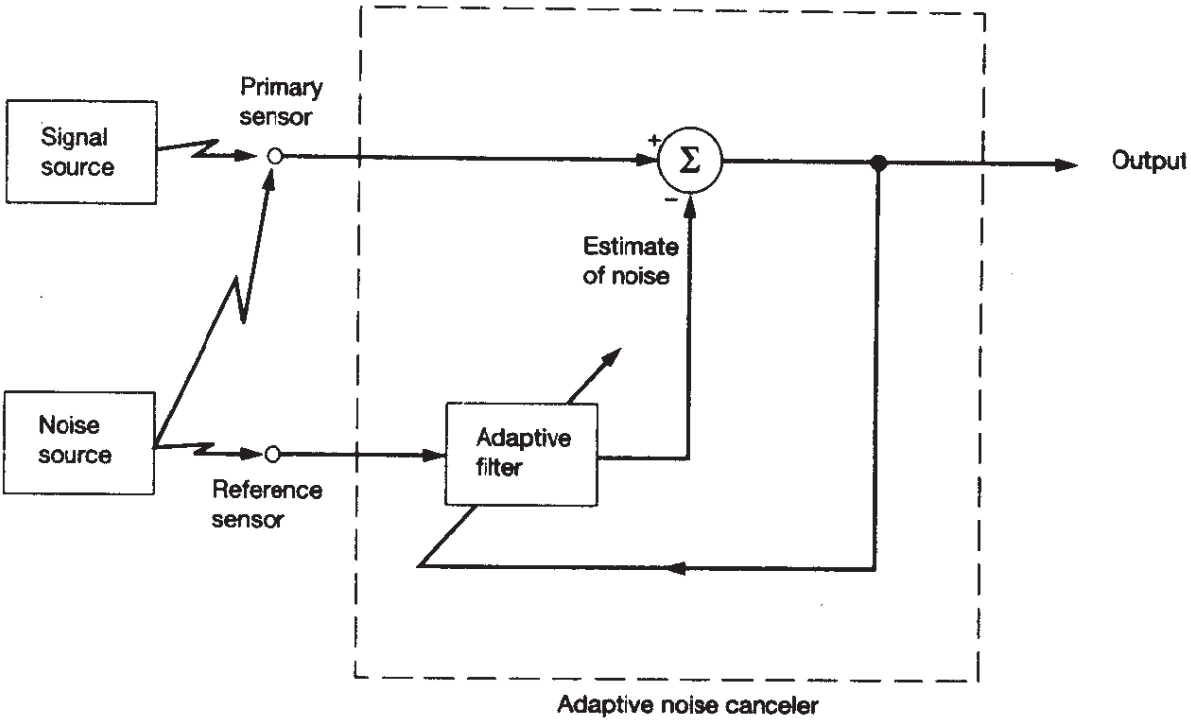

Figure 24 Adaptive noise cancelation.

$$y(n) = \sum_{k=0}^{M-1} \hat{w}_k(n) v_1(n-k) \tag{39}$$

where the $\hat{w}_k(n)$ are the adjustable (real) tap weights of the adaptive filter. The filter output $y(n)$ is subtracted from the primary signal $d(n)$, serving as the "desired" response for the adaptive filter. The error signal is defined by

$$e(n) = d(n) - y(n) \tag{40}$$

Thus, substituting Eq. (35) in (40), we get

$$e(n) = s(n) + v_0(n) - y(n) \tag{41}$$

The error signal is, in turn, used to adjust the tap weights of the adaptive filter, and the control loop around the operations of filtering and subtraction is thereby closed. Note that the information-bearing signal $s(n)$ is indeed part of the error signal $e(n)$, as indicated in Eq. (41).

The error signal $e(n)$ constitutes the overall *system output*. From Eq. (41) we see that the noise component in the system output is $v_0(n) - y(n)$. Now, the adaptive filter attempts to minimize the mean-square value (i.e., average power) of the error signal $e(n)$. The information-bearing signal $s(n)$ is essentially unaffected by the adaptive noise canceler.

