WIDEX
HIGH DEFINITION HEARING

WIDEXPRESS

NO.23

FEBRUARY 2010

Review article:
David Morris, Carsten Paludan-Müller, Karolina Smeds and Hanne Pernille Andersen

THE **SPEECH ENHANCER** IN MIND440

A novel approach to noise reduction

Introduction

The Speech Enhancer in mind440 employs a novel approach to noise reduction by using the individual hearing threshold as a point of reference in the reshaping of the frequency response to obtain audibility of speech and comfort in noise for the hearing aid user.

Hearing aid users often deem noise reduction to be a very useful feature because it can reduce the listening effort involved in attending to speech in noisy situations. Most previously developed noise reduction strategies have focused on listening comfort rather than on maximizing speech intelligibility, and it has been difficult to show that noise reduction in hearing aids has a positive effect on speech intelligibility (e.g., Bentler et al., 2008; Mueller et al., 2006). Noise reduction might even have a detrimental effect on speech intelligibility. This is not the case with the Speech Enhancer, as this algorithm bases gain modification on a real-time optimization of the Speech Intelligibility Index (SII). The advantages of using the SII-based approach in noise reduction will be discussed in more detail in the following.

The masking effect of noise on speech

Background noise compromises the ability of a listener to hear sounds at or above their auditory threshold. In the presence of background noise, the hearing sensitivity of a listener is essentially decreased. This phenomenon is called masking. Noise can be responsible for partially or completely masking a speech signal.

When a competing noise source is present, the signal-to-noise ratio describes the relative level of the speech signal and the noise. When the background noise level is high relative to the speech signal, the signal-to-noise ratio is low and this has a detrimental effect on speech intelligibility, particularly for listeners with impaired hearing.

A speech signal has a characteristic frequency spectrum, with some frequency areas being particularly important for speech intelligibility. A band importance function, which describes the relative importance of the different frequency bands for speech may be seen in Figure 1.

The SII

The Speech Intelligibility Index (SII) is a measure of the total speech information available to a listener's ear for a particular speaker in a given listening environment. The SII is calculated numerically as a number between 0 and 1. If the SII is 1, all of the speech information is available to the listener; if the SII is 0 none of the speech

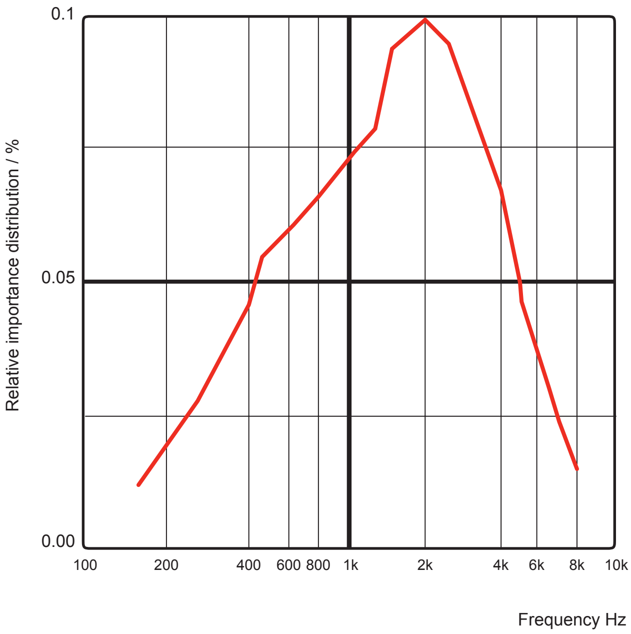

Figure 1. Band importance function showing the relative importance of different frequencies for speech intelligibility (ANSI S3.5, 1997).

