Noise Reduction in Speech Applications

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Noise Canceling Headsets for Speech Communication

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Introduction

Headsets for speech communication are used in a wide range of applications. The basic idea is to allow hands-free speech communication, leaving both hands available for other tasks. One typical headset application is aircraft pilot communication. The pilot must be able to communicate with personnel on the ground and at the same time use both hands to control the aircraft.

A communication headset usually consists of a pair of headphones and a microphone attached with an adjustable boom. Headphone design varies considerably between different manufacturers and models. In its simplest

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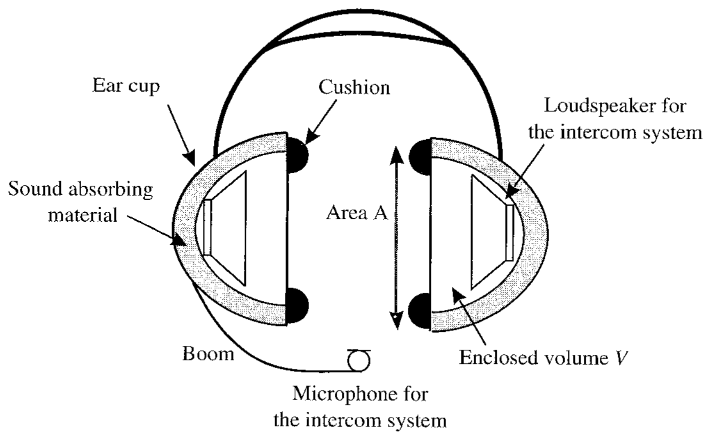


FIGURE 12.1
A closed-back passive headset for two-way communication.

where $p_i(f)$ is the sound pressure inside the ear cup, $p_e(f)$ is the external sound pressure acting on the ear cup, M is the mass of the rigid ear cup, R is the damping in the cushion, and K_c is the mechanical stiffness of the cushion. The mechanical stiffness of the air inside the headset K_a is given by:

$$K_a = \frac{A^2 c_0^2 \rho_0}{V} \quad [N/m] \tag{12.2}$$

Here c_0 and ρ_0 are the speed of sound in air and the density of air at normal temperature and pressure, respectively. A is the area of the plane surface enclosed by the external curvature of the headset cup where it is attached to the cushion and V is the volume of air enclosed by the headset cup. Based on parameters originating from Shaw et al.,[3] the frequency response of the transmission ratios, $p_i(f)/p_e(f)$, for two well-designed passive headsets with different values of cushion damping have been calculated and are shown in Figure 12.2.

A passive headset produces good attenuation of noise at frequencies above its eigenfrequency, as Figure 12.2 illustrates. To maximize the attenuation of noise at lower frequencies, the parameters of the headset need to be chosen appropriately. From Equation (12.1), it follows that the transmission ratio below the eigenfrequency is given approximately by $K_a/(K_a + K_c)$. Hence to reduce transmission of noise at low frequencies, the mechanical stiffness of the air inside the headset K_a should be small, while the mechanical stiffness of the cushion K_c should be large. From Equation (12.2), it follows that small K_a can be achieved by increasing the air volume inside the headset cup and decreasing the area A, of the plane surface enclosed by its external curvature. Both approaches to reducing K_a, however, have practical limits. Since a

Active Noise Control Headsets

As discussed above, bulky headsets are required in order to attenuate low-frequency noise through traditional passive methods.[3] Active approaches can complement these passive methods, and by combining these two approaches, high noise attenuation over a wide frequency range is made possible. Indeed, attenuation can be achieved over the entire audible frequency range from 30 Hz up to 20 kHz.[4,5]

This section discusses active control techniques for headset applications. Both analog and digital controllers are introduced as well as their combination. Analog feedback controllers are covered first, followed by a discussion of digital control algorithms. Finally, the combination of analog feedback controllers and digital controllers, which may be of either feedback or feed-forward type, are discussed.

