Katholieke Universiteit Leuven

Departement Elektrotechniek

ESAT-SISTA/TR 2001-94

Multi-microphone noise reduction techniques for hearing aids [1]

Ann Spriet[2], Marc Moonen[3] ,Jan Wouters[4]

Published in journal of Nederlands Akoestisch Genootschap, nr. 157, pp. 51-60, September 2001

[1]This report is available by anonymous ftp from *ftp.esat.kuleuven.ac.be* in the directory *pub/sista/spriet/reports/01-94.pdf*

[2]K.U.Leuven, Dept. of Electrical Engineering (ESAT), SISTA, Kasteelpark Arenberg 10, 3001 Leuven-Heverlee, Belgium, Tel. 32/16/32 18 99, Fax 32/16/32 19 70, WWW: *http://www.esat.kuleuven.ac.be/sista*. E-mail: *ann.spriet@esat.kuleuven.ac.be*. K.U.Leuven, Lab. Exp. ORL, Dept. Neurowetenschappen, Kapucijnenvoer 33, 3000 Leuven, Belgium, Tel. 32/16/33 24 15, Fax 32/16/33 23 35, WWW: *http://www.kuleuven.ac.be/exporl/Lab/Default.htm*. Ann Spriet is a Research Assistant supported by the Fonds voor Wetenschappelijk Onderzoek (FWO) - Vlaanderen. This research work was carried out at the ESAT laboratory and Lab. Exp. ORL of the Katholieke Universiteit Leuven, in the framework of the Concerted Research Action GOA-MEFISTO-666 (Mathematical Engineering for Information and Communication Systems Technology) of the Flemish Government, IUAP P4-02 (1997-2001) 'Modeling, Identification, Simulation and Control of Complex Systems' and FWO Research Project nr. G.0233.1 ('Signal processing and automatic patient fitting for advanced auditory prostheses'). The scientific responsibility is assumed by its authors.

[3]K.U.Leuven, Dept. of Electrical Engineering (ESAT), SISTA, Kasteelpark Arenberg 10, 3001 Heverlee, Belgium, Tel. 32/16/32 17 09, Fax 32/16/32 19 70, WWW: *http://www.esat.kuleuven.ac.be/sista*. E-mail: *marc.moonen@esat.kuleuven.ac.be*. Marc Moonen is a professor at the Katholieke Universiteit Leuven.

[4]K.U.Leuven, Lab. Exp. ORL, Dept. Neurowetenschappen, Kapucijnenvoer 33, 3000 Leuven, Belgium, Tel. 32/16/33 23 42, Fax 32/16/33 23 35, WWW: *http://www.kuleuven.ac.be/exporl/Lab/Default.htm* E-mail: *jan.wouters@uz.kuleuven.ac.be*. Jan Wouters is a professor at the Katholieke Universiteit Leuven.

Multi-microphone noise reduction techniques for hearing aids

Ann Spriet[1,2], Marc Moonen[1], Jan Wouters[2]

[1]K. U. Leuven, ESAT-SISTA
Kasteelpark Arenberg 10
B-3001 Leuven-Heverlee, Belgium
{spriet,moonen}@esat.kuleuven.ac.be

[2]K. U. Leuven, Lab. Exp. ORL
Kapucijnenvoer 33
B-3000 Leuven, Belgium
jan.wouters@uz.kuleuven.ac.be

Summary: Noise reduction algorithms are crucial to improve the speech intelligibility for hearing impaired people in background noise. Multi-microphone systems exploit spatial in addition to spectral information of the desired and noise signal, and are thus preferred to single microphone procedures.
This study evaluates the performance achievable with a small-sized endfire array. Two different adaptive multi-microphone algorithms are studied: a widely studied beamforming algorithm, the Generalized Sidelobe Canceller (GSC), and a new, more robust subband algorithm based on a Generalized Singular Value Decomposition (GSVD). In contrast to the GSC, the GSVD-based algorithm does not make any a priori assumptions about speaker location, microphone characteristics and positions. Deviations from these assumptions are harmful to the performance of the classical approach.
The influence of the number of microphones and their interdistance is investigated. In realistic scenarios, the use of an additional, third microphone in combination with the new algorithm results in an additional improvement in intelligibility weighted SNR of 1.5 to 4 dB compared to 2-microphone systems with the same dimensions.