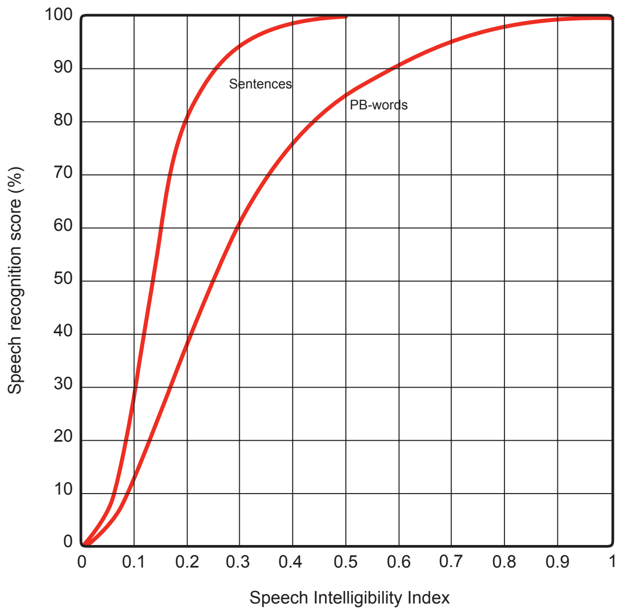

Figure 2. Recognition scores for sentences and phonemically balanced (PB) words as a function of the SII (adapted from Kringlebotn, 1999).

information is available to the listener. The SII can be calculated for listeners with normal hearing and also for those with hearing loss. The speech information can be limited by noise and by the hearing threshold. This means that the louder the noise, the lower the SII, and the greater the degree of hearing loss, the lower the SII. For a person with normal thresholds, noise is often the limiting factor for speech intelligibility. For a hearing-impaired person, on the other hand, the hearing threshold is often the limiting factor, particularly for a larger hearing loss.

For correct calculation, the SII requires three input values: the speech spectrum level, the noise spectrum level, and the hearing loss of the listener. The first two of these input values are levels which will alter according to the listening situation. The hearing loss is, of course, relatively stable and is unaffected by changes in the listening environment.

The SII correlates with speech recognition scores. That is to say, the higher the SII value, the higher the speech recognition score for a given speech material. Transfer functions may be derived for different speech materials, such as phonemically balanced word lists or sentences (see figure 2).

The calculation of the SII estimates the intelligibility of speech based on how audible the speech signal is. Audibility is determined by how much the speech spectrum is above the hearing threshold of the listener and the masking noise in various frequency bands. Because each frequency region of speech carries different weight to the overall speech intelligibility, this audibility is scaled by a band importance weighting. The SII is calculated according to the formula:

$$\mathrm{SII} = \sum_{i=1}^{n} I_i A_i$$

Where Ai is the speech audibility in band i, and Ii gives the importance of band i. This scaled audibility is then summed across all frequency bands (1 to n) (ANSI S3.5-1997).

SII optimization and the Speech Enhancer

The Speech Enhancer uses a real-time optimization of the SII to determine the hearing-aid gain reduction based on the hearing-aid user's hearing threshold and on continuously updated estimates of the speech and the noise spectra. With this information and with the knowledge about the relative intelligibility importance of various frequency regions of speech, the gain settings in each of the 15 hearing-aid compression channels are adjusted adaptively for the highest speech intelligibility. The consequence of a noise reduction strategy based on SII optimization is typically a large gain reduction in the low-frequency region, where noise is mostly found, and no gain reduction or an *increase* in gain at high frequencies, where the speech signal is weak and the hearing loss is often large.

Noise reduction in hearing aids

Noise reduction strategies vary considerably in commercially available hearing aids. In a measurement study (Smeds et al., 2009), noise reduction algorithms in a number of hearing aids were compared using coupler gain measurements. The study was inspired by similar measurements made by Hoetink et al. (2009). Real speech in speech-shaped noise at various SNRs was used as the measurement signal. Measurements were made over 30 seconds (with 30 seconds pre-conditioning) and therefore reflect the long-term average performance of the algorithms. Measurements were made with the noise reduction turned on and off, and gain reduction was calculated as the difference between the results of the two measurements. The calculated gain reductions for the various SNRs were transformed to gain reduction contour plots (presented in figure 3).

