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Use of an adaptive noise canceler as an input preprocessor for a hearing aid*

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Abstract—The use of a two-channel adaptive noise canceler as a preprocessor for a hearing aid was evaluated. An omnidirectional microphone and a directional microphone were used as the inputs to the primary and reference channels, respectively, of the adaptive noise canceler. The microphones were mounted just above one ear on the head of a KEMAR mannikin. The system was found to work well under anechoic conditions, but showed only modest improvements in a moderately reverberant room (reverberation time 380 ms).

INTRODUCTION

The steady improvement in the power and speed of microprocessors, computer memory chips, and miscellaneous integrated digital circuits that has occurred in recent years, combined with the equally steady reduction in the cost of these devices, has made it possible and practical to use sophisticated digital signal processing techniques in small and lightweight consumer products. One potential product, the so-called digital hearing aid, has recently attracted a great deal of interest. Digital techniques can make it possible to shape the frequency transfer function of a hearing aid more easily and with greater accuracy than can be accomplished by use of analog circuits. It is possible to implement digitally controlled amplitude compression systems whose characteristics (e.g., compression ratio, compression threshold, limiting levels) can be adjusted easily. Finally, digital techniques may make it possible to incorporate into hearing aids some powerful methods for attenuating background noises.

A number of sophisticated techniques are available that can be used to enhance the signal-to-noise ratio of speech that has been mixed additively with noise. Some of them, particularly those designed to attenuate narrow-band noises (e.g., tones, whistles, buzzes) or short-duration noises (clicks, transient sounds shorter than 10 ms) can be very effective and can result in substantial improvements in speech intelligibility (4,5). On the other hand, techniques that have been tried for attenuating wideband random noise have proven to be of very limited value at best. However one method that has received a great deal of attention shows considerable promise in this regard. It is usually referred to as adaptive noise canceling.

An adaptive noise canceler is a two-channel system in which information about the noise, in one channel, is used to cancel the noise itself in the other channel. The system exploits the properties of an adaptive filter. Unlike simple filters, adaptive filters are able to modify their own characteristics (e.g., impulse response or, equivalently, frequency transmission function) in response to an applied error signal. Typically, the filter characteristics are controlled by an adaptation algorithm, which seeks

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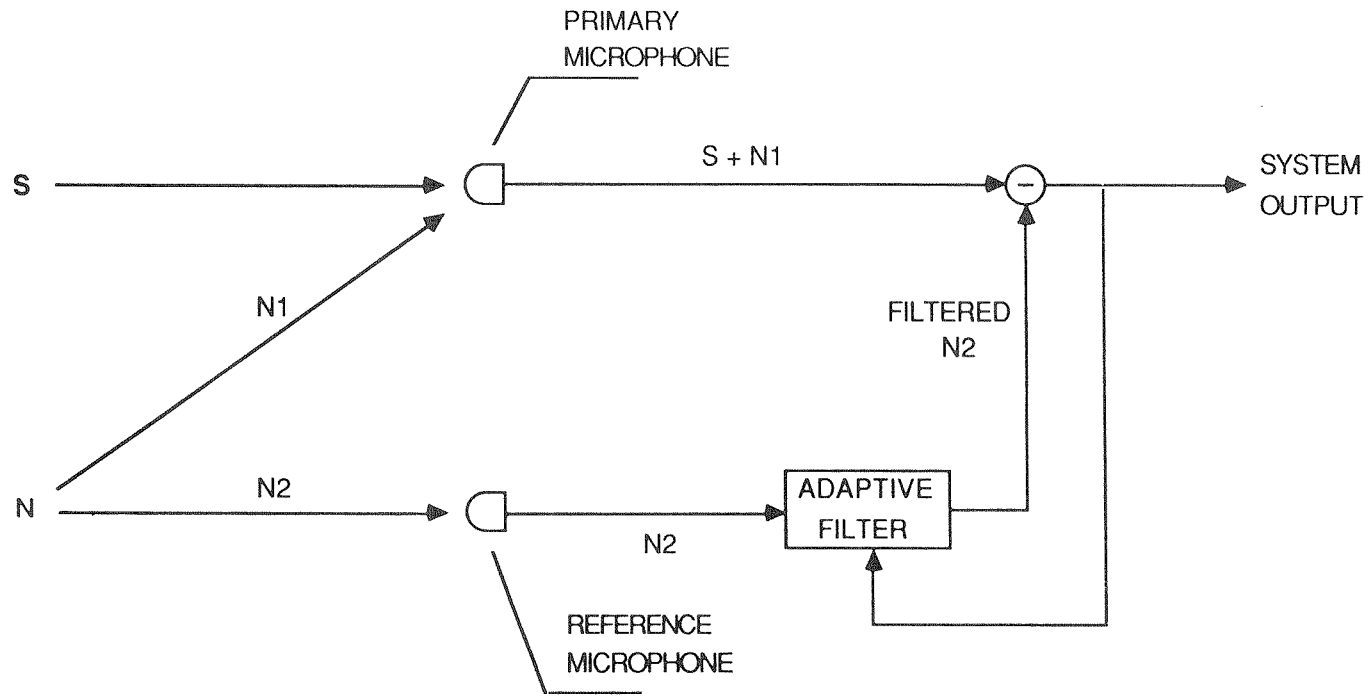

