Proceedings - 19th International Conference - IEEE/EMBS Oct. 30 - Nov. 2, 1997 Chicago, IL. USA

NEW TARGET-SIGNAL-DETECTION SCHEMES FOR MULTI-MICROPHONE NOISE-REDUCTION SYSTEMS FOR HEARING AIDS

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***Abstract* - Poor speech intelligibility in noise is a major source of dissatisfaction for users of both conventional hearing aids and cochlear implants. Many noise reduction schemes have been proposed so far. One of the more promising approaches, the adaptive beamformer, combines the input of two microphones to enhance target signals which are emitted in front of the user, while suppressing noise from other directions. In order to attain satisfactory performance, the adaptive beamformer must be combined with a reliable target-signal-detection scheme to control adaptation. In this work, two new target-signal-detection schemes are proposed and compared with a previously published algorithm by Greenberg *et al.* [1]. The proposed delta-sigma algorithm combines a more reliable SNR-estimation with a very low computational load. The new multi-correlation algorithm is computationally more expensive, but enables the user to define the angle of incidence, which differentiates between target signals and noise. To allow further evaluation in everyday listening situations, a real-time version of the adaptive beamformer with the proposed target-signal-detection schemes is being implemented on a portable system.**

I. INTRODUCTION

Many users of hearing aids and cochlear implant systems are discontented with the performance of their device in noisy environments. From the numerous noise reduction schemes that have been proposed so far, the most promising ones assume the target signal to be emitted by a source lying in front of the listener, while signals arriving from other directions are considered to be noise. Noise reduction schemes based on these assumptions can be realized by using directional microphones or microphone arrays [2,3], adaptive postprocessing of two (or more) microphone signals [1], or a combination of both [4]. This work focuses on the adaptive beamformer, which is a promising post-processing scheme for multiple microphone signals from both directional or omni-directional microphones [1,4,5].

This work has been supported by the Swiss National Research Foundation, Grant No. 4018-10864, Ascom Tech Ltd., the Josephine Clark Foundation, Bern, and the Ciba-Geigy-Jubiläums-Foundation.

II. PRINCIPLE OF THE ADAPTIVE BEAMFORMER

Fig. 1. shows a block diagram of a two microphone adaptive beamformer. The two microphones are mounted next to the ears of the user. The sum and the difference of the two microphone signals is calculated first. The sum d will already contain predominantly target signal, while the difference signal x will contain mainly noise. An adaptive filter linearly transforms x in such a way that it can serve as a model of the remaining noise in d. This signal can be directly subtracted from d. The coefficients of the FIR-structured adaptive filter are updated by a least-mean-squares (LMS) algorithm [1,4] which minimizes the total variance of the output signal. It relies on the assumption that target and noise signals are uncorrelated. Typically, the filter lngth is 10-50 ms. The delay in the target signal path d optimizes noise reduction [4] and is typically chosen to be half of the filter length.

III. TARGET-SIGNAL-DETECTION SCHEMES FOR THE ADAPTIVE BEAMFORMER

A. Importance of an adaptation control

The adaptive beamformer minimizes the variance of any signal, of which a (possibly linearly transformed) copy is present in the reference signal x. Therefore, the reference x should not contain any part of the target signal, while adaptation is in progress. - However, x will only be completely free of target signal components, if the target signal source is perfectly aligned with respect to the two

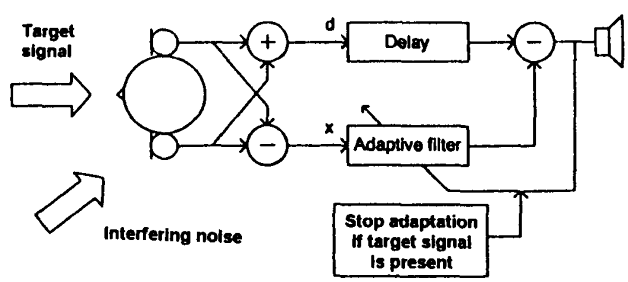

Fig. 1. Block diagram of a two-microphone adaptive beamformer

microphones and if there is no reverberation. In most practical situations, these conditions are not met. If no further precautions are taken, this results in partial cancellation of the desired signal and thus in a degradation of the signal-to-noise-ratio (SNR) and the sound quality of the output of the adatpive beamformer.