1 Introduction

Hearing impaired people experience difficulties to understand speech when background noise is present. Compared to normal hearing people, they may need a Signal-to-Noise Ratio (SNR) which is up to 15 dB (on the average 4.5 dB) higher. An increase in SNR of 1 dB may correspond to 15% improvement in intelligibility. Noise reduction algorithms are as a consequence crucial to improve the intelligibility.
In the past, hearing aids were only equipped with one single microphone. However, single microphone noise reduction techniques can only differentiate between signals that have a different spectrum. Quite often the disturbing signals are however also speechlike (e.g. cocktail party situation), such that these techniques will fail.
Multi-microphone techniques offer the advantage of differentiation in space: a beam can be steered towards a desired source and/or zeroes can be steered to undesired directions. These noise reduction techniques are often referred to as *beamforming techniques*. Two classes of *beamforming* techniques can be distinguished: fixed beamforming and adaptive beamforming. In the former, the filter weights in the signal processing algorithm are kept fixed. They are optimized such that a focussed beam is steered to the target direction (i.e. the forward-look direction) while reducing background noise coming from other directions as much as possible. In adaptive beamformers, the filter weights are adapted/changed to respond to varying environments. A limited number of interferers with changing characteristics can efficiently be attenuated by steering zeroes towards their directions.
Thanks to the ongoing miniaturization in the electronics, more microphones and more processing power can be built-in in future Behind-The-Ear (BTE) hearing aids. In this paper, we assess the practical improvement in the performance of some adaptive algorithms thanks to additional microphones for a small-sized endfire array having the dimensions of a BTE hearing aid. The

effect of the microphone interspacing is also investigated. The performance assessment is based on physical measures. In a later study, perceptual tests with normal hearing and hearing impaired persons will be done. Section 2 describes the physical performance measures that are used. Section 3 explains the two different adaptive noise reduction algorithms that will be evaluated: a widely studied adaptive beamforming algorithm, called Generalized Sidelobe Canceller (GSC) [1],[2], will be compared with a new, more robust adaptive noise reduction algorithm based on a Generalized Singular Value Decomposition (GSVD) [3]. Section 4 investigates the influence of the number of microphones and their interspacing on the performance.

2 Performance measures

2.1 Improvement in intelligibility

Since the major goal of a noise reduction algorithm is to improve the speech intelligibility, performance measures incorporating speech intelligibility such as the intelligibility weighted SNR, $\mathrm{SNR}_{\mathrm{intellig}}$, have been proposed [4]

$$\mathrm{SNR}_{\mathrm{intellig}} = \sum_i I_i \rho_i, \tag{1}$$

with

- I_i = the band importance of the i-th one-third octave band [5],
- ρ_i = the signal-to-noise ratio SNR_i in the i-th one-third octave band.

2.2 Noise sensitivity and robustness

In addition to the improvement in intelligibility, the sensitivity of a noise reduction algorithm to errors in the assumed signal model (e.g. microphone mismatch) and internal noise -also called the noise sensitivity- is an important issue. When the interspacing between adjacent microphones gets smaller than half the acoustical wavelength, large filter weights are needed to reduce environmental noise. Such small arrays are also called *superdirective arrays* [6],[7]. A drawback of these large weights however, is an amplification of uncorrelated noise such as internal noise of amplifiers and microphones, internal and quantization noise of the Analog-to-Digital Convertors (ADC) and wind turbulence. A measure used to quantify the amplification of uncorrelated noise by the array is the noise sensitivity, often defined as the ratio of the array response to mutually uncorrelated white noise sources and the array response to desired signals [8],[9]. In the sequel, we propose the following measure to calculate the noise sensitivity

$$\Psi_{\mathrm{intellig}} = \sum_i I_i \Psi_i, \quad \Psi_i = G_{\mathrm{white},i} - G_{\mathrm{desired},i} \tag{2}$$

with

- $G_{\mathrm{desired},i}$ the gain (in dB) of the desired signal in the i-th one-third octave band i.e. $E_{\mathrm{out},i}^{\mathrm{desired}} - E_{\mathrm{in},i}^{\mathrm{desired}}$, where $E_{\mathrm{out},i}^{\mathrm{desired}}$ and $E_{\mathrm{in},i}^{\mathrm{desired}}$ equal the energy (in dB) of the desired signal at the output of the algorithm and the front microphone signal respectively,
- $G_{\mathrm{white},i}$ the gain (in dB) of uncorrelated, white noise in the i-th one-third octave band.