Figure 1.
Basic Adaptive Noise Canceler. The primary microphone receives the speech signal, S, together with one version of the noise signal, N1. The reference microphone receives a second version of the noise signal, N2. The adaptive filter adjusts itself until the system output is minimized, which occurs when N2 filtered is approximately the same as N1.

to minimize the average power of the error signal. A two-channel noise-canceling system that uses an adaptive filter to attenuate additive noise is illustrated in **Figure 1**. Speech signal S and background noise N1 are both received at one microphone, referred to as the primary microphone. A second noise N2, which is correlated with N1 but not with S, is received at a second microphone, referred to as the reference microphone. The noise received by the reference microphone is delivered to the adaptive filter, whose output is subtracted from S and N1 to form the error signal. The filter shapes N2 such that its waveform closely approximates that of N1. It accomplishes this by adjusting its transfer function in response to the applied error signal until the average power in the error signal has been minimized. The error signal is S + N1 − filtered N2. Since S is uncorrelated with both N1 and filtered N2, the power in the error signal will be a minimum when N1 minus filtered N2 is a minimum. This will occur when the filtered reference noise most closely resembles the primary input noise.

Adaptive cancellation of noise offers significant potential advantages over other techniques of signal enhancement. Many of those other techniques cause audible distortion of the enhanced speech signals, and although most of them provide substantial attenuation of wideband random noise (e.g., 6 dB to 12 dB) none of them appear capable of improving the intelligibility of the enhanced speech (3). By contrast, adaptive noise cancelers can achieve substantial attenuation of the background noise with little if any distortion of the speech signal and with a substantial improvement in intelligibility (1,2,6).

Unlike some techniques (e.g., comb filtering, linear predictive coding) adaptive noise cancelers do not require the estimation of speech parameters or of signal coefficients, with their attendant possibility of introducing distortion of speech or loss of speech cues due to estimation errors. Other techniques, such as Weiner filtering, require accurate descriptions of the statistics and/or power spectra of the speech and noise signals, or, in the case of a Weiner filter used in a noise canceler, knowledge of the transfer characteristics of their transmission paths from the sources to the microphones. An adaptive noise canceler does not require such information. Since the adaptive filter adjusts its prop-

erties automatically to minimize the power in the error signal, the filter will automatically acquire a form close to that of an optimum Weiner filter for the applied speech and noise signals (6). For an unconstrained filter, e.g., one whose impulse response can approach infinity in both directions, the filter solution will become the optimum Weiner filter. Because these filters adapt automatically, adaptive noise cancelers are able to continue to function during periods when the transmission paths from the sound sources to the microphones are changing.

Four Factors That Limit Performance—An ideal adaptive noise canceler operating under ideal conditions can provide a completely noise-free output. In practice, however, system performance will be limited by at least four factors. The first factor is the presence of noise in the primary channel that is not correlated with noise in the reference channel. This will occur when noise is generated internally in either channel, or when the noises received at the channel inputs are generated by more than one noise source. Internally generated noise can be reduced to insignificant levels by the use of appropriate techniques in the design and construction of the system. Multiple external noise sources cannot be as easily eliminated or avoided. External noise is seldom limited to one source and often, as in a so-called cocktail party situation, the noise may be generated by a large number of sources.

The second performance-limiting factor is the finite length of the adaptive filter, which, in turn, limits the noise attenuation that can be achieved in reverberant environments. The length of the filter determines the maximum duration of its impulse response. Delayed reflections of the noise sounds that are received at the primary microphone can be attenuated for delays up to the duration of the filter's impulse response. Reflections whose delays are longer cannot be cancelled.