One solution to this problem is to limit filter adaptation to those time segments, when no target signal is present at all, and consequently cannot be contained in the reference [1]. If the target signal is speech, there is plenty of time for the filter to adapt during the short pauses between speech segments.

To control the adaptation process and prevent target-signal cancellation, algorithms which can reliable estimate the SNR are needed. Furthermore, it has been shown that the target-signal detection scheme will define the opening angle of the beam, i.e. the angle beyond which acoustic sources are considerd to emit noise rather than target signals [5].

B. Three Different Target-Signal-Detection Schemes

One target signal detection scheme has been proposed and tested by Greenberg et al. [1]. It is depicted on the top of Fig. 2 and labeled single-correlation-based algorithm. The microphone signals are bandpass-filtered between 700 and 1400 Hz to optimize the opening angle of the beam. Then the cross-corelation between the two signals is calculated and smoothed by low-pass filter with a time constant of 10 ms. In the original version by Greenberg *et al.* [1], calculation of the correlation is simplified by using only the sign of the filtered microphone signals. The detection parameter D will become large, whenever a signal arrives simultaneously at both microphones, thus indicating a high SNR.

Because of the band-pass filters, the single-correlation based algorithm is computationally relatively expensive. Furthermore, only a part of the incoming signal spectrum is used. Target signals with substantial parts of the signal energy outside of the band-pass frequencies will not be detected correctly.

To eliminate these two drawbacks, the sigma-delta algorithm was developed. It is depicted in the middle of Fig. 2. The sum and difference signals of the microphones, which have to be calculated by the adaptive beamformer anyway, are squared and then smoothed by a low-pass filter with a time constant of 10 ms. As for the correlation-based algorithm, the detection parameter D will tend to increase with SNR.

The target signal detection scheme determines the opening angle, in which acoustic sources are treated as target sources. It has been suggested that this opening angle should be in the order of magnitude of $\pm 13°$ to allow normal head movements during conversation [6]. Assuming a sampling rate of 10 kHz and an inter-microphone distance of 20 cm, the opening angle of the sigma-delta algorithm will be around $\pm 5°$. In order to be more flexible with this parameter, the

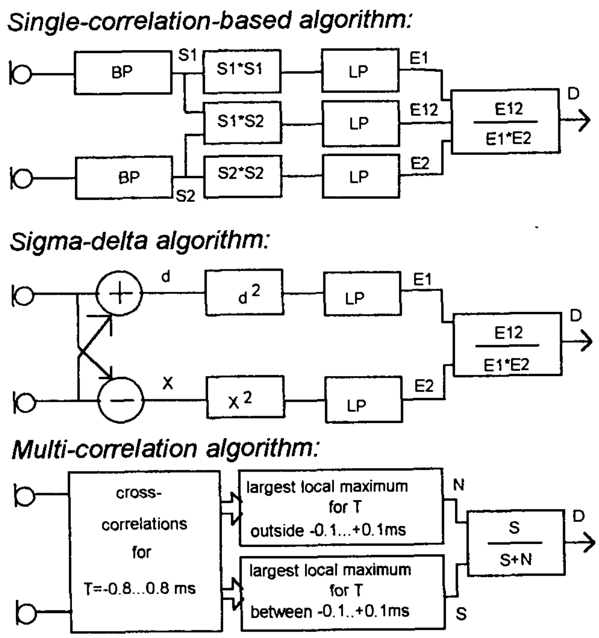

Fig. 2. Schematic of the three different target signal detection schemes (BP=Bandpass, LP=Lowpass).

multi-correlation algorithm was developed. It is shown on the bottom of Fig. 2. Cross-corelations are being calculated for all possible directions of incdence around the head. Local maxima are then extracted within and outside of the desired beam width and their values are combined into a detection parameter D. This algorithm is computationally considerably more expensive than any of the two previously described.

IV. Evaluation of the Target-Signal-Detection Schemes

A. Methods

The three target signal - detection schemes were implemented on a personal computer using a sampling frequency of 10 kHz.. Two experiments were performed. In the first experiment, the correlation between SNR and the detection parameter D was studied. In a second experiment, the effect of the direction of incidence was investgated.