Hence, minimizing the mean-square value of the error signal $e(n)$ is equivalent to minimizing the mean-square value of the output noise $\nu_0(n) - y(n)$. With the signal $s(n)$ remaining essentially constant, it follows that *the minimization of the mean-square value of the error signal is indeed the same as the maximization of the output signal-to-noise ratio of the system.*

The signal-processing operation described herein has two limiting cases that are noteworthy:

1. The adaptive filtering operation is *perfect* in the sense that

$$y(n) = \nu_0(n)$$

   In this case, the system output is *noise free* and the noise cancelation is perfect. Correspondingly, the output signal-to-noise ratio is infinitely large.
2. The reference signal $\nu_1(n)$ is *completely uncorrelated* with both the signal and noise components of the primary signal $d(n)$; that is,

$$E[d(n)\nu_1(n-k)] = 0 \qquad \text{for all } k$$

   In this case, the adaptive filter "switches itself off," resulting in a zero value for the output $y(n)$. Hence, the adaptive noise canceler has *no* effect on the primary signal $d(n)$, and the output signal-to-noise ratio remains unaltered.

The effective use of adaptive noise canceling therefore requires that we place the reference sensor in the noise field of the primary sensor with two specific objectives in mind. First, the information-bearing signal component of the primary sensor output is *undetectable* in the reference sensor output. Second, the reference sensor output is *highly correlated* with the noise component of the primary sensor output. Moreover, the adaptation of the adjustable filter coefficients must be near optimum.

In the remainder of this subsection, we describe three useful applications of the adaptive noise-canceling operation:

1. *Canceling 60-Hz interference in electrocardiography.* In *electrocardiography (ECG)*, commonly used to monitor heart patients, an *electrical discharge* radiates energy through a human *tissue* and the resulting output is received by an *electrode*. The electrode is usually positioned in such a way that the received energy is maximized. Typically, however, the electrical discharge involves very low potentials. Correspondingly, the received energy is very small. Hence extra care has to be exercised in minimizing signal degradation due to external *interference*. By far, the strongest form of interference is that of a 60-Hz periodic waveform picked up by the receiving electrode (acting like an antenna) from nearby electrical equipment (Huhta and Webster, 1973). Needless to say, this interference has undesirable effects in the interpretation of electrocardiograms. Widrow et al. ((1975b) have demonstrated the use of adaptive noise canceling (based on the LMS algorithm) as a method for reducing this form of interference. Specifically, the primary signal is taken from the ECG preamplifier, and the reference signal is taken from a wall outlet with proper attenuation. Figure 25 shows a block diagram of the adaptive noise canceler used