The third factor that can degrade the performance of an adaptive noise canceler is the time taken for the filter to minimize the noise in the error signal. The faster the filter adapts, the lower will be the average level of noise at the output of the adaptive noise canceler for a changing transmission path but constant input noise level. A practical situation of great interest is that in which a noise canceler is used with a wearable hearing aid. In this case, the location of the noise source may not be fixed, nor are the transmission paths from noise source to either the primary or reference microphones likely to remain constant. Any change in these transmission paths will result in an increase in the noise component in the error signal which will persist until the filter has adapted to the new conditions.

The time it takes an adaptive filter to fully adjust to a change in the error signal depends partly on the length of the filter and partly on the adaptation algorithm. High adaptation speeds can be achieved by use of a filter with a short-duration impulse response. However, this may conflict with the need for the filter to have a long-duration impulse response to cancel reflections in a reverberant room. Adaptation speed also depends on the type of algorithm that is used to control the filter's characteristics. The slowest of these, the Widrow-Hoff LMS algorithm, is also one of the simplest to implement and requires the least number of computations per step as the filter seeks to minimize the average power of the error signal (6). The LMS response time for a 60-ms-long filter that is processing signals of 5 kHz bandwidth is typically on the order of several seconds. Faster algorithms are available, but they are much more computationally intensive and consequently are much more difficult to implement in hearing aids small enough to be worn conveniently, and that are constrained to low levels of battery power.

The fourth factor limiting performance is the presence of speech components in the reference input. For an ideal adaptive noise canceler (i.e., one in which the noises in both channels are mutually correlated and the filter solution is unconstrained) the signal-to-noise spectrum density ratio at the system output is equal at all frequencies to the noise-to-signal spectrum density ratio at the reference input (6). Hence it is desirable to minimize the signal-to-noise ratio at the reference input. However, in an application such as a hearing aid, the reference and primary signals will be obtained from microphones that will be close to one another. Consequently, the signal-to-noise ratios at the inputs to both microphones will be comparable, unless special steps are taken to reduce the signal level in the reference input. If this is not done, the performance of the noise canceler will be severely limited.

A simple way of reducing signal-to-noise ratio (i.e., improving noise-to-signal ratio) at the reference microphone is to use a directional microphone. Miniature directional microphones for use in hearing aids are currently available. Most of these exhibit a

cardiod or hypercardiod polar sensitivity pattern. Typically, their front-to-back rejection ratios range from 14 dB to 20 dB, and their rejection ratios at azimuth angles of 90 degrees and 270 degrees range from 5 to 10 dB.

It is of interest to compare the improvement in signal-to-noise ratio obtained with a directional microphone for the following two conditions: 1) that in which the directional microphone is the sole input in a conventional hearing aid; and 2) that in which the directional microphone is used to improve the noise-to-speech ratio at the reference input of a two-channel adaptive noise canceling system. For the directional microphone used as the sole input to a hearing aid, then, with a speech source at zero degrees azimuth, (i.e., in front of the listener) and a single, equally loud noise source at any other angle, the resulting signal-to-noise ratio would typically be as indicated in the left-hand column of **Table 1**. However, if the microphone were used to provide the reference input to an ideal adaptive noise canceler, with the back of the microphone pointed toward the speech source in front of the listener, the resulting speech-to-noise ratio at the system output would be as indicated in the right-hand column of the table. It is seen that for these conditions an ideal adaptive filter would provide only slightly better performance than a simple directional microphone used with a conventional hearing aid. Further, under practical conditions, this small advantage is likely to diminish or disappear.

In view of the foregoing and of the added cost and complexity of an adaptive noise canceler, it would appear that this approach has little to offer. There is, however, a second way in which the effect of speech in the reference input could be minimized.

Table 1. Signal enhancements achievable by use of a directional microphone (input S/N = 0 dB).

Azimuth angle of noise source (degrees)	Output S/N (dB)	
	Microphone only	Microphone plus adaptive filter
0	0	0
30	1	2
60	3	7
90	7	12
120	12	16
150	17	18
180	19	19

This is to permit the filter to adapt to the noise inputs only when speech is absent, and to inhibit or suspend adaptation when speech is present. If this could be done, the filter would provide much greater attenuation of the noise. To minimize any spectral distortion of the speech signal at the system output it would again be desirable to use a directional microphone as input to the reference channel.

EVALUATION OF AN ADAPTIVE NOISE CANCELER

The amount of noise suppression that can be achieved when an adaptive noise canceler is used with a hearing aid was evaluated in a series of tests. The primary objective was to obtain an estimate of the maximum noise attenuation that such a combination could reasonably provide. Since this quality is independent of adaptation speed, the tests were designed to evaluate only the effects of Factors 1, 2 and 4 discussed earlier. The effect of adaptation speed (Factor 3) on the usefulness of a combined hearing aid/noise canceler is currently being examined in a separate experiment. (See paper by Schwander and Levitt in this issue).