Active noise control is based on the principle of destructive interference between two sound fields, one sound field originating from the primary noise source, e.g., an engine, and the other generated by a secondary sound source such as a loudspeaker.[6] The loudspeaker produces a sound field of equal amplitude and opposite phase — 180° out of phase — to the unwanted sound field. The accuracy of the amplitude and phase of the generated sound field, the antisound, determine the noise attenuation achievable.

Active noise control works best on low-frequency sounds where the acoustic wavelengths are large compared to the space in which the noise is to be attenuated. In such a case, the antisound is approximately 180° out of phase in the whole space.[6] In general, the closed cavity within the ear cup of a headset and the eardrum is small compared to the wavelengths of sounds for which passive noise cancellation is poor and active techniques are of interest. Using such active control methods, attenuation of noise at frequencies below approximately 1 kHz by up to 20 dB has been achieved.[4,5] Such active control systems have been based on analog and/or digital techniques,[4,6] and both approaches are discussed in the following sections.

Analog Active Noise Control Headsets

Today, most commercial active headsets are based on analog feedback control technology. This type of headset typically includes a loudspeaker, an error microphone, and an analog control unit. The error microphone is generally placed as close as possible to the ear canal, since the objective of the active control is principally to minimize the perceived sound pressure.

The sound pressure under control, $p_c(f)$, inside an analog hearing protector can be written as:

$$p_c(f) = \frac{p_i(f)}{1 + KC(f)H(f)} \tag{12.3}$$

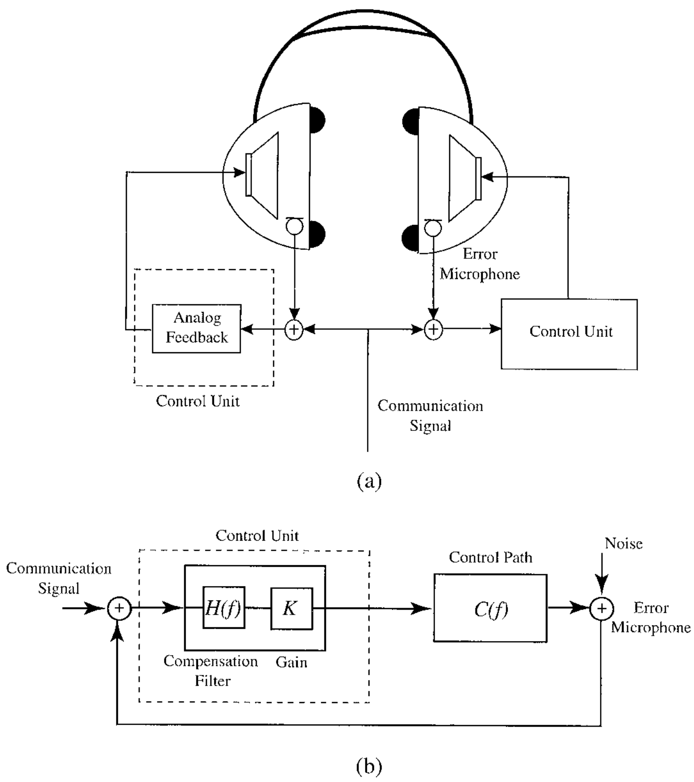


FIGURE 12.3
(a) Analog active noise control headset for two-way communication and (b) the corresponding block diagram of the feedback control system with communication signal injection.

Here $p_s(f)$ is the sound pressure due to the communication signal, $p_i(f)$ is the sound pressure inside the analog hearing protector without control, K is the amplifier gain, $H(f)$ is the frequency function of the compensation filter, and $C(f)$ is the frequency function of the control path. If the amplifier gain K is chosen to be large to produce a large loop gain, $|KC(f)H(f)| >> 1$, the sound pressure under control, $p_c(f)$, is given approximately by:

$$p_c(f) \approx \varepsilon + p_s(f), \quad \text{where } |\varepsilon| << 1 \tag{12.5}$$

Hence, the influence of the feedback control on the communication signal $p_s(f)$ is reduced significantly, and, in addition, the distortion generally introduced in this signal by filtering caused by the loudspeaker and headset cavity is also reduced.