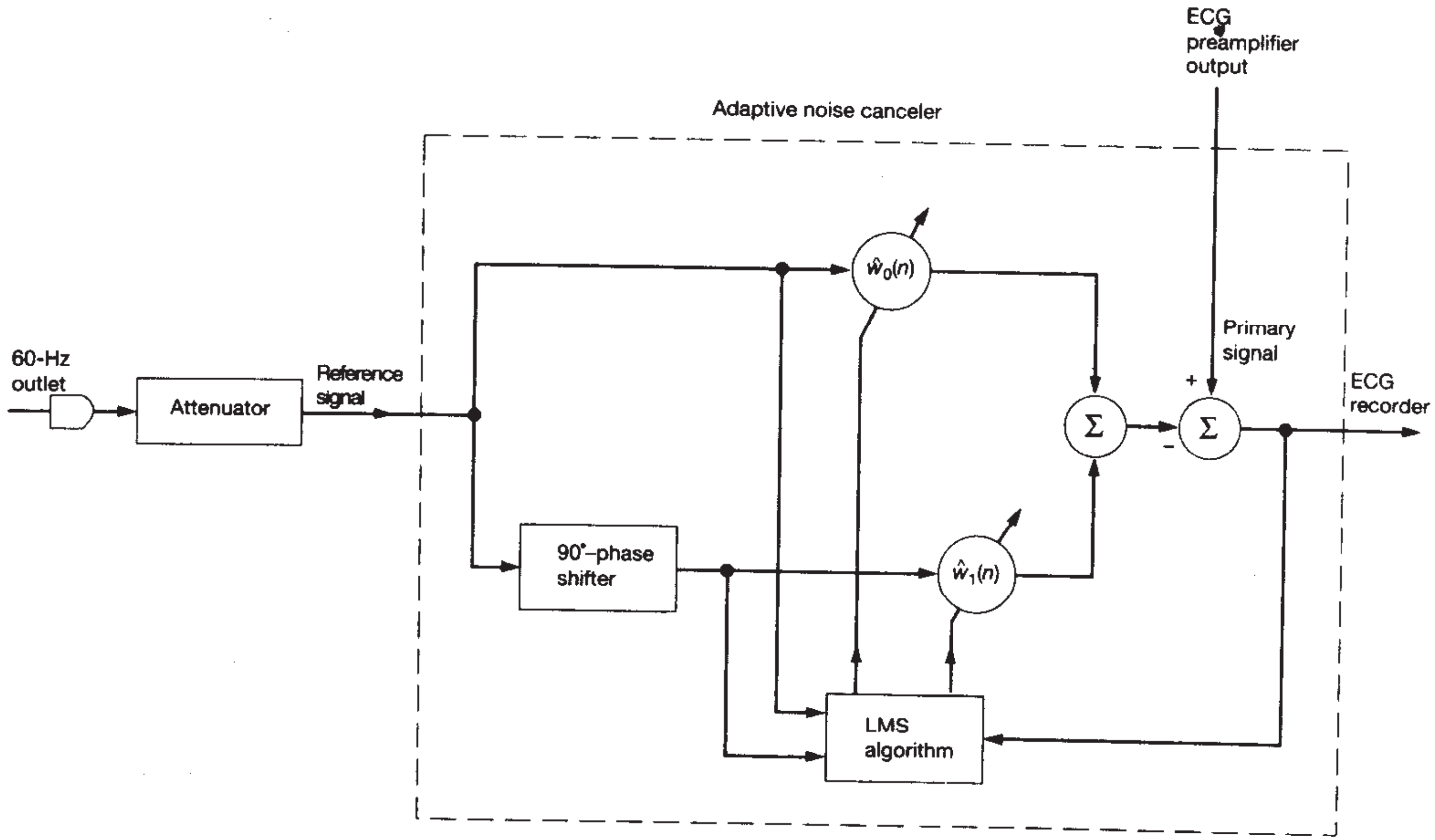

Figure 25 Adaptive noise canceler for suppressing 60-Hz interference in electrocardiography (After Widrow et al., 1975b).

by Widrow et al. (1975b). The adaptive filter has two adjustable weights, $\hat{w}_0(n)$ and $\hat{w}_1(n)$. One weight, $\hat{w}_0(n)$, is fed directly from the reference point. The second weight, $\hat{w}_1(n)$, is fed from a 90°-phase-shifted version of the reference input. The sum of the two weighted versions of the reference signal is then subtracted from the ECG output to produce an error signal. This error signal together with the weighted inputs are applied to the LMS algorithm, which, in turn, controls the adjustments applied to the two weights. In this application, the adaptive noise canceler acts as a variable "notch filter." The frequency of the sinusoidal interference in the ECG output is presumably the same as that of the sinusoidal reference signal. However, the amplitude and phase of the sinusoidal interference in the ECG output are unknown. The two weights $\hat{w}_0(n)$ and $\hat{w}_1(n)$ provide the two *degrees of freedom* required to control the amplitude and phase of the sinusoidal reference signal so as to cancel the 60-Hz interference contained in the ECG output.

2. *Reduction of acoustic noise in speech.* At a noisy site (e.g., the cockpit of a military aircraft), voice communication is affected by the presence of *acoustic noise*. This effect is particularly serious when linear predictive coding (LPC) is used for the digital representation of voice signals at low bit rates; LPC was discussed earlier. To be specific, high-frequency acoustic noise severely affects the estimated LPC spectrum in both the low- and high-frequency regions. Consequently, the intelligibility of digitized speech using LPC often falls below the minimum acceptable level. Kang and Fransen (1987) describe the use of an adaptive noise canceler, based on the LMS algorithm, for reducing acoustic noise in speech. The noise-corrupted speech is used as the primary signal. To provide the reference signal (noise only), a reference microphone is placed in a location where there is sufficient isolation from the source of speech (i.e., the known location of the speaker's mouth). In the experiments described by Kang and Fransen, a reduction of 10 to 15 dB in the acoustic noise floor is achieved, without degrading voice quality. Such a level of noise reduction is significant in improving voice quality, which may be unacceptable otherwise.