To this end, the location of the noise sources and microphones were fixed during each test. The parameters that were varied in these tests were the reverberation time (RT) of the room in which the test sounds were generated, and the number of sources of noise. Two rooms were selected. To permit the effects of more than one noise source to be measured independently of those of reverberation, one of the rooms had an RT of less than 20 ms. Since this room served as a good approximation to an anechoic chamber for the purposes of this experiment, it is referred to as the anechoic room. The other room, a large office with an RT of 380 ms, represented rooms with a moderate level of reverberation. It is referred to here as the reverberant room. The RT of 380 ms was about three times the duration of the longest impulse response that could be simulated by the adaptive filter used in these tests.

Up to three independent noise sources were used. If the speech source used in the tests is viewed as representing a talker to whom the hearing aid user is listening, then the noise sources represent other

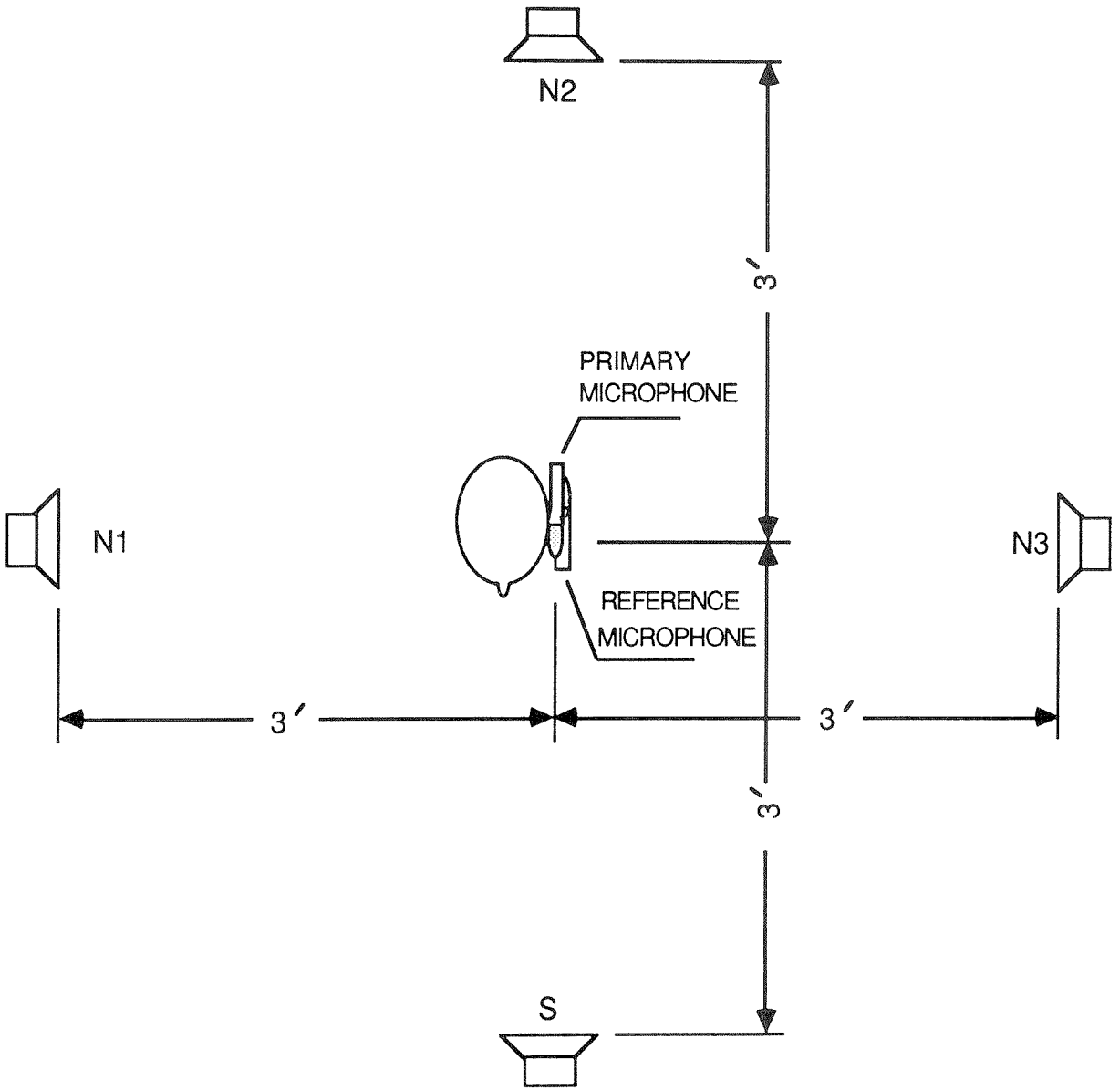

Figure 2.
Arrangement of Sound Sources and Microphones for Test Recordings. The speech source loudspeaker was located at an azimuth of 0 degrees relative to the primary microphone. The noise source loudspeakers were located at 90 degrees, 180 degrees, and 270 degrees azimuth. The primary and reference microphones were mounted above one another and 3 inches apart.

nearby talkers in the same room. While this may not typify a cocktail party situation, it can be a useful approximation of one when the noise sources are close to and distributed around the listener. The presence of three noise sources of equal intensity distributed in this manner constitutes a condition in which a hearing aid user is likely to need the assistance of a noise attenuation system.

The configuration of microphones and sound sources used in each room is illustrated in **Figure 2**. The microphones were mounted just above one ear on the head of a Kemar mannikin with their active elements 3 inches apart. The noise sources were loudspeakers located behind and on each side of the head of the mannikin at a distance of 3 feet from the primary microphone. A fourth loudspeaker, located 3 feet in front of the primary microphone, produced the speech sounds. The outputs of the microphones were recorded during each test. To minimize any distortion of these signals during

Table 2. Signal-to-noise ratio in the reference channel.

Sound source*	Signal-to-noise ratio (dB)	
	Anechoic room	Reverberant room
N1	− 8.5	−4.0
N2	−17.5	−9.5
N3	− 6.0	−1.5