Closed-back headsets can be uncomfortable to wear, especially if the requirements are such that they have to be worn continuously for considerable periods of time. For such a headset, heat building up in the acoustic reference signal can be provided to the digital controller by a microphone mounted on the exterior of the ear cup. For reducing tonal noise, e.g., generated by engines and propellers, the reference microphone can be replaced by a nonacoustic reference sensor, e.g., a tachometer or an optical or inductive sensor.[11,12] In this case, the periodic reference signals can be produced internally within the digital controller by using the output signal from the nonacoustic reference sensor.

There are several advantages in using nonacoustic sensors. For example, the reference signals based on such sensors will contain only the tonal components that are desired to be controlled and the properties of the reference signals, i.e., frequency and signal power, are known. With reference signals generated in this manner, the adaptive control becomes extremely selective. It is possible to determine which frequencies are to be controlled and which are not. Compared with a reference microphone, a nonacoustic sensor usually results in a reference signal with a lower noise level, resulting in higher performance.[6,11] In addition, undesired acoustic feedback from the loudspeaker to the reference sensor, which can cause instability, is also eliminated. In this case, the controller is purely feedforward, i.e., its performance is completely unaffected by the action of the loudspeaker.

Acoustic feedback, experienced when using a reference microphone, can be reduced by electronic techniques. However, this approach requires the control system to have a complicated structure in order to compensate for the acoustic feedback.[6,11] The acoustic feedback problem is particularly important for open-back headsets, where significant coupling between the loudspeaker and the reference microphone is likely to be present. For stationary noise, i.e., noise whose statistical properties are time invariant, a convenient estimate of the maximum noise suppression achievable by an active feedforward noise control system in decibels (dB) is given by $-10\log_{10}(1-\gamma_{xd}^2(f))$,[6,11] where $\gamma_{xd}^2(f)$ is the coherence function[6] between the uncontrolled noise and the reference signal, which is a measure of the linear relationship between them. For a coherence $\gamma_{xd}^2(f)$ of 0.99, an attenuation of potentially 20 dB may be achieved.

In broadband active feedforward control it is important that the causality condition is fulfilled, i.e., the delay introduced by the controller plus the control path does not exceed the acoustic delay from the reference microphone to the error microphone.[6,11] Consequently, in a headset application, feedforward control of broadband noise requires that the reference microphone be positioned such that it picks up the acoustic noise sufficiently in advance of its arrival at the ear to allow time for the processing of the noise signal and for it to be fed to the loudspeaker.

To fulfill the causality condition, a reference microphone mounted on an adjustable boom attached to the headset may be used. By manually tuning the microphone boom to point toward the noise source, the acoustic noise can be arranged to arrive at the error microphone in advance of its arrival at the ear and hence the causality condition to be fulfilled. However, as the headset user moves around, the direction of the microphone boom has to

