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

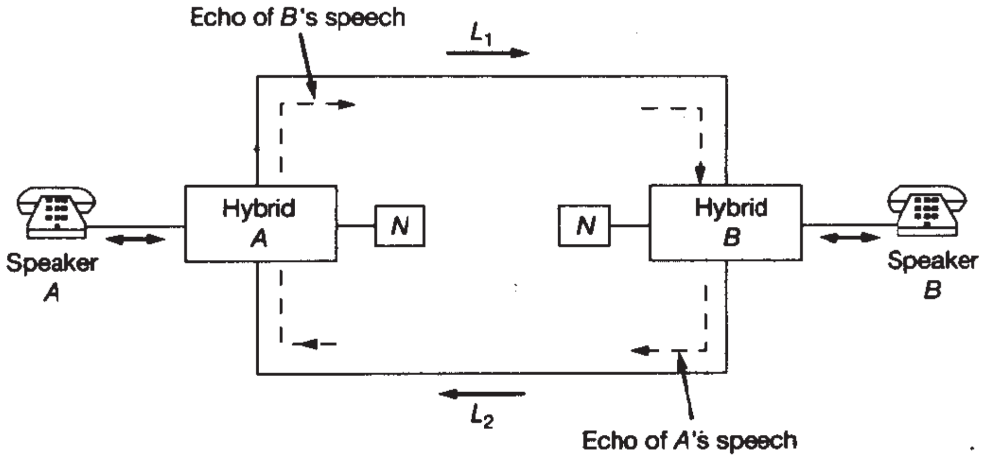

Figure 27 Long-distance telephone circuit; the boxes marked N are balancing impedances.

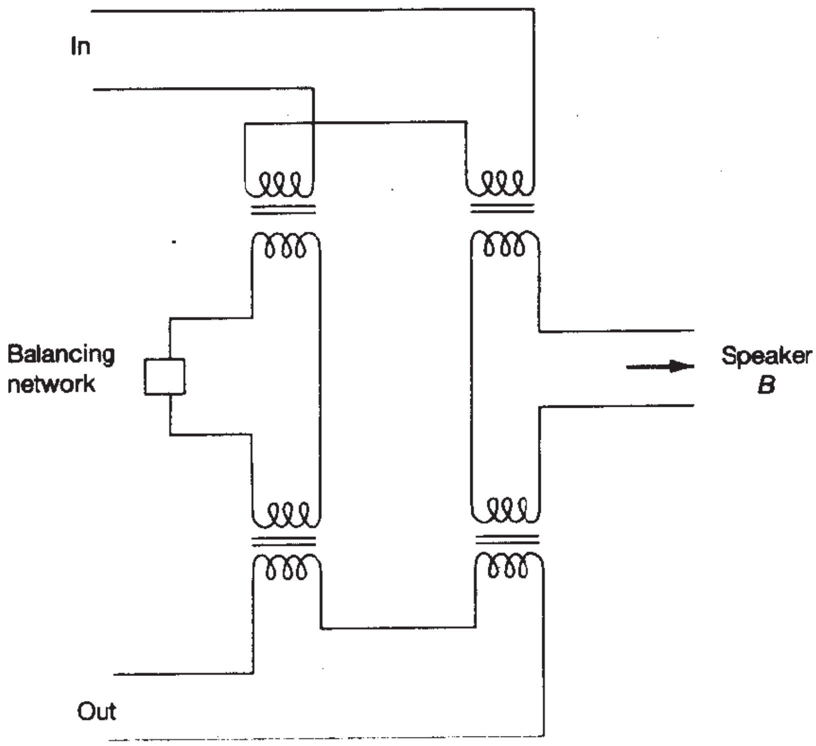

Figure 28 Hybrid circuit.

