LETTERS TO THE EDITOR

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Noise reduction for hearing aids: Combining directional microphones with an adaptive beamformer

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Many hearing aid users complain about a reduced intelligibility of speech in noisy environments. Directional systems are a successful approach for noise reductions in hearing aids. These systems transmit signals from acoustic sources lying in front of the hearing aid user while suppressing signals from other directions, which are assumed to be noise. Several methods are known to obtain directivity. One is to use directional microphones, another is digital postprocessing of several microphone signals. In this letter, the combination of directional microphones with the adaptive beamformer, a directional signal processing approach, is discussed. Intelligibility tests with both normal-hearing and hearing-impaired subjects are presented. It is shown that the combination of directional microphones with digital postprocessing is able to improve the intelligibility of speech in a noisy environment significantly, when compared to any one of these two approaches by itself.

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INTRODUCTION

Many hearing aid users are discontented with the performance of their device in noisy environments. Most of the noise reduction schemes presented so far are not able to improve intelligibility of speech in noise.[1] Some of the successful schemes are based on the assumption that desired signals are emitted by acoustic sources lying in front of a listener, while signals arriving from other directions are noise.

Several methods are known to realize systems with directional characteristics. Directional microphones, widely used in commercially available hearing aids, show a small, but consistently demonstrable gain in intelligibility for many acoustical environments.[2] Larger gains in signal-to-noise ratio (SNR) can be obtained by combining several directional microphones mounted on the frame of a pair of spectacles to a microphone array, as demonstrated by Soede[3] and Beckenbauer.[4] However it is difficult to realize such a device in a cosmetically satisfactory way. A different approach to achieve directivity is digital postprocessing of two microphone signals. A promising postprocessing scheme, the adaptive beamformer, was investigated by Peterson *et al.*,[5] Greenberg and Zurek,[6] and Kompis.[7]

The experiments described in this text compare the performance of (i) the adaptive beamformer approach with (ii) a small but cosmetically unproblematical microphone array with only two directional microphones and (iii) the combination of directional microphones with an adaptive beamformer. The intelligibility of speech in noise is tested with normal-hearing and hearing-impaired volunteers.

I. EXPERIMENTAL SETTING

Figure 1 shows a diagram of the experimental setting with its three stages for recording, processing and presentation of the test signals. Signals were recorded in an office sized room (34 m^3) which has an average reverberation time of 0.4 s. A dummy head with two omnidirectional microphones in the ears (all from a Sennheiser kit MKE 2002) was supplemented by two directional microphones placed just above the ears. The directional microphones (Knowles type EB-1979 configured as cardioids in a free field) with a specified forward–backward difference of 28, 22, and 12 dB at 1, 2, and 4 kHz, respectively were mounted in behind-the-ear hearing aid housings. Two loudspeakers were placed at a distance of 1 m from the dummy head. One loudspeaker immediately in front of the dummy head emitted the desired signal (test words), while the interfering noise was emitted by the second loudspeaker at 45° to the right. The estimated direct-to-reverberant ratio of both signals was −0.4 dB.

The microphone signals were processed by a PC-based TMS320C30 floating point digital signal processing system. The sampling rate was 10 kHz for all experiments. One of three different processing schemes could be chosen: Adding two microphone signals to form a single output, passing on two microphone signals unprocessed to stereo headphones,

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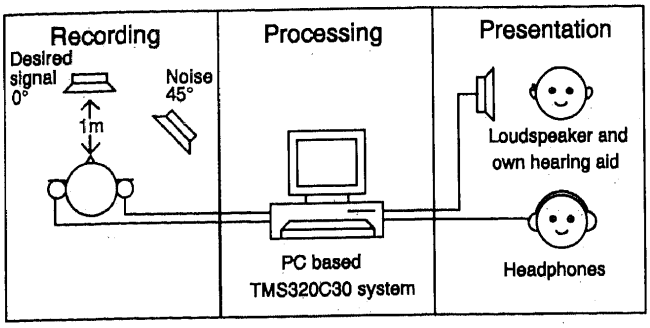

FIG. 1. The experimental setting.

or postprocessing of two microphone signals by an adaptive beamformer.

A block diagram of the adaptive beamformer used is shown in Fig. 2. The sum and the difference of the microphone signals is calculated first. The difference signal, which contains mainly noise, drives the adaptive filter which calculates an estimate of the noise in the sum signal.

The coefficients of the FIR-structured adaptive filter are updated continuously in real time by a least-mean-squares (LMS) adaptation algorithm.[8] The filter has a size of 500 coefficients (50 ms). The adaptation step size was chosen to be 0.1 of the size which leads to instability of the adaptation algorithm, resulting in an adaptation time constant of 0.125 s. In order to maximize the improvement in SNR, the sum signal was delayed by 25% of the length of the adaptive filter (12.5 ms).[7]

Whenever a strong desired signal arrives at the microphones, the adaptation of the filter is stopped. This is useful in order to prevent the adaptive beamformer from canceling those parts of the desired signal, which do not arrive with the same phase and amplitude at both microphones. The idea of an adaptation inhibition was introduced by Greenberg and Zurek,[6] and Van Compernolle.[9] The method of desired-signal detection used in this investigation is described and evaluated by Kompis.[7] The variances of the sum and difference signals are compared in an exponential window with a time constant of 10 ms. The filter adaptation is stopped, whenever the variance of the sum signal is greater than 1.5 times the variance of the difference signal.