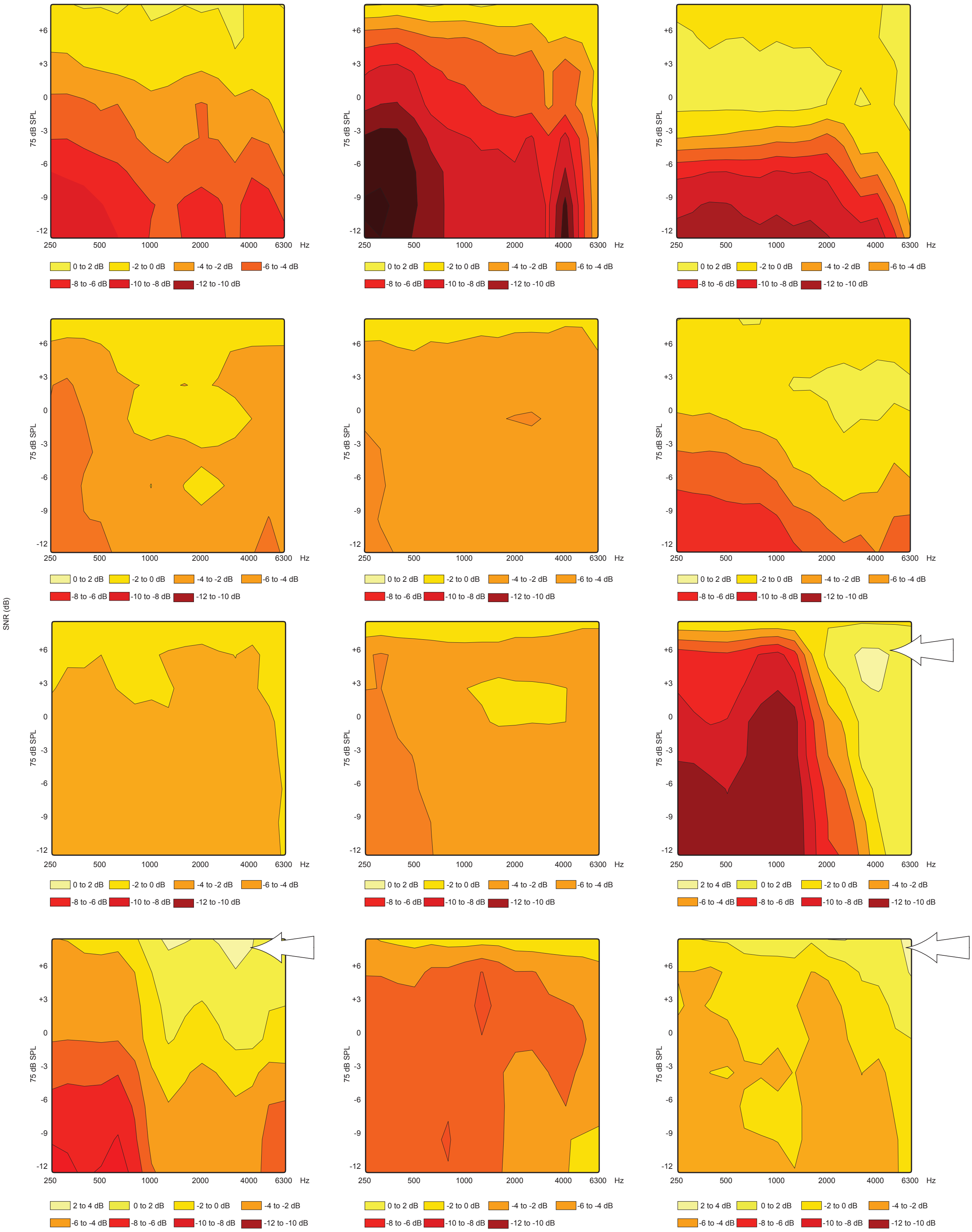

Figure 3. Gain reduction contour plots for twelve commercially available hearing aids as a function of frequency and SNR, measured with a speech signal level of 75 dB SPL. The hearing aids were programmed with their default prescription for a gently sloping audiogram. Darker colouring indicates greater gain reduction. Arrows mark the regions in three of the contour plots where gain is increased more than 2 dB (adapted from Smeds et al., 2009).