Since errors in signal processing (e.g. rounding errors due to the finite computational accuracy of digital signal processors) and deviations from assumptions being made will to some extent be uncorrelated between the microphones, the noise sensitivity also reflects the robustness of the array to these errors. In the presence of these errors, the performance of a noise reduction algorithm might deteriorate. The amount of allowable noise sensitivity will thus depend on the amount of the internal noise, the accuracy of the DSP and the processing algorithm.
In fixed algorithms, the noise sensitivity $\Psi_{intellig}$ has to be limited such that the audible internal noise during periods of silence or low noise level is acceptable. Since fixed algorithms assume matched microphones, $\Psi_{intellig}$ has to be confined as well to guarantee stability: if $\Psi_{intellig}$ is high, only a slight deviation from assumed microphone characteristics will suffice to deteriorate the performance drastically. Adaptive algorithms on the other hand will never amplify the internal noise more than the environmental-to-internal noise ratio [8]. A priori assumptions about the signal model and processing errors however limit the acceptable noise sensitivity. The more an algorithm relies on certain assumptions, the more the noise sensitivity will have to be constrained to prevent instability.

3 Multichannel adaptive noise reduction algorithms

3.1 A widely studied beamformer: the Generalized Sidelobe Canceller (GSC)

Concept

Most studies on multi-microphone adaptive filtering techniques use a Generalized Sidelobe Canceller (GSC) [1],[2],[8] or a slight variant of it as a noise reduction scheme [10],[11],[12]. The concept of the generalized sidelobe canceller is depicted in Figure 1.

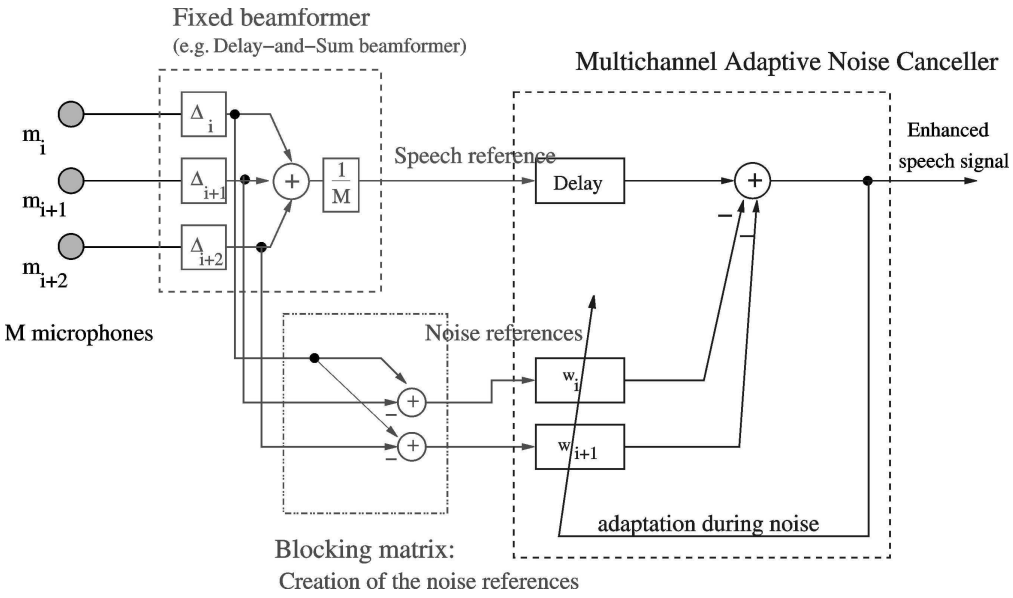

Figure 1: Concept of the Generalized Sidelobe Canceller

The GSC is based on an Adaptive Noise Canceller (ANC) [13]. The latter removes the part of the speech reference that is correlated with the noise references. The speech and noise references are

generally created through fixed beamforming: the former is created by steering a fixed beam towards 0° (i.e. the forward-look direction), the latter are created in the blocking matrix by steering zeroes towards 0°.
In this paper, the speech reference is created through fixed delay-and-sum beamforming [14]: in case of an endfire array, a signal coming from 0° arrives later in time at the rear microphones than at the front microphone. The delay-and-sum beamformer compensates for this by inserting delays. In practice, the delays are implemented by filters. The speech reference equals the sum of the delayed signals. By subtracting the aligned signals in pairs, signals are obtained in which the portion coming from 0° is cancelled. These signals are used as noise references.

Drawbacks

The GSC relies on the following assumptions: it assumes the desired signal to come from the frontal direction only and the microphone characteristics to be matched or their difference to be perfectly compensated. In real life scenarios however, these assumptions are often violated. Due to non-ideal room acoustics, the desired signal is composed of a direct path and reflections. The off axis reflections will leak into the noise references. To avoid cancellation of the desired speech signal, the ANC is adapted during periods of noise only. Due to speaker movements, even the direct path doesn't necessarily come from the front. This also introduces some leakage of speech into the noise references. In practice, microphones are never matched and moreover their characteristics may even slightly change in time. Compensating each hearing aid individually for the differences in microphone characteristics is not acceptable from a production point of view.