*See **Figure 2** for the locations of sound sources N1, N2, and N3 relative to the microphones. The speech and noise sources were adjusted for equal intensity at the input to the primary microphone.

recording or reproduction, or the addition of extraneous noise, the microphone outputs were sampled and converted to 14-bit digital form and recorded on a wide bandwidth VCR recorder.

An EV model 635A microphone was used to provide input signals to the primary channel. Like a hearing aid microphone, it has an omnidirectional polar sensitivity pattern, but a wider and flatter frequency response and a greater dynamic range. A Beyer model M201N hypercardioid directional microphone provided the inputs to the reference channel. The free-field polar sensitivity pattern of this microphone exhibits a rejection of about 8 dB for signals arriving from either side. For signals arriving from an azimuth of 180 degrees the rejection is at least 13 dB for frequencies below 6,000 Hz and at least 14 dB for frequencies of 500 Hz or less.

The speech signal and each of the noise signals were derived from independent sources of speech-distributed random noise. Wideband random noise was used for these signals, rather than either speech or cafeteria babble, to improve the stability and precision of the measured signal levels. The measurement accuracy was estimated to be 0.5 dB. At the start of the tests in each room, the levels of the signals applied to the four loudspeakers were adjusted to produce approximately equal sound intensity levels at the primary microphone. This was done to compensate for distortion of the omnidirectional character of the primary microphone caused by mounting it on the side of the head of the mannikin. As a result of this procedure, the signal-to-noise ratio in the primary channel was approximately 0 dB when the sound source and any one noise source were active. To maintain a constant signal-to-noise ratio as the number of sound sources was increased, the level of the signals applied to the speech source loudspeaker was increased by 3 dB over the single-noise-source level when any two noise sources were active and by 4.8 dB when all three were active.

Because the primary and reference microphones were only 3 inches apart, the S/N at the input of the reference microphone was also 0 dB. However, due to the microphone's directional charactertistics, the signal-to-noise ratio at the output of the microphone was negative. The signal-to-noise ratios obtained for the reference channel, listed in **Table 2**, establish performance limits that are the best that could be achieved under ideal conditions (i.e., a single source of noise and an unconstrained adaptive filter). Consistent with the estimated measurement accuracy, the values shown in this table and elsewhere are given to the nearest 0.5 dB.