To study the correlation between true SNR and the detection parameter D of each algorithm, a listening situation with a target speaker in the front and a interfering speaker at 45° to the right was simulated. For this purpose, the impulse responses of a moderately reverberant room (reverberation time 0.4 s) was simulated [7]. Both target and noise signals were german sentences for a total of 7.5 s uttered by a female speaker. The SNR and the detection

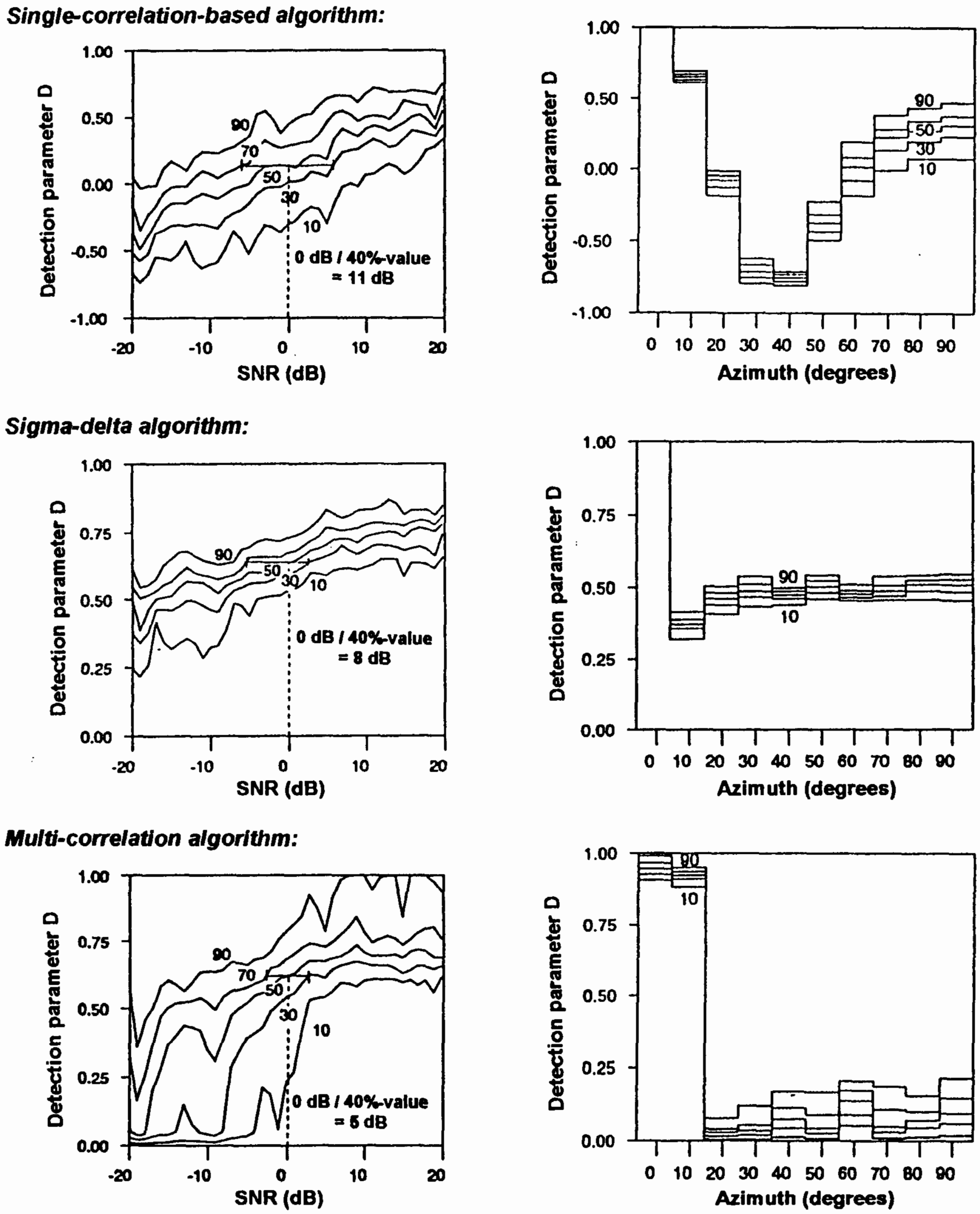

Fig. 3. Comparison of the three target-signal detection schemes. Diagrams on the right-hand side: correlation between true SNR and detection parameter D. Left-hand side: dependence of D on the angle of incidence (0°=front, 90°=side).