3.2 Robust, GSVD-based multichannel speech enhancement algorithm

Concept

In [3] an alternative multichannel speech enhancement technique based on optimal filtering, has been proposed. It produces an optimal (in Minimum Mean Squared Error (MMSE) sense) estimate of the desired signal contribution as received at a certain chosen microphone, minimizing the total noise energy and distortion of the desired speech present in the estimate. In this paper, we estimate the desired signal contribution at the front microphone. The optimal filter is obtained by means of a Generalized Singular Value Decomposition (GSVD) of an input and noise data matrix, constructed during periods of speech and noise-only respectively. Since no assumptions are made concerning microphone characteristics, room acoustics and position of the desired source, the technique is more robust to deviations than the GSC [3],[15].

Subband implementation: improved intelligibility at lower cost

In hearing aid applications, the major purpose is to maximize speech intelligibility. The definition of speech intelligibility weighted measures [4] indicates that independently minimizing the noise in separate frequency bins optimizes intelligibility more than minimizing the noise over the full frequency range. The input microphone signals are therefore split into K frequency bins or subbands by passing them through a bank of passband filters (the analysis filterbank). The optimal filtering technique is then carried out in each frequency bin independently. Finally, the enhanced subband signals are recombined into one fullband signal through a synthesis filterbank. This is depicted in Figure 2.

In [16], it is shown that the subband implementation indeed yields improved intelligibility weighted SNR compared to the fullband approach. Since the subband signals have a smaller frequency content, the processing of each signal can be done at a D times lower speed. Another advantage that arises from the subband approach is thus a reduction in processing power needs, making the implementation for hearing aid and cochlear implant applications possible in the near future. In the sequel, we call this algorithm the subband GSVD algorithm [16].

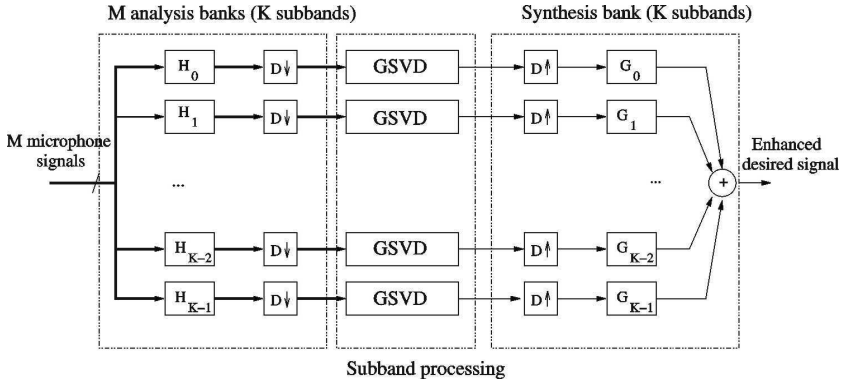

Figure 2: Concept of the subband signal processing approach

4 Experimental results

This section evaluates the performance of the noise reduction algorithms described in Section 3 for a small-sized endfire array. In Section 4.1, we describe the experimental set-up that is used. Section 4.2 analyses the robustness of the GSC and the GSVD-based approach. The influence on the performance of the number of microphones and the microphone interspacing is investigated in Section 4.3. Based on the experimental results, conclusions can be drawn about the improvement possible by incorporating more microphones in a single BTE hearing aid.