The outputs of the primary and reference microphones were recorded for all possible combinations of the noise sources. Two sequences of recordings were made for each test condition. For the first

Table 3. Attenuation of noise sources in the anechoic room

Sound sources active during adaptation	Noise sources														
	N1					N2					N3				
	Filter length in ms					Filter length in ms					Filter length in ms				
	1	10	20	40	60	1	10	20	40	60	1	10	20	40	60
1 Noise	7.5	11.0	15.5	22.5	23.5	9.5	20.5	29.0	32.0	33.0	7.5	15.5	20.5	28.5	32.0
2 Noise	3.0	4.5	4.0	4.0	4.0	5.5	12.5	12.0	10.5	10.0	0	1.5	0.5	0	0
3 Noise	2.5	3.5	3.5	3.0	3.0	5.0	10.5	10.5	9.0	8.0	0	0	0	0	0
1 Noise & speech	7.0	7.5	7.5	7.0	6.5	7.5	14.0	14.5	12.0	10.5	4.0	4.5	4.5	4.5	4.0
2 Noise & speech	3.0	4.5	4.0	4.0	4.0	5.0	10.5	10.5	9.5	8.5	0	0	0	0	0
3 Noise & speech	3.0	3.0	3.0	3.0	3.0	5.0	8.0	6.0	5.5	5.0	0	0	0	0	0

Table 4. Attenuation of noise sources in the reverberant room

Sound sources active during adaptation	Noise sources														
	N1					N2					N3				
	Filter length in ms					Filter length in ms					Filter length in ms				
	1	20	60	100	140	1	20	60	100	140	1	20	60	100	140
1 Noise	1.5	2.5	3.5	4.0	5.5	3.5	4.5	6.5	9.5	13.5	1.0	1.5	2.0	2.0	2.5
2 Noise	1.5	1.5	1.5	2.0	2.0	3.0	3.0	5.0	5.0	6.5	0	0	0	0	0
3 Noise	1.0	1.5	1.5	1.5	1.5	3.0	3.5	4.0	3.5	4.5	0	0	0	0	0
1 Noise & speech	1.0	2.5	2.5	3.0	3.0	3.0	3.5	6.0	5.0	6.5	0.5	1.0	0.5	0	0
2 Noise & speech	1.0	1.5	1.0	1.0	1.5	2.5	2.5	3.0	3.0	3.0	0	0	0	0	0
3 Noise & speech	1.0	1.0	1.0	1.0	1.0	2.5	2.0	1.5	1.0	1.5	0	0	0	0	0

sequence, the first recording consisted of 15 seconds of the speech source plus either a single noise source or of a combination of noise sources used in the test condition. This recording was used to train the adaptive filter. It was followed by 10-second-long recordings of each of the noise sources that were active during the first recording. These were recorded first one at a time and then in the same combination that was used for the first recording. Finally, 10 seconds of the speech source alone were recorded. The second sequence was the same as the first with one difference: the sounds recorded during the first 15-second segment did not include the speech source. This was done to make it possible to measure how much the presence of speech in the reference input affected the performance of the adaptive noise canceler under the various test conditions.

The reproduced signals were applied to the primary and reference inputs of a MASP model 404 adaptive noise canceler. This instrument, which is manufactured by Adaptive Digital Systems, implements the Widrow-Hoff LMS algorithm. The operation of the MASP can be controlled by switches on its front panel. One of these enables or disables the noise canceler mode of operation. Another one permits the user to freeze the filter (that is, to stop the adaptive process) or, alternatively, to allow adaptation to proceed. Other switches are used to select the sampling rate, which was set to 12,000 Hz for these tests. Anti-aliasing filters limited the signal bandwidth to 5 kHz. For the selected sampling rate, the MASP was able to maintain real time operations on the input signals for adaptive filter lengths up to 140 ms.

TEST RESULTS AND DISCUSSION

The following procedure was used to measure noise attenuation for each sequence of test recordings. First, the filter in the noise canceler was allowed to adapt for 14 seconds during the reproduction of the first recording in the test sequence. At the end of this period, the filter characteristics were frozen. The length of this adaptation period assured that, for all test conditions, the filter had converged on its final setting before adaptation was suspended. Then, for each 10-second-long subsequent recording in the sequence, the dB level of the signal at the output of the noise canceler was measured both with and without the cancellation mode enabled. The difference between the canceled and the non-canceled output levels represented the noise attenuation of the applied test signals.

The results of the attenuation measurements are shown in **Tables 3 and 4**. All measurements were obtained with a precision of 0.5 dB. Values shown in the first row of each table are for tests in which the indicated noise source (N1, N2, or N3) is the only one active during the adaptation of the filter. The second row of each table shows the average attenuation for those in which the indicated source and one of the other noise sources were concurrently active during adaptation. The third row shows the attenuation of the indicated source when all three noise sources were active during adaptation. The fourth, fifth, and sixth rows are similar to the first three with the addition of the speech source active during adaptation.