parameter were evaluated in segments of 10 ms and represented in terms of percentiles in a *SNR* vs. *D* plot, as shown the left three diagrams of Fig. 3. The closer the percentiles come together and the steeper they are, the better can the actual SNR be estimated from the output of the target signal detection scheme under investigation. A simple measure is the horizontal distance between the 30th and the 70th percentile centered around the 0 dB point, as depicted in Fig.3. The smaller this value, the more reliable the correspondence between SNR and D around 0 dB. 0 dB has been chosen as a reference point, as no noise redution will be needed at much higher SNR's and no speech intelligibility is likely to be restored at much lower SNR's.

In a second part of the study, an anechoic room was simulated [7] and the female speaker was moved around the listener's head in steps of 10° and at a distance of 1 m. No other acoustic source was present in this second experment. Again, the result is represented using percentiles in a D vs. *angle of incidence* plot (right-hand diagrams in Fig. 3.)

B. Results

Fig. 3. shows the results of both experiments for all of the three signal detection schemes. The 0 dB/40% value is highest (11 dB) for the single correlation algorithm and lowest (5 dB) for the multi-correlation algorithm. This indicates that around 0 dB, the true SNR can be most reliably estimated from the output of the multi-correlation algorithm and least reliably for the single correlation based algorithm. The computationally inexpensive sigma-delta-algorithm performs somewhere in between.

The three diagrams on the right-hand side of Fig. 3. show a sharp discrimination between signal and noise for the sigma-delta algorithm (between 0 and 10°) and between 10 and 20° for the multi-correlation algorithm. The single correlation algorithm shows a more complex behavior with the tendency to misclassify signals coming from the side (90°) as desired signals. When choosing a cutoff value for the detection parameter, below which it can be assumed that no significant amount of target signal is present, one more difference between the algorithms becomes visible. While any value between 0.5 and 0.7 seems reasonable for the sigma-delta and the multi-correlation algorithm, choosing a value for the single correlation algorithm is more difficult. It is essentially a tradeoff between either allowing too much target signal to be present (high D, upper left diagram in Fig. 3) or to misclassify noise sources from 90° as target signals (low D, upper right diagram in Fig. 3).

V. Real-Time Implementation

A portable digital-signal-processing (DSP) system was developed to accommodate a real-time adaptive beamformer with the proposed target signal detection algorithms. The system is based on the Motorola 3.3V-DSP 56303, which can handle up to 60 Mega instructions per second and still work for approximately 10-12 hours on one set of 4 AA-batteries. The output of the sytsem can directly drive headphones or several types of cochlear-implant systems. A photograph of the device with its two microphones in hearing-aid housings is shown in Fig. 4. The real-time version of the adaptive beamformer is currently being implemented and tests in everyday listening situations are planned.

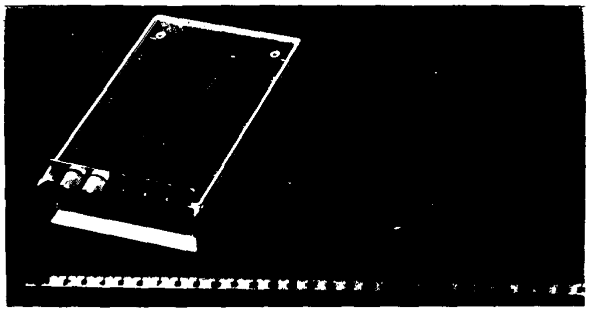

Fig. 4. Photograph of the portable DSP-system with two microphones in hearing aid-housings

VI. Discussion

Two new target signal detection schemes for adaptive-beamforming noise reduction schemes were presented and compared to a previously published algorithm by Greenberg *et al.* [1]. Experiments in simulated rooms have shown that SNR-estimation can be made more reliable without necessarily introducing a greater computational load. The newly developed multi-correlation algorithm has demonstarted the possibility to regulate the opening angle of the beam, in which acoustic sources are recognized as target signals. The impact of these improvements on the practical usefulness of the adaptive beamformer has yet to be evaluated with the portable real-time version of the adaptive beamformer.

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