4.1 Set-up

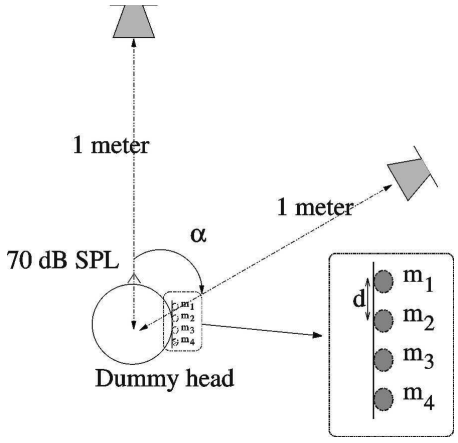

Figure 3: Description of the set-up used in the experiments

A linear uniform endfire microphone array has been mounted above the right ear of a dummy head in an office room. The endfire array consists of 4 omnidirectional microphones (KNOWLES EM-4368) of which the interspacing can be varied. Two microphone interspacings are used: a microphone interspacing of 1 cm and 2 cm corresponding to a total array length of 3 cm and 6 cm respectively. The dimensions in the first setting are similar to the ones of a Behind-The-Ear (BTE). The second setting is used to investigate the influence of the total array length on the performance. To study the effect of additional microphones only, the total array length is kept fixed. This means that for the 2-microphone array the outer two microphone signals are used and for the 3-microphone array, the outer two and the second microphone signals are used.
The microphone signals are sampled at 16 kHz. The reverberation time of the room equals about 760 msec for a speech weighted noise.
The desired signal, a stationary noise signal having the average spectrum of speech, is presented by a YAMAHA CBX-S3 loudspeaker 1 meter in front of the head. The interfering signals are also produced at a distance of 1 meter from the head. Each interfering signal has the average spectrum of speech and is uncorrelated with the desired signal and the other noise signals.
The angle between the interfering and desired signal is varied. The level of the desired signal and the sum of interfering signals equals 70 dB SPL at the center of the head. Figure 3 shows the position of the array and defines the angle α between the interferer and the desired source.
The complexity of an algorithm is mainly determined by the number of filter weights (i.e. filter length) used in each filter. The filter lengths used in the noise reduction processing schemes are decreased if more microphones are used such that the total complexity of the algorithm is kept constant as a function of the number of microphones.

4.2 Robustness analysis of the GSC and the GSVD-based approach

As mentioned in Section 2, noise reduction can be achieved with small-sized arrays, but at the expense of an increased sensitivity to errors in the assumed signal model. Due to the high sensitivity, the robustness of the algorithm that is used is crucial. In [15], the robustness of the GSC and GSVD-based algorithm is evaluated analytically and experimentally for a small-sized uniform, endfire array. Analytical evaluation of the noise sensitivity of both algorithms as a function of frequency learns that the sensitivity of both algorithms grows with:

- increasing number of microphones per array length
- increasing complexity of the noise scenario, increasing environment-to-uncorrelated noise ratio and decreasing 'frequency $\times$ microphone interspacing'-product.

The high sensitivity of small-sized arrays constrains the useful number of microphones per array length. The amplitude of the filter weights in the ANC of the GSC are proportional to the noise sensitivity. If a small deviation in the assumptions being made exists, the speech leakeage into the noise references gets significantly amplified, causing distortion to the desired speech signal. Figure 4 shows the effect on the Power Transfer Function (PTF) of the desired speech signal and noise signal from the front microphone to the output of the GSC if a deviation in gain of 1dB is applied to the 2nd microphone of a 3-microphone array (with an array length of 3cm). The noise consists of 3 noise sources: one at 60°, 150° and 270°. The desired signal at the output of the GSC is distorted if a deviation is applied. Since the noise sensitivity increases with decreasing frequency, the distortion especially occurs at the lower frequencies. Also the noise reduction

performance is degraded due to the deviation. The robustness of the GSC can be increased by putting a constraint on the norm of the ANC filters [10],[17], but at the expense of a decrease in noise reduction performance.
For comparison, the PTFs of the GSVD-based filtering algorithm are given too. The behaviour of the GSVD-based algorithm is independent of the microphone gain applied. Thanks to the higher robustness of the GSVD-based algorithm, a higher noise sensitivity can be allowed. Hence, the number of microphones that can be used per array length is larger. Section 4.3 investigates the influence of the number of microphones and their interspacing on the improvement achieved by the subband GSVD algorithm.

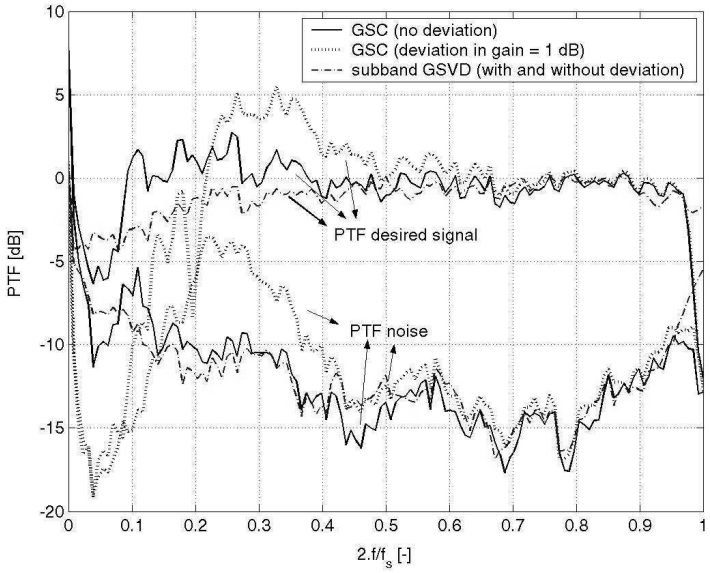

Figure 4: Influence of a deviation in 2nd microphone gain of 1 dB on the Power Transfer Function (PTF) of the desired and interfering signal for the GSC and the GSVD-based algorithm