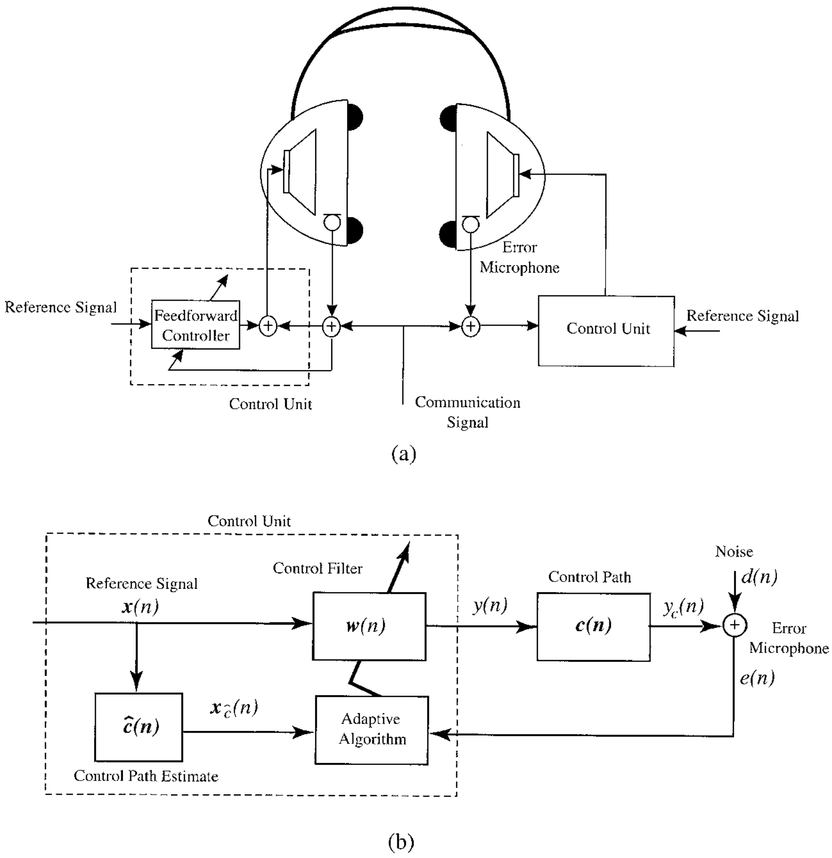


FIGURE 12.4
(a) Feedforward digital active noise control headset for two-way communication and (b) the corresponding block diagram of the adaptive feedforward control system.

of the control path $\hat{c}(i),\ i \in \{0,1,\ldots,I-1\}$ and the product of this filtered reference signal vector and the estimation error usually produces the FXLMS algorithm's gradient estimate, i.e., $x_{\hat{c}}(n)e(n)$. However, the gradient estimate in the FXLMS algorithm's weight vector adjustment algorithm is by definition based on a filtered reference signal that is produced by filtering the reference signal $x(n)$ with the actual impulse response of the control path.[6,11] As a result, the filtered reference signal vector will be an approximation, and differences between the estimate of the control path and the true control path influence both the stability properties and the convergence rate of the algorithm.[6,11] Differences between the estimate of the control path and the actual control path will influence the gradient estimate used in the algorithm, i.e., $x_{\hat{c}}(n)e(n)$, and this will cause the algorithm to adjust its coefficient vector in a direction that is biased compared with the direction of steepest descent.[6,11]

$$w(n+1) = \gamma w(n) - \mu x_{\hat{C}}(n)e(n) \tag{12.11}$$

where γ is a real positive leakage factor, $0 < \gamma < 1$.

Some applications of the FXLMS algorithm require long-control FIR filters, for example, where tonal components that are close in frequency are to be controlled. Long-control FIR filters result in slow convergence of the adaptive algorithm and a considerable computational burden.[11] To improve the convergence speed and reduce the computational burden in such narrowband control applications, the complex FXLMS algorithm described below may be used.

The capacity of an adaptive control system to handle tonal components that are close in frequency depends on the structure of the controller, i.e., how the multiple frequencies are processed. Suitable controllers are generally based on either a single-filter structure or a parallel-filter structure using several filters.[11] The single-filter structure is based on a composite reference signal containing all frequencies to be controlled. For tonal components that are close in frequency, a long-control FIR filter is required resulting in slow convergence of the adaptive algorithm as mentioned above.[11] For the parallel-filter structure, each frequency component is individually processed. This enables shorter filters, and thereby better convergence performance, to be achieved. If possible, therefore, the parallel structure rather than the single-filter structure should be used to achieve efficient and robust control of frequencies that are close together.[12] An example of such a noise field is the beating sound produced by propellers rotating at slightly difference speeds.