Gain reduction, as a function of frequency and SNR, is presented as the colouring in the graph – the darker the colour, the larger the gain reduction.

The differences in noise reduction performance among the tested hearing aids are very large. While it can be seen that all noise reduction strategies reduce gain as the SNR decreases, it is clear that the amount of gain reduction varies considerably. It may also be observed that gain increases are uncommon. Only three of the 12 noise reduction strategies increased gain above 2 dB at certain frequency regions in the more favourable SNR conditions.

Noise reduction as implemented by the Speech Enhancer in mind440

A more detailed look at the performance of the Speech Enhancer in the mind440 hearing aid in a number of simulated environments demonstrates that the gain manipulations are dependent on the SNR condition (see figure 4). This is evident for instance in hearing loss 2 at the 62 dB SPL speech level (middle plot). In this plot it can be seen that low frequency damping is between 0 and 4 dB at positive SNRs, while at the -12 dB SNR condition the damping increases to between 6 and 10 dB.

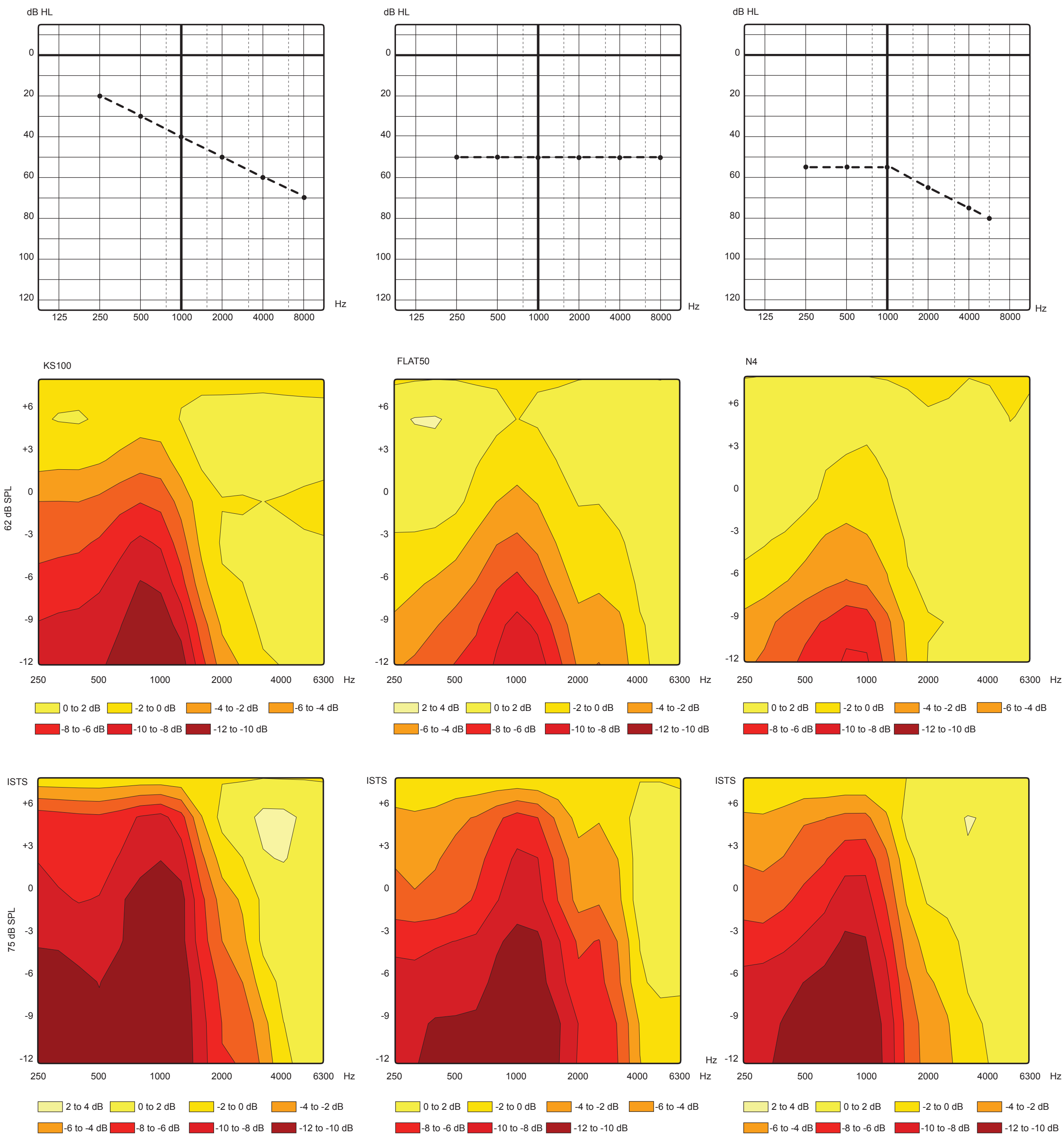

Figure 4. A set of gain reduction contour plots for the Speech Enhancer implemented in a mind440 hearing aid. The hearing aid was programmed for three hearing losses (illustrated in the top row). Gain reduction contour plots for speech at 62 dB SPL (middle row) and 75 dB SPL (lower row) are presented. Darker colouring indicates greater gain reduction (adapted from Smeds et al., 2009).

The measurements also show that the Speech Enhancer adjusts the gain reduction depending on the speech presentation level. The gain reduction is larger when the presentation level is 75 dB SPL (Figure 4, lower row) than when the presentation level is 62 dB SPL (middle row).

The measurements further show that the Speech Enhancer reacts differently when the hearing aid is programmed for different hearing losses. For instance, as the low frequency hearing thresholds become progressively worse from hearing loss 1 (left column) to hearing loss 3 (right column), the amount of low-frequency damping (area with dark shading) decreases. This is a desirable quality of the Speech Enhancer and one that is a result of the SII optimization (as described above). It means that greater degrees of hearing loss, with narrower dynamic ranges, lead to less damping by the Speech Enhancer.

The efficacy of the Speech Enhancer

The Speech Enhancer has been extensively tested in laboratory listening tests. Our internal results at Widex have confirmed that the Speech Enhancer has a demonstrable positive effect on speech intelligibility. This was seen in a study involving 12 participants with a range of audiometric configurations. Average audiograms for the 12 subjects are shown in figure 5 below.