4.3 Influence of the number of microphones and their interspacing on intelligibility

1 noise source

Figure 5 shows the $SNR_{intellig}$ of the front microphone signal and the output of the subband GSVD-based algorithm as a function of the angle between desired source and interferer for the endfire array with an array length of 3 cm. The improvement is plotted for the different number of microphones. The $SNR_{intellig}$ of the front microphone signal illustrates the head shadow effect. The use of an additional third microphone offers an additional improvement of about 2 dB. The improvement thanks to a fourth extra microphone is less pronounced.
Figure 6 plots the noise sensitivity $\Psi_{intellig}$ for the algorithm as a function of the angle between desired and interfering source. $\Psi_{intellig}$ drastically increases if a third microphone is added. It also grows if the angle between desired and interfering source decreases. This confirms the statement made in Section 4.2. Figure 7 and 8 depict the $SNR_{intellig}$ and $\Psi_{intellig}$ for an array length of 6 cm. A larger array length results in a decrease in $\Psi_{intellig}$.

In the presence of 1 noise source, a 2-microphone adaptive noise reduction algorithm performs already reasonably well such that the benefit of additional microphones might seem negligible. In real life situations though, multiple noise sources will be present. An example is the cocktail party effect where speech has to be understood among multiple competitive speakers. In the next section, the contribution of additional microphones to the performance of the subband GSVD algorithm will be studied in the presence of multiple noise sources and diffuse noise (i.e. noise coming uniformly from all directions).

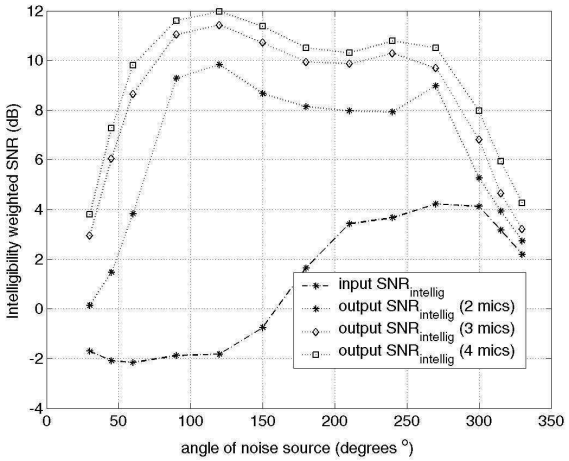

Figure 5: $SNR_{intellig}$ of subband GSVD as a function of the angle between speech and noise source, array length = 3 cm

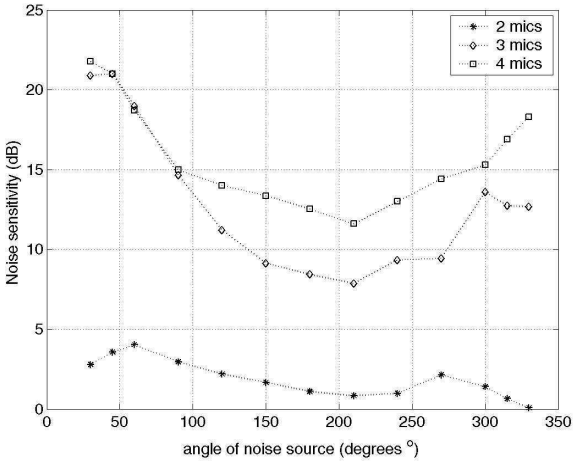

Figure 6: noise sensitivity of subband GSVD as a function of the angle between speech and noise source, array length = 3 cm

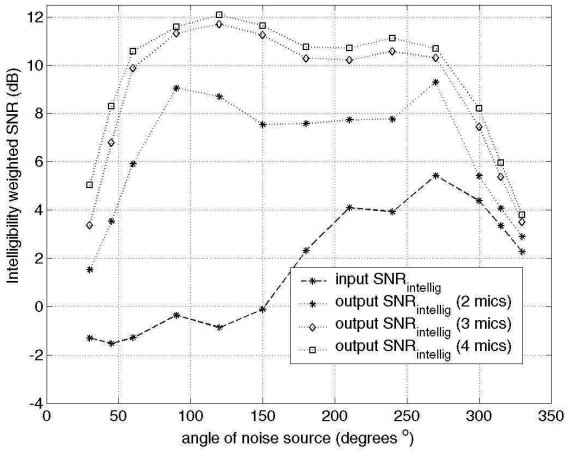

Figure 7: $SNR_{intellig}$ of subband GSVD as a function of the angle between speech and noise source, array length = 6 cm