The test signals were presented by headphones to the normal hearing subjects. In order to allow the hearing aid users to use their own hearing aids during the experiments,

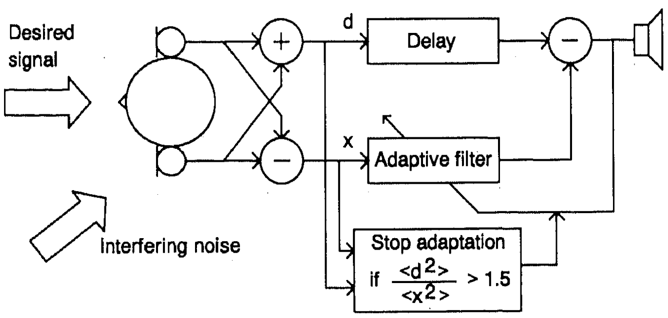

FIG. 2. Block diagram of the adaptive beamformer including a desired-signal-detection scheme.

the test signals were presented through a loudspeaker in a separate and silent but not sound proof room to this group. The estimated direct-to-reverberant ratio of this presentation was 3.6 dB. The frequency response of the overall system is a complex function of the direction of incidence of the acoustic signal. This is not only due to the characteristics of the directional microphones, but also an effect of the head shadow on any microphone attached to a head-sized object[10] and of the signal processing used. When adaptive beamforming is active, the frequency response may even vary with time. The resulting differences between the frequency responses of the tested procedures were found to be small in informal listening tests. They could be compensated by using fixed filters, however one filter can compensate only for one single direction of incidence and, due to reverberation, for one acoustic environment. In order to avoid the inherent arbitrariness of this approach and to facilitate comparison with earlier studies using adaptive beamformers,[5–7] no fixed filters were used.

II. EXPERIMENTS AND RESULTS

Intelligibility tests were performed with nine normal-hearing and six hearing-impaired volunteers. To qualify for the experiments, normal hearing subjects were required to have no known hearing disorder and no lower hearing thresholds than 20 dB in the range 125 Hz to 8 kHz. The hearing-impaired subjects were all regular hearing aid users with pure sensorineural hearing loss and used their hearing aid throughout all experiments described in this text. For both groups of volunteers the intelligibility of a desired signal was compared in the three following microphone/processing conditions:

(i) adaptive beamformer with omnidirectional microphones;

(ii) small microphone array with two directional microphones;

(iii) adaptive beamformer with directional microphones.

In condition (ii) the outputs of the two directional microphones at the ears of the dummy head were simply added. In addition, for the normal-hearing subjects the condition

(iv) stereophonic presentation of the unprocessed signals recorded by the omnidirectional microphones; was tested in order to obtain a comparison with the binaural processing of the normal hearing volunteers.

The desired signal emitted by the front loudspeaker were test words from a German minimal pair test.[11] In this test, the intelligibility of vowels and consonants is tested separately. A test word is presented together with a multiple choice of four possible answers which differ in one phoneme only, for example "Lebe," "Liebe," "Lobe," and "Labe." The raw intelligibility scores I_{raw} were chance-level corrected for the effect of guessing by the formula

$$I_{\mathrm{corrected}}=(I_{\mathrm{raw}}-C)/(1-C), \tag{1}$$

with $C=0.25$.

The noise emitted by the loudspeaker to the right of the dummy head, was a speech-spectrum shaped and randomly amplitude-modulated noise introduced by Fastl.[12] For every test situation, two different SNR's were tested. This set up

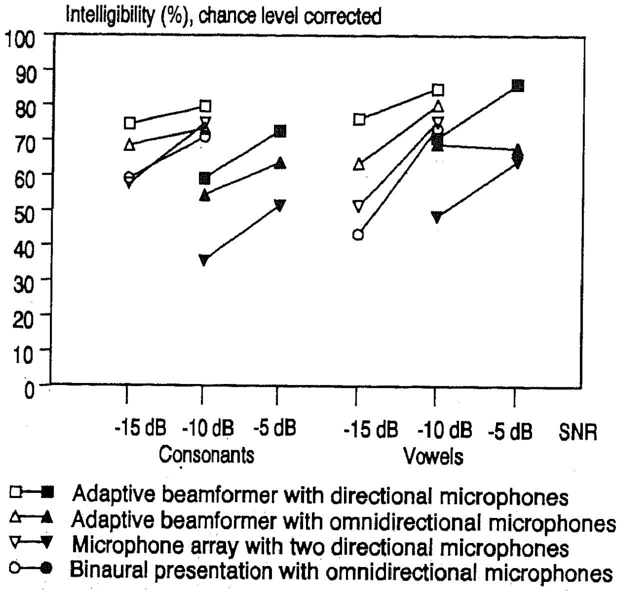

FIG. 3. Results of the intelligibility tests for consonants and vowels at three different signal-to-noise ratios. Unfilled symbols are used to represent the results from experiments with nine normal hearing volunteers (450 test words per data point); filled symbols denote the results from experiments with six hearing-impaired volunteers (300 test words per data point).