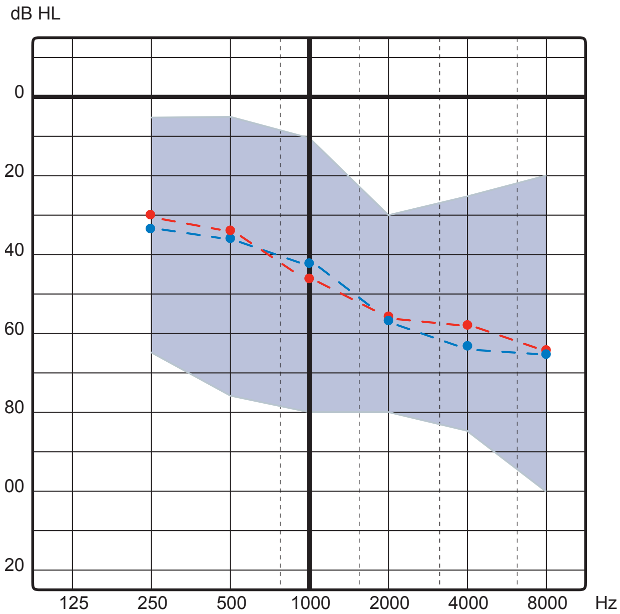

Figure 5. Average audiograms for the left (blue dotted line) and right (red dotted line) ears of the 12 subjects in Widex' internal laboratory listening tests. The shaded area indicates the minimum and maximum hearing threshold levels for the subjects.

The participants were tested with sentences from the Danish DANTALE II sentence test (Wagner et al., 2003), which has been developed in analogy with the Swedish sentence test by Hagerman (Hagerman, 1982) and the German Oldenburg sentence test (Wagener et al., 1999). The speech material was presented at 75 dB SPL and the noise was adjusted adaptively until an 80% reception threshold was reached. The test hearing aid was set in the omnidirectional microphone mode and speech and noise were presented from a speaker placed directly in front of the listener. The results are displayed in figure 6, which shows the mean SNR advantage of the Speech Enhancer as compared to no noise reduction and Widex classic noise reduction, known from Senso Diva.

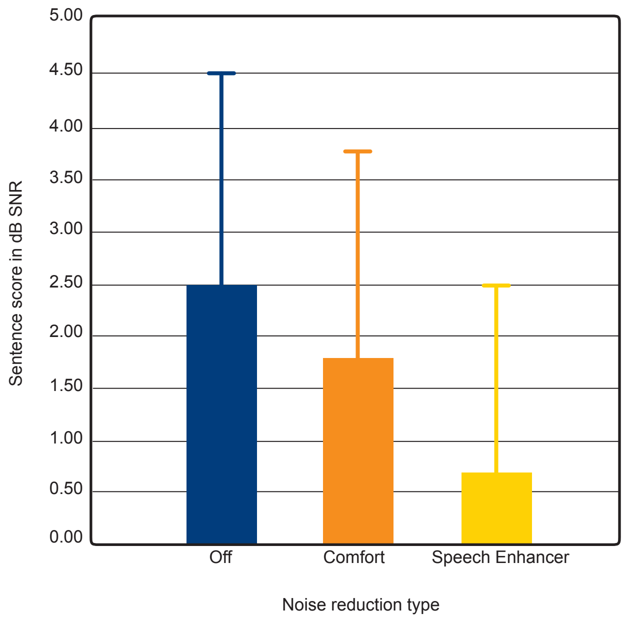

Figure 6. The mean (bars) and first standard deviation (lines) of the 80% speech reception threshold achieved with a prototype hearing aid set in omnidirectional mode. The first bar shows the results with the noise reduction switched off, the second bar the Widex classic, comfort-based noise reduction setting, and the third bar shows the results with the Speech Enhancer.

This research was performed as part of the Widex development cycle and was encouraging as it provided us with evidence that the Speech Enhancer has a demonstrable positive effect on speech intelligibility in noise. The tendency of the Speech Enhancer to benefit speech intelligibility in noise has been replicated in subsequent research (Kuk et al., 2007).

Peeters et al. (2009) found a significant improvement in speech-in-noise performance with the Speech Enhancer. Listening with the Speech Enhancer activated improved the mean speech reception threshold of 18 listeners with impaired hearing by 2.5 dB SNR when listening with the omnidirectional microphone mode and by 0.6 dB when listening with the directional microphone mode (see figure 7). The improvement in SNR was statistically significant for the omnidirectional mode. A similar, statistically significant improvement in speech perception in noise has not been reported in evaluations of other noise reduction systems (e.g., Bentler et al., 2008; Mueller et al., 2006).