An alternative approach to the FIR-based control system for controlling tonal noise is a system based on complex arithmetic.[12] Here each frequency is controlled by an adaptive complex weight. The complex FXLMS algorithm is based on a recursive weight adjustment which is made for each tone required to be controlled, i.e., for each frequency f to be controlled the complex adaptive weight $w_f(n)$ is updated according to[12]

$$w_f(n+1) = w_f(n) - \mu_f x_f^*(n)\hat{C}_f^* e(n) \tag{12.12}$$

where $x_f(n)$ is a complex scalar reference signal at the frequency f, $\hat{C}_f$ is a complex control path estimate corresponding to the frequency f, $(\cdot)^*$ denotes the complex conjugate, μ_f is the adaptation step size at the frequency f, and $e(n)$ is the broadband error microphone signal.[11,12] The output signal from the parallel adaptive filter is produced by $y(n) = \sum_{f \in F} \mathcal{R}\{w_f(n)x_f(n)\}$, where $\boldsymbol{F}$ is the set of controlled frequencies and $\mathcal{R}\{\cdot\}$ denotes the real part of the complex quantity $w_f(n)x_f(n)$. In a practical implementation, $\mathcal{R}\{\cdot\}$ implies that only the real part is evaluated.

For the same reason as in the case of the FXLMS algorithm of reducing susceptibility to reference signal power variations, it can be important to

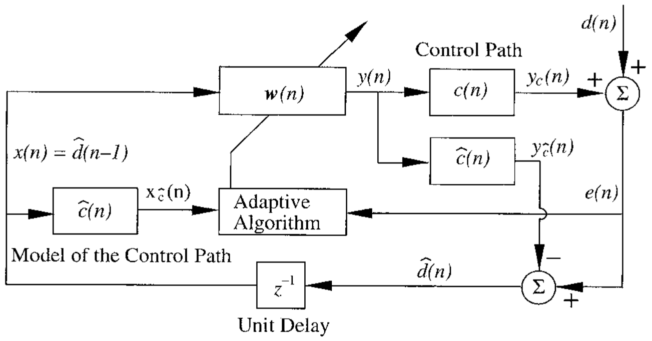


FIGURE 12.5
Block diagram of internal model control (IMC) controller based on an adaptive control FIR filter steered by the FXLMS.

delay comparable to or below that of an analog system makes an analog controller a more cost efficient solution for broadband control problems.[6]

However, in narrowband applications the performance of a feedback controller is less sensitive to delays in the control loop, since narrowband signals exhibit more deterministic behavior. Furthermore, as narrowband noise components may be time variable and are likely to differ between different environments, adaptive control is often required. Consequently, in the case of nonstationary narrowband noise, a digital feedback controller, which can provide adaptive control more easily and cost effectively than an analog controller, is preferable. For example, at the error microphone of an active noise control headset an adaptive digital feedback controller may provide up to 20 dB more attenuation of the narrowband noise than can be achieved with an analog controller.[6] An active noise control headset for use in a variety of environments involving both broadband and narrowband noise is thus likely to be one which involves both an adaptive digital feedback controller and an analog feedback controller.[6]

An adaptive digital feedback controller suitable for use in active noise control headsets is the internal model control (IMC) controller based on an adaptive control FIR filter steered by the FXLMS algorithm.[6] In Figure 12.5, a block diagram of this adaptive IMC controller is shown. The adaptive IMC controller algorithm is obtained by adding the two equations

$$\begin{aligned} \hat{d}(n) &= e(n) - \sum_{i=0}^{I-1} \hat{c}(i)y(n-i) \\ x(n) &= \hat{d}(n-1) \end{aligned} \tag{12.14}$$

to those defining the FXLMS algorithm (see Equation [12.6]). Here, $\hat{d}(n)$ is an estimate of the noise to be controlled, $\hat{c}(i), i \in \{0,1,\ldots,I-1\}$ is an FIR-filter estimate of the control path between the loudspeaker and error microphone, Since the feedback FXLMS algorithm does not rely on cancellation of the feedback path, it might be argued that this algorithm is likely to be favorable compared with the IMC controller in circumstances where the control path shows large variability.