It is apparent that the attenuation of each source was at a maximum when that source was the only

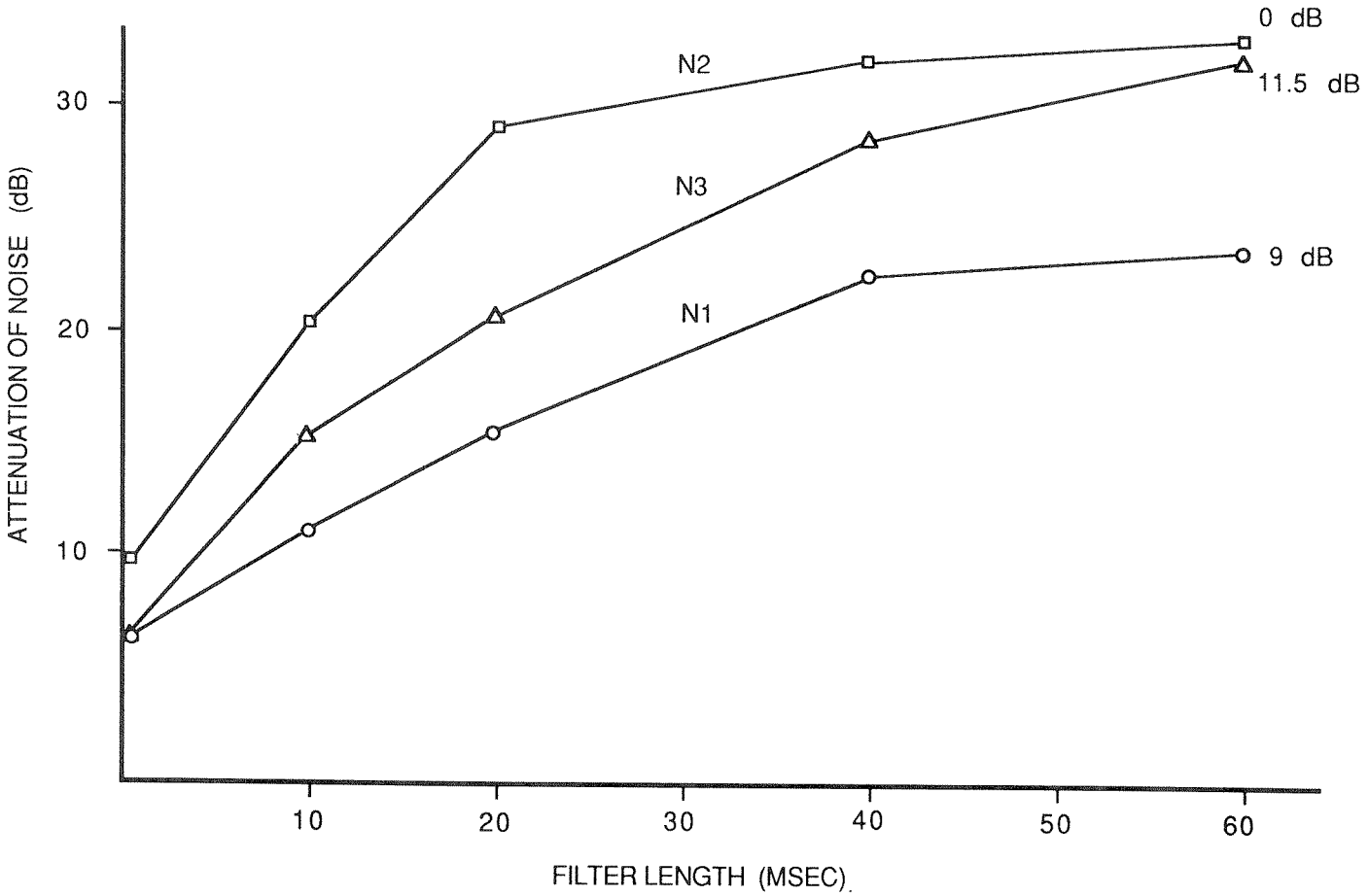

Figure 3.
Increase of Attenuation With Filter Length for Each Noise Source. The curves show the attenuation of noise as a function of filter length. For these data, only one noise source was active during adaptation. The parameter values indicate the difference (in dB) between the noise levels in the reference and primary channels.

one active while the filter was adapting. Both N2 and N3 reach peak attenuation levels greater than 30 dB in the anechoic room. The lower peak level of N1, 23.5 dB, may be due to the greater complexity of the transmission paths from the N1 loudspeaker to the microphones. The N2 and N3 sounds reached the microphones directly while for N1 the sounds involved diffraction around the head of the mannikin.