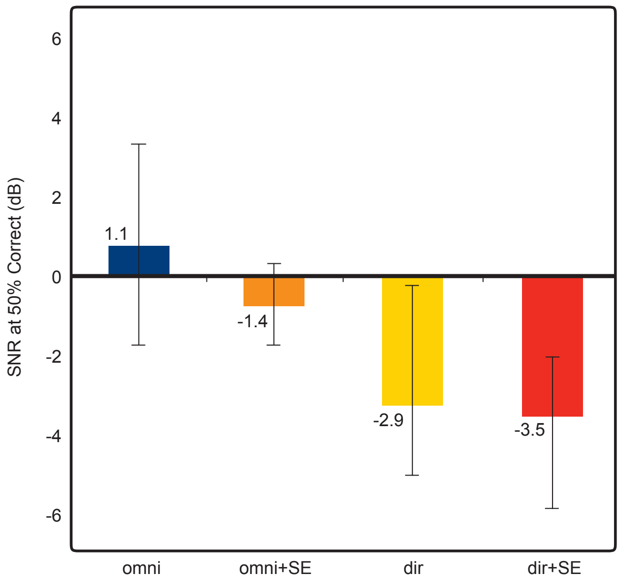

Figure 7. Average dB SNR required for 50 % correct scores on the HINT test with the Speech Enhancer activated and deactivated using different microphone settings. The error bars represent one standard deviation (reproduced from Peeters et al., 2009).

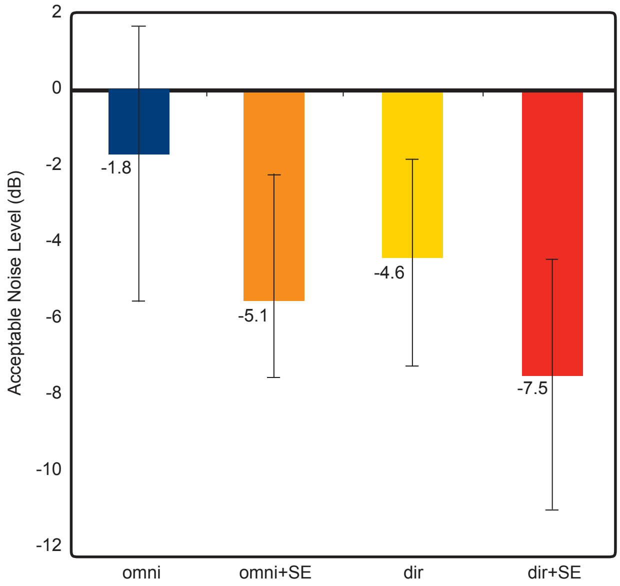

Figure 8. The decrease in average ANL with the Speech Enhancer activated and deactivated in different microphone settings. A lower ANL indicates increased tolerance of background noise (reproduced from Peeters et al., 2009).

Comfort is not compromised by the noise reduction strategy of the Speech Enhancer

Peeters et al. (2009) also investigated listener reactions to background noise while wearing a hearing aid with the Speech Enhancer using an ANL test. The Acceptable Noise Level (ANL) is found by first adjusting a speech signal to a listener's most comfortable listening level (MCL). Then background noise (BNL) is added and adjusted to a level that the listener is willing to accept. The difference between the MCL and the BNL is the ANL. The measure has proved to be useful in investigating listeners' reaction to, and tolerance of background noise. This metric is not directly related to the intelligibility of speech in noise. Instead it is a clinical measure of a listener's comfort under challenging listening conditions. ANL measurements are stable and not susceptible to acclimatization. They are also of some interest as a pre-fitting prognostic indicator of potential hearing-aid benefit (Nabelek et al., 2004).

In an ANL-comparison between directional and omnidirectional microphone settings, Freyaldenhoven et al., (2003) found a decrease in the ANL of 3.5 dB (lower ANLs indicate more tolerance) when the directional microphone was used compared to when the omnidirectional microphone was used. In the study by Peeters et al. (2009), the Speech Enhancer produced a reduction in the ANL of 3.3 dB in the omnidirectional mode and of 2.9 dB in the directional mode (see figure 8).