allows the assessment of the processing schemes at different SNR's rather than the estimation of an average improvement in SNR. The SNR was defined by the levels of two test signals, corresponding to the average noise level and the average level of all test words used (i.e., for the consonant and the vowel test) respectively, as measured above the dummy head. Fifty test words were presented to each volunteer for each combination of processing conditions, vowel/consonant test, and SNR. The test sequence was systematically varied in order to eliminate possible effects of training or fatigue.

The noise signal was always present well in advance of the first test word. With its short adaptation time constant of 0.125 s the beamformer was in an adapted steady state throughout all experiments.

Figure 3 shows the results of the intelligibility tests. The intelligibility for hearing aid users (filled symbols) is inherently lower than for normal hearing volunteers (unfilled symbols), and therefore the average SNR used was chosen 5 dB higher for the former group. Generally, the adaptive beamformer alone, i.e., without directional microphones, already produces a more intelligible output signal than the simple two-directional-microphone array. The best intelligibility for both groups of volunteers and all SNR's is achieved by the combination of directional microphones with adaptive beamforming. The binaural processing of the normal hearing volunteers shows similar results as the microphone array. Since the binaural processing capabilities are also affected in most sensorineurally hearing impaired persons, still lower intelligibility scores would have been expected for this group. However this situation was excluded from the test sequence because of the difficulties associated with a binaural presentation of signals to users of a single hearing aid. Subjective signal quality, as evaluated in informal listening tests, corresponds roughly to the intelligibility scores found.

To evaluate the statistical significance of the improvements by the combination of directional microphone with adaptive beamforming, a Wilcoxon pair-difference test was used. For all combinations of normal-hearing subjects/hearing-aid users, consonants/vowels and all SNR's the intelligibility obtained with the combination of directional microphones and adaptive beamforming was compared to every other microphone/processing condition tested. The improvement was significant on a 5% level for all but three tests, the exceptions being consonants at -10 dB for normal hearing volunteers with microphone array; consonants and vowels at -10 dB for hearing aid users with the adaptive beamformer alone. Although all volunteers did profit from the combination of directional microphones with adaptive beamforming in the majority of all the conditions tested, they did so at different absolute levels of intelligibility. The standard deviations of the individual data points in Fig. 3 range from 3.1% to 13.4% and are omitted for clarity.

III. DISCUSSION

Experiments were performed to estimate the benefit of a combined directional microphone/adaptive beamformer approach for noise reduction for hearing aids. Normal hearing volunteers as well as hearing aid users were tested at different SNR's. The intelligibility of vowels and consonants was measured separately. The results suggest that the combination of directional microphones and adaptive beamforming can improve the intelligibility of speech in noise significantly more than any one of these two approaches by itself, or the binaural processing of normal hearing subjects. The directional microphones also improve the reliability of the target signal detection, which results in a reduced cancellation of the desired signal at high SNR's. This might explain the different slopes for the adaptive beamformer with directional and omnidirectional microphones between -10 and -5 dB in Fig. 3. The experiments support our underlying hypothesis that combining directional microphones with an adaptive beamformer is a useful and promising approach.

To rate the relevance of these results for future practical hearing aids, several factors which have not been varied in the simple experiment deserve closer examination. The performance of the combined noise reduction scheme proposed depends most probably mainly on the following factors: (a) the direct-to-reverberant ratio of the signal- and noise sources, (b) the number, and (c) the placement of the noise sources. The placement of the listener in a given environment (e.g., close to a wall) may also have some limited influence. Since the performance of both the adaptive beamformer and directional microphones increases at high direct-to-reverberant ratios, it can be expected that the combined noise reduction scheme behaves similarly. This suggests that hearing aid users might profit most of the combined approach there, where they are most affected, i.e., in the vicinity of noise sources.

Additional noise sources have no impact on the directional microphones but they can reduce the performance of the adaptive beamformer substantially. However this is only critical when several noise sources with similar levels and spectra are present.[6,7]

As to the direction of incidence of the noise signal, adaptive beamforming, and directional microphones complement each other favorably. The opening angle for desired signals is narrow for the adaptive beamformer alone,[7] but due to the symmetry with respect to the ear axis the front-to-back discrimination is poor. Directional microphones, in contrast, show a broad opening angle but can reliably distinguish between the front and back directions.

These considerations suggest that an improvement in intelligibility by the combined directional microphones/adaptive beamforming approach can also be expected in more realistic than the given experimental situation.

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