Figure 3 shows that for each of the noise sources, attenuation increased with increasing filter length. This attenuation increased rapidly at first, then much more slowly. For N2, the change in the slope of the curve occurs at a filter length of about 20 ms while for N1 and N3 the break is at about 40 ms. The parameter value shown for each curve indicates the level (in dB) of the noise in the reference channel relative to N2. Since the noise levels in the primary channel were all equal, these parameter values also represent the reference-to-primary noise level ratios. The point at which the slope of each curve changes sharply appears to be related to these values.

When more than one sound source was active during adaptation, the level and pattern of the attenuation of the noise sources changed substantially. As the number of active noise sources increased, the attenuation of a given noise source decreased. This was to be expected, since as the number of independent active sound sources increased the correlation of the signals in the reference and primary channels decreased, and consequently the impulse response acquired by the filter during adaptation became increasingly less appropriate for the transmission path characteristics of any one noise source. The degree to which the filter characteristic was appropriate for the active noise source

depended directly on the relative strengths of the signals from that source as compared with the strengths of all other signals in the reference channel.

For the experiment reported here, the strengths of N1 and N3 relative to that of N2 were −8.5 dB and −11.5 dB, respectively. Accordingly, as shown in **Table 3**, the attenuation of N2 for any given combination of filter length and number and type of active sources was always greater than the attenuation of N1 for the same conditions, and similarly, the attenuation of N1 was always greater than that of N3.

The presence of more than one active sound source during adaptation also affected the relationship between the attenuation of a noise source and the length of the adaptive filter. As the data in the first row of **Table 3** show, when only one noise source was active, the attenuation of that noise increased monotonically with increasing filter length. When more than one sound source was active, the attenuation of a given noise source reached a peak value (or within 0.5 dB of the peak value) at a filter length that appeared to be independent of the number of active sources. For all three noise sources that filter length was about 10 ms.

The effects of reverberation on the performance of the adaptive noise canceler are summarized in **Table 4**. For each adaptation condition and filter length, the attenuation level obtained was substantially lower than that obtained in the anechoic room. The data in the first row of **Table 4** show a monotonic increase in attenuation with increasing filter length, but the rate of increase is much slower than the corresponding values in **Table 3** for an anechoic room. The failure to reach an asymptote indicates that the maximum available filter length, 140 ms, was insufficient to cancel a significant fraction of the reflections in a room with a reverberation time of 380 ms.

Reverberation also reduced the directional selectivity of the reference microphone, and increased the signal-to-noise ratio in the reference channel, as shown in **Table 2**. The first effect reduced the difference between the attenuation levels obtained for N2 and N1. The second effect decreased the correlation between signals in the reference and primary channels, resulting in a uniform reduction in the cancellation of noise in the system output as compared with that obtained under anechoic conditions. For N3, the noise-to-signal ratio in the reference channel was so low that the attenuation of its signal in the primary channel was reduced to 0 dB when one other noise source was active.

SUMMARY AND CONCLUSIONS

An experiment was performed to evaluate the use of an adaptive noise canceler as a preprocessor for a hearing aid. An omnidirectional microphone and a directional microphone, mounted just above one ear on the head of a Kemar mannikin, supplied inputs to the primary and reference channels of the adaptive noise canceler. Tests of the system were conducted for various filter lengths in both anechoic and reverberant environments, using up to three independent noise sources to simulate practical situations in which background noises would reduce speech intelligibility for a hearing-impaired person. The results showed that, for the anechoic room, the noise canceler was able to provide significant attenuation of sounds generated by at least one of the three noise sources. For the noise source located behind the mannikin, the attenuation was greater than 3 dB under all but the most extreme test condition. At least 3 dB attenuation was achieved in half the test conditions when two noise sources were simultaneously active and in two conditions when all three noise sources were active. In the reverberant room, where the performance of the adaptive noise canceler was clearly inferior to that in the anechoic room, significant attenuation was achieved for one active noise source in five of the six conditions and for two active noise sources in two of the conditions. It can be concluded from these results that:

1. An adaptive noise canceler can be used with closely spaced microphones in a hearing aid configuration and, under a limited but practical set of conditions, provide useful attenuation of background noise;
2. Good performance can be achieved by use of a directional microphone to supply the reference channel signal; and
3. Outstanding performance can be obtained if adaptation of the filter can be controlled in such a way that the filter is allowed to adapt only when the desired speech signal either is absent or is very much weaker than the background noise.

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