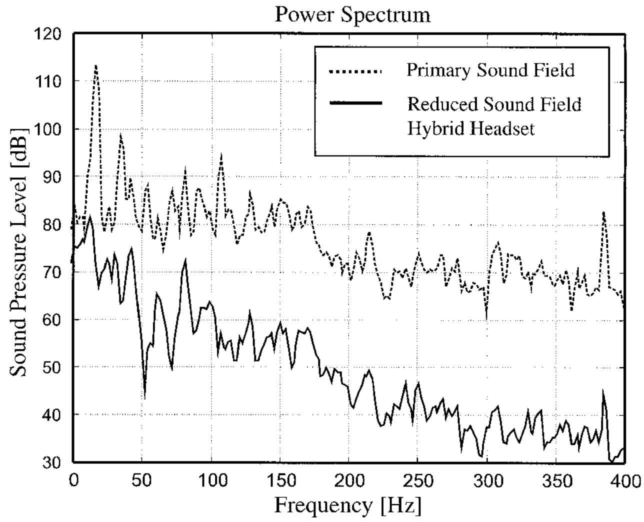
As mentioned in "Analog Active Noise Control Headsets," analog active noise control headsets typically produce an attenuation of about 20 dB at 100 to 200 Hz that falls to zero below approximately 30 Hz and above approximately 1 kHz. Combining such analog feedback control with the digital feedback controllers described above will result in broadband attenuation of the noise due to the former and further reduction of nonstationary narrowband noise components due to the latter. Thus, using digital feedback control in combination with analog feedback control is likely to result in the best overall performance active noise headsets.[6]

In such a combined system, the digital adaptive feedback controller is implemented as an outer control loop of the analog feedback control system.[6] When such a digital feedback loop is used as part of a headset communication system, degradation of the communication signal may occur since the digital feedback controller is likely to affect the communication signal. Selecting the adaptation step size μ sufficiently small, and thereby prohibiting the adaptive filter from tracking the nonstationary speech signal, might allow a communication signal of sufficient quality to be achieved while simultaneously maintaining the attenuation of the narrowband noise.

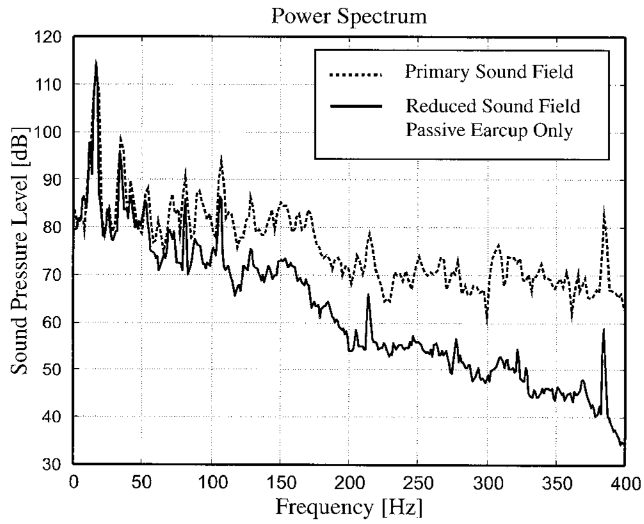
Hybrid Active Noise Control Headsets

An active control system that combines feedforward and feedback techniques is usually called a *hybrid* active control system.[11] Such a system can provide narrowband as well as broadband noise attenuation and can be combined with open- or closed-back headsets. The principle of a hybrid active noise control headset is illustrated in Figure 12.7. Hybrid active noise control can be used to improve the noise attenuation achievable within an environment which has dominant low-frequency tonal noise embedded in broadband noise. Such environmental noise dominates that found in the interior cabins of propeller aircraft and helicopters. Compared with an analog feedback controller giving broadband noise attenuation, an adaptive digital feedforward controller is likely to produce greater attenuation of the nonstationary tonal noise components.