3. *Adaptive speech enhancement.* Consider the situation depicted in Fig. 26. The requirement is to listen to the voice of the desired speaker in the presence of background noise, which may be satisfied through the use of adaptive noise canceling. Specifically, *reference microphones* are added at locations far enough away from the desired speaker such that their outputs contain *only* noise. As indicated in Fig. 26, a weighted sum of the auxiliary microphone outputs is subtracted from the output of the desired speech-containing microphone, and an adaptive filtering algorithm (e.g., the LMS algorithm) is used to adjust the weights so as to minimize the average output power. A useful application of the idea described herein is in the adaptive noise cancelation for hearing aids[12] (Chazan et al., 1988). The so-called "cocktail party effect" severely limits the usefulness of hearing aids. The cocktail party phenomenon refers to the ability of a person with normal hearing to focus on a conversation taking place at a distant location in a crowded room. This ability

[12]This idea is similar to that of adaptive spatial filtering in the context of antennas, which is considered later in this section.

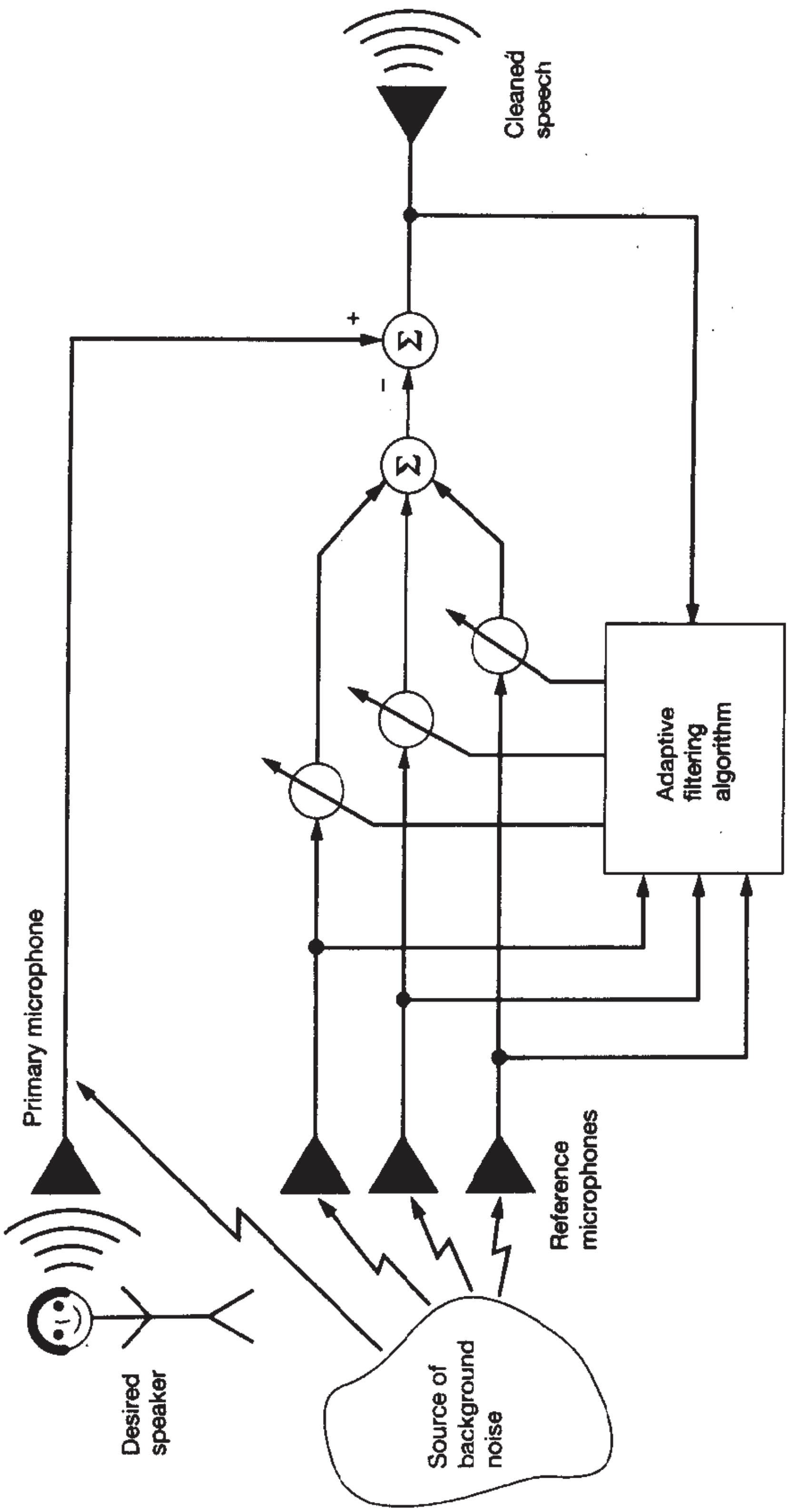

Figure 26 Block diagram of an adaptive noise canceler for speech.

is lacking in a person who wears hearing aids, because of extreme sensitivity to the presence of *background noise*. This sensitivity is attributed to two factors: (a) the loss of directional cues, and (b) the limited channel capacity of the ear caused by the reduction in both dynamic range and frequency response. Chazan et al. (1988) describe an adaptive noise canceling technique aimed at overcoming this problem. The technique involves the use of an *array of microphones* that exploit the difference in spatial characteristics between the desired signal and the noise in a crowded room. The approach taken by Chazan et al. is based on the fact that each microphone output may be viewed as the sum of the signals produced by the individual speakers engaged in conversations in the room. Each signal contribution in a particular microphone output is essentially the result of a speaker's speech signal having passed through the *room filter*. In other words, each speaker (including the desired speaker) produces a signal at the microphone output that is the sum of the direct transmission