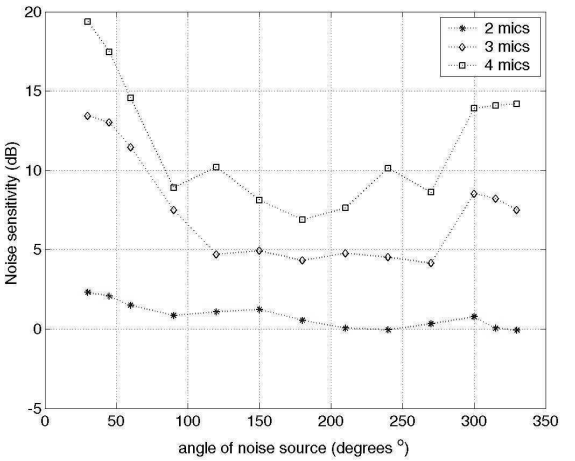

Figure 8: noise sensitivity of subband GSVD as a function of the angle between speech and noise source, array length = 6 cm

Multiple noise sources and diffuse noise

Table 1 gives the intelligibility weighted SNR and noise sensitivity $\Psi_{intellig}$ at the output of the subband GSVD algorithm for a multiple noise source scenario (noise at 60°, 150°, 180°, 240° and 300°) and diffuse noise respectively. The intelligibility weighted SNR of the front omnidirectional microphone signal is given too. An additional third microphone improves the

performance of the subband GSVD strongly by 1.5 to 4 dB respectively. Compare this to the improvement of only 2 à 3 dB $SNR_{intellig}$ offered by the 2-microphone array! The improvement of an additional fourth microphone is less. Together with the improvement, also the noise sensitivity $\Psi_{intellig}$ grows with an increasing number of microphones.
For comparison, the improvement obtained with an array length of 6 cm in a diffuse noise field is also given. The noise sensitivity $\Psi_{intellig}$ drops significantly with increasing array length. The additional improvement in $SNR_{intellig}$ by a fourth microphone is larger than for the array length of 3 cm. Large structures thus have the advantage of lower internal noise amplification and lower sensitivity to processing errors, permitting the use of less robust algorithms such as the GSC. The use of more robust algorithms is still beneficial since more microphones can be used and a better performance achieved compared to a less robust algorithm.

Table 1 : $SNR_{intellig}$ and noise sensitivity $\Psi_{intellig}$ as a funtion of the number of microphones M and the array length A.L. at the output of subband GSVD for a multiple noise source scenario and diffuse noise respectively

		input	*M=2*	*M=3*	*M=4*
Multiple noises:					
A.L. = 3 cm	$SNR_{intellig}$ [dB]	*0.43*	*3.13*	*7.28*	*8.26*
	$\Psi_{intellig}$ [dB]	-	*6.30*	*19.80*	*21.73*
Diffuse noise:					
A.L. = 3 cm	$SNR_{intellig}$ [dB]	*3.13*	*5.17*	*6.62*	*6.82*
	$\Psi_{intellig}$ [dB]	-	*0.62*	*11.83*	*13.26*
A.L. = 6 cm	$SNR_{intellig}$ [dB]	*3.20*	*4.83*	*6.58*	*7.47*
	$\Psi_{intellig}$ [dB]	-	*-0.29*	*4.99*	*11.33*

5 Conclusions

At present, digital Behind-The-Ears (BTE) with 2 microphones and a built-in Digital Signal Processor (DSP) are available. Thanks to the ongoing miniaturization in electronics, it will be feasible to build in more microphones and processing power in these hearing aids. We investigated the influence of the number of microphones, the microphone interspacing and the noise reduction algorithm on the performance of a small-sized endfire array with dimensions similar to the ones of a BTE.
Two adaptive noise reduction algorithms have been evaluated: a widely studied beamforming algorithm, the Generalized Sidelobe Canceller (GSC), and a new, more robust algorithm based on a Generalized Singular Value Decomposition (GSVD). The former relies critically on certain, seldom fulfilled assumptions. The high noise sensitivity of a 3- or 4-microphone small-sized array hence leads to a degradation in performance in the presence of deviations from these assumptions. The GSVD-based algorithm is much more robust, since it doesn't rely on these assumptions. A subband implementation of the algorithm results in improved intelligibility at a significantly lower cost, making its implementation feasible for hearing aid and cochlear implant applications in the near future.
By using the more robust subband GSVD-based algorithm, the $SNR_{intellig}$ can be significantly increased by adding a third microphone. In the presence of 1 noise source, an extra improvement of 2 dB is achieved. An even larger benefit is obtained in more real life scenarios. In these situations, the performance of 2-microphone small-sized arrays is limited to 2 à 3 dB. The use of

an extra 3rd microphone increases the $SNR_{intellig}$ by up to 4 dB! The subband GSVD algorithm is thus an ideal candidate for being implemented in future 3-microphone hearing aids …