A hybrid system can be based on either digital technology or a combination of analog and digital technologies. The performance of a closed-back hybrid headset based on a nonadaptive analog feedback controller combined with a digital adaptive feedforward controller is shown in Figure 12.8(a).[14] In this figure the dominant low-frequency tones correspond to the fundamental blade passage frequency, i.e., rotational speed of the propeller axis multiplied with the number of the propeller blades, and the harmonics of the main rotor blade passage frequency of the helicopter. If only a passive commercial headset is used, limited low-frequency noise attenuation can be obtained, as



(a)



(b)

FIGURE 12.8
In (a–c), the sound pressure level of the interior helicopter cabin noise — AS332 "Super Puma" MKII helicopter — outside the ear cups is shown with a dashed line. (a) Solid line; reduced sound pressure level inside the ear cups after the *hybrid headset* has been switched on; (b) solid line; reduced sound pressure level inside the ear cups when only *passive* damping is applied.

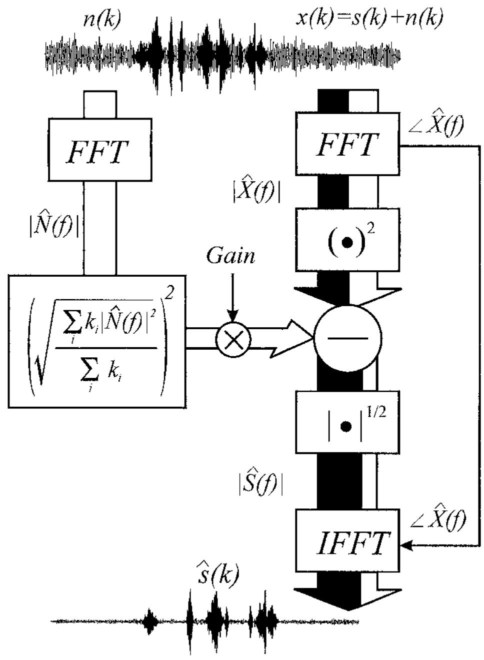


FIGURE 12.9
The spectral subtraction principle. The white parts of the arrows indicate the noise content and the black parts of the arrows indicate the desired speech. The upper waveform shows the original noise sequence and the lower waveform shows the spectral subtraction cleaned sequence. The left branch shows the noise estimation process, and the right branch the spectral subtraction stage.

noise environments, where sound pressure levels are high, alternative microphone techniques can be used to improve performance. One approach is to replace the ordinary communication microphone with a more sensitive microphone mounted in a foam plug that is placed in the auditory canal (auditory meatus). The foam plug itself can provide substantial reduction of the noise picked up by the microphone. Further noise reduction can be achieved by using a miniature active noise control system or by using an additional passive or active headset together with the in-ear microphone. Figure 12.10 shows an active noise control headset that uses such an in-ear microphone.

Many of the techniques for improving noise cancellation with ordinary headsets, passive and active or a combination thereof, which were covered earlier in this chapter, can also be applied to a headset incorporating an in-ear microphone. For example, as in the case of a passive headset, both the choice of sound-absorbing earplug material, as well as the plug-to-auditory-canal fit, are important noise-damping factors. Combining an in-ear microphone with an active noise control headset can be useful in various situations, civil as well as military, in order to enhance the speech intelligibility of the intercom system.[17]

of the headphone. Alternatively, the use of a headset microphone boom with spectral subtraction may be replaced by in-ear microphone techniques.

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