of his or her speech signal and its reflections from the walls of the room. The requirement is to reconstruct the desired speaker signal, including its room reverberations, while canceling out the source of noise. In general, the transformation undergone by the speech signal from the desired speaker is not known. Also, the characteristics of the background noise are variable. We thus have a signal-processing problem for which adaptive noise canceling offers a feasible solution.

Echo Cancelation

Almost all conversations are conducted in the presence of *echoes*. An echo may be nonnoticeable or distinct, depending on the time delay involved. If the delay between the speech and the echo is short, the echo is not noticeable but perceived as a form of spectral distortion or reverberation. If, on the other hand, the delay exceeds a few tens of milliseconds, the echo is distinctly noticeable. Distinct echoes are annoying.

Echoes may also be experienced on a telephone circuit (Sondhi and Berkley, 1980). When a speech signal encounters an *impedance mismatch* at any point on a telephone circuit, a portion of that signal is reflected (returned) as an echo. An echo represents an *impairment* that can be annoying subjectively as the more obvious impairments of low volume and noise.

To see how echoes occur, consider a long-distance telephone circuit depicted in Fig. 27. Every telephone set in a given geographical area is connected to a central office by a *two-wire line* called the *customer loop*; the two-wire line serves the need for communications in either direction. However, for circuits longer than about 35 miles, a separate path is necessary for each direction of transmission. Accordingly, there has to be provision for connecting the two-wire circuit to the four-wire circuit. This connection is accomplished by means of a *hybrid transformer*, commonly referred to as a *hybrid*. Basically, a hybrid is a bridge circuit with three ports (terminal pairs), as depicted in Fig. 28. If the bridge is *not* perfectly balanced, the "in" port of the hybrid becomes coupled to the "out" port, thereby giving rise to an echo.

Echoes are noticeable when a long-distance call is made on a telephone circuit, particularly one that includes a *geostationary satellite*. Due to the high altitude of such a satellite, there is a one-way travel time of about 300 ms between a ground station and the