From this study it can be seen that the magnitude of the ANL-decrease with the Speech Enhancer activated in the omnidirectional setting is greater than the decrease in ANL when the microphone setting changes from omnidirectional to directional. This indicates that the Speech Enhancer, while benefiting speech intelligibility, does not compromise listener comfort in noisy environments.

Perspectives

Listening in the real world is a difficult problem for someone with a hearing impairment. The masking effects of noise, and the difficulty of separating noise and signals of interest exacerbate the inherent problems of hearing loss. Due to the complex nature of this problem, there is no flawless multi-factorial predictive measure of speech intelligibility. Nevertheless, the SII is the best standardized measure currently available for predicting speech intelligibility in noise, and therefore the best theoretical basis for the design of a noise reduction system that aims at optimizing speech intelligibility.

The SII provides many benefits to the Speech Enhancer, including the incorporation of hearing loss information into the workings of the noise reduction system. The unique behaviour of the Speech Enhancer has been confirmed in coupler gain measurements, and the benefit of the Speech Enhancer for listeners with hearing impairment has been demonstrated in laboratory listening studies.

References

ANSI S3.5. 1997. American National Standard: Methods for the calculation of the Speech Intelligibility Index.

Bentler, R., Wu. Y. H., Kettel, J., and Hurtig, R. (2008). Digital noise reduction: outcomes from laboratory and field studies. International Journal of Audiology, 47, 447-460.

Freyaldenhoven, M., Nabelek, A. K., and Burchfield, S. B. (2003). Comparison of acceptance of background noise and speech reception threshold in quantifying the hearing aid directivity benefit (A). Journal of the Acoustical Society of America 113(4), 2288.

Hagerman, B. (1982). Sentences for testing speech intelligibility in noise. Scandinavian Audiology, 11(2), 79-87.

Hoetink, A.E., Körössy L. & Dreschler, W.A. (2009). Classification of steady state gain reduction produced by amplitude modulation based noise reduction in digital hearing aids. International Journal of Audiology, 48, 444-455.

Kringlebotn, M. (1999). A graphical method for calculating the speech intelligibility index and measuring hearing disability from audiograms. Scandinavian Audiology 28, 151-160.

Kuk, F. Keenan, D., and Bækgård, L. 2007. Speech-in-noise performance of a micro-size BTE. The Hearing Review 14, 64-67.

Mueller, H. G., Weber, J. and Hornsby, B. W. (2006). The effects of digital noise reduction on the acceptance of background noise. Trends in Amplification 10, 83-93.

Nabelek, A. K., Tampas, J. W., and Burchfield, S. B. (2004). Comparison of speech perception in background noise with acceptance of background noise in aided and unaided conditions. Journal of Speech, Language, and Hearing Research 47, 1001-1011.

Peeters, H., Kuk, F., Lau, C. and Keenan, D. (2009). Subjective and objective evaluation of noise management algorithms. Journal of the American Academy of Audiology 20, 89-98.

Smeds K., Bergman N., Hertzman S., and Nyman T. (2009). Noise reduction in modern hearing aids – Long-term average gain measurements using speech. Presented at the International Symposium on Auditory and Audiological Research (ISAAR). Helsingør, Denmark. August, 2009.

Wagener, K., Brand, T., & Kollmeier, B. (1999). Entwicklung und Evaluation eines Satztests für die deutsche Sprache I-III: Design, Optimierung und Evaluation des Oldenburger Satztests. Audiologische Akustik/Audiological Acoustics, 38(1-3), 4-15, 44-56, 86-95.

Wagner, K., Josvassen, J. L., & Ardenkjær, R. (2003). Design, optimization and evaluation of a Danish sentence test in noise. International Journal of Audiology 42, 10-17.