6 Acknowledgements

Ann Spriet is a Research Assistant supported by the Fonds voor Wetenschappelijk Onderzoek (FWO) - Vlaanderen. This research work was carried out at the ESAT and Lab. Exp. ORL laboratory of the Katholieke Universiteit Leuven, in the framework of the Concerted Research Action GOA-MEFISTO-666 (Mathematical Engineering for Information and Communication Systems Technology) of the Flemish Government and IUAP P4-02 (1997-2001) 'Modeling, Identification, Simulation and Control of Complex Systems' and FWO Research Project nr. G.0233.1 ('Signal Processing and automatic patient fitting for advanced auditory prostheses'). The scientific responsibility is assumed by its authors.

References

[1] B.D. VanVeen and K.M. Buckley. Beamforming: A Versatile Approach to Spatial Filtering. *IEEE ASSP Magazine*, 5(2):4–24, April 1988.

[2] L.J. Griffiths and C.W. Jim. An alternative approach to linearly constrained adaptive beamforming. *IEEE Trans. AP*, 30:27–34, January 1982.

[3] S.Doclo and M.Moonen. GSVD-Based Optimal Filtering for Multi-Microphone Speech Enhancement*, chapter 6 in "Microphone Arrays: Signal Processing Techniques and Applications"* (Brandstein, M. S. and Ward, D. B., Eds.), pages 111–132. Springer-Verlag, May 2001.

[4] J.E. Greenberg, P.M. Peterson, and P.M. Zurek. Intelligibility-weighted measures of speech-to-interference ratio and speech system performance. *J. Acoust. Soc. Amer.*, 94(5):3009–3010, November 1993.

[5] Acoustical Society of America. ANSI S3.5-1997 American National Standard Methods for Calculation of the Speech Intelligibility Index, June 1997.

[6] J.M. Kates. Superdirective Arrays for Hearing Aids. *J. Acoust. Soc. Amer.*, 94(4):1930–1933, 1993.

[7] R.W. Stadler and W.M. Rabinowitz. On the potential of fixed arrays for hearing aids. *Journal of the Acoustical Society of America*, 94(3):1332–1342, September 1993.

[8] P.M. Peterson. Adaptive array processing for multiple microphone hearing aids. *PhD thesis*, Dept. Elect. Eng. and Comp. Sci., M. I. T, Cambridge, MA, 1989. available as Res. Lab. Elect. Techn. Rept. 541.

[9] H.Cox, R.M. Zeskind, and T.Kooij. Practical Supergain. *IEEE Trans. ASSP*, 34(3):393–398, June 1986.

[10] M.W. Hoffman, T.D. Trine, K.M. Buckley, and D.J. VanTasell. Robust Adaptive microphone array processing for hearing aids: realistic speech enhancement. *J. Acoust. Soc. Amer.*, 96:759–770, 1994.

[11] J.E. Greenberg and P.M. Zurek. Evaluation of an Adaptive Beamforming Method for Hearing Aids. *J. Acoust. Soc. Amer.*, 91(3):1662–1676, March 1992.

[12] J.VandenBerghe and J.Wouters. An adaptive noise canceller for hearing aids using two nearby microphones. *J. Acoust. Soc. Amer.*, 103:3621–3626, 1998.

[13] B.Widrow and S.Stearns. Adaptive Signal Processing. *Prentice Hall, Englewood Cliffs*, 1985.

[14] D. H. Johnson and D. E. Dudgeon. Array Signal Processing. *Prentice Hall, Englewood Cliffs,* 1993.

[15] A.Spriet, M.Moonen, and J.Wouters. Robustness analysis of GSVD-based optimal filtering and generalized sidelobe canceller for hearing aid applications. In *Proc. of WASPAA*, New Paltz, New York, pages 31-34, October 2001.

[16] A.Spriet, M.Moonen, and J.Wouters. A multichannel subband GSVD-based approach for speech enhancement in hearing aids. In *Proc. of IWAENC*, Darmstadt, Germany, pages 187-190, September 2001.

[17] H.Cox, R.M. Zeskind, and M.M. Owen. Robust Adaptive Beamforming. *IEEE Trans. ASSP*, 35(10):1365-1376, October 1987