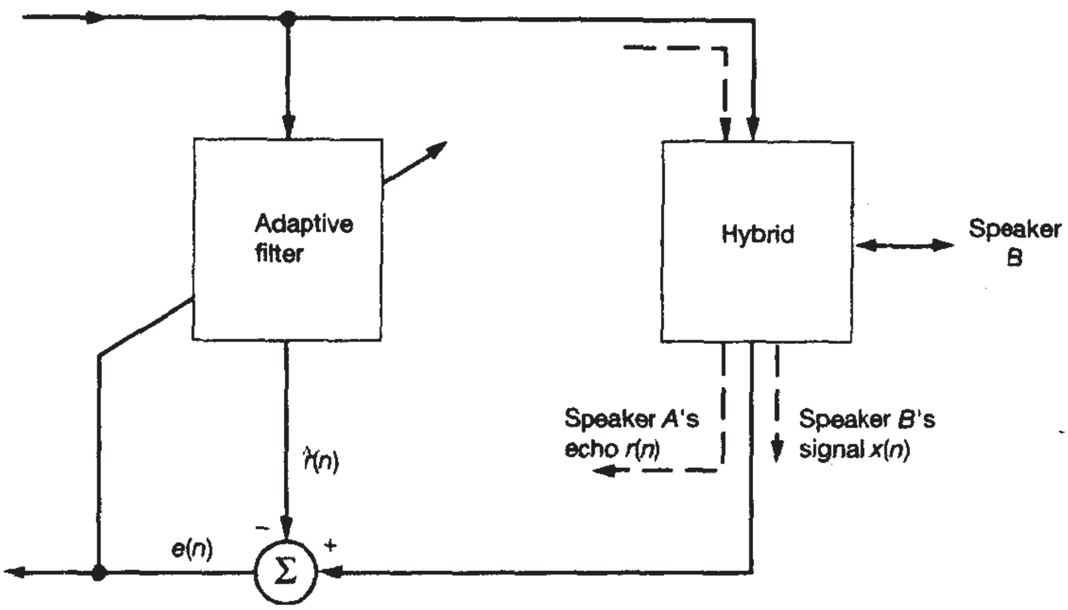

Figure 29 *Signal definitions for echo cancelation.*

satellite. Thus, the *round-trip delay* in a satellite link (including telephone circuits) can be as long as 600 ms. Generally speaking, the longer the echo delay, the more it must be attenuated before it becomes noticeable.

The question to be answered is: How do we exercise echo control? It appears that the idea with the greatest potential for echo control is that of *adaptive echo cancelation* (Sondhi and Prasti, 1966; Sondhi, 1967; Sondhi and Berkley, 1980; Messerschmitt, 1984; Murano et al., 1990). The basic principle of echo cancelation is to *synthesize a replica of the echo and subtract it from the returned signal.* This principle is illustrated in Fig. 29 for only one direction of transmission (from speaker A on the *far* left of the hybrid to speaker B on the right). The adaptive canceler is placed in the four-wire path *near* the origin of the echo. The synthetic echo, denoted by $\hat{r}(n)$, is generated by passing the speech signal from speaker A (i.e., the "reference" signal for the adaptive canceler) through an adaptive filter that ideally matches the transfer function of the echo path. The reference signal, passing through the hybrid, results in the echo signal $r(n)$. This echo, together with a near-end talker signal $x(n)$ (i.e., the speech signal from speaker B) constitutes the "desired" response for the adaptive canceler. The synthetic echo $\hat{r}(n)$ is subtracted from the desired response $r(n) + x(n)$ to yield the canceler error signal

$$e(n) = r(n) - \hat{r}(n) + x(n) \tag{42}$$

Note that the error signal $e(n)$ also contains the near-end talker signal $x(n)$. In any event, the error signal $e(n)$ is used to control the adjustments made in the coefficients (tap weights) of the adaptive filter. In practice, the echo path is highly variable, depending on the distance to the hybrid, the characteristics of the two-wire circuit, and so on. These variations are taken care of by the adaptive control loop built into the canceler. The control loop continuously adapts the filter coefficients to take care of fluctuations in the echo path.

For the adaptive echo cancelation circuit to operate satisfactorily, the impulse response of the adaptive filter should have a length greater than the longest echo path that needs to be accommodated. Let T_s be the sampling period of the digitized speech signal, M be the number of adjustable coefficients (tap weights) in the adaptive filter, and τ be the longest echo delay to be accommodated. We must then choose

$$MT_s > \tau \tag{43}$$

As mentioned previously (when discussing adaptive differential pulse-code modulation), the sampling rate for speech signals on the telephone network is conservatively chosen as 8 kHz, that is,

$$T_s = 125\ \mu\text{s}$$

Suppose, for example, that the echo delay $\tau = 30$ ms. Then we must choose

$$M > 240 \text{ taps}$$

Thus, the use of an echo canceler with $M = 256$ taps, say, is satisfactory for this situation.

Adaptive Beamforming

For our last application, we describe a *spatial* form of adaptive signal processing that finds practical use in radar, sonar, communications, geophysical exploration, astrophysical exploration, and biomedical signal processing.

In the particular type of spatial filtering of interest to us in this book, a number of independent *sensors* are placed at different points in space to "listen" to the received signal. In effect, the sensors provide a means of *sampling* the received signal *in space*. The set of sensor outputs collected at a particular instant of time constitutes a *snapshot*. Thus, a snapshot of data in spatial filtering (for the case when the sensors lie uniformly on a straight line) plays a role analogous to that of a set of consecutive tap inputs that exist in a transversal filter at a particular instant of time.[13]

In radar, the sensors consist of antenna elements (e.g., dipoles, horns, slotted waveguides) that respond to incident electromagnetic waves. In sonar, the sensors consist of hydrophones designed to respond to acoustic waves. In any event, spatial filtering, known as *beamforming*, is used in these systems to distinguish between the spatial properties of signal and noise. The device used to do the beamforming is called a *beamformer*. The term "beamformer" is derived from the fact that the early forms of antennas (spatial filters) were designed to form *pencil beams*, so as to receive a signal radiating from a specific direction and attenuate signals radiating from other directions of no interest (Van Veen and Buckley, 1988). Note that the beamforming applies to the radiation (transmission) or reception of energy.

[13]For a discussion of the analogies between time- and space-domain forms of signal processing, see Bracewell (1986) and Van Veen and Buckley (1988).