Intelligent processing to optimize the benefits of hearing aids
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CHAPTER 1.

INTRODUCTION
1. Introduction

This thesis describes a number of clinical studies that investigate the benefits of different methods of compensation for hearing-impairment by hearing aids. One of the most important methods is the restoration of binaural hearing by the application of bilateral hearing aids. Therefore, the first part of this thesis is devoted to the benefits of bilateral amplification.

In addition, the introduction of digital hearing aids facilitated advanced signal processing schemes like noise reduction and dual-microphone directionality. The second part of this thesis describes three studies that assess the added value of these complex and sometimes expensive algorithms.

1.1. Benefits of bilateral hearing aids

The most intelligent processing that will help the hearing-impaired listener to compensate for his/her auditory deficit is the processing of his/her own brain. One of the important mechanisms in this respect is binaural processing, that can be optimized by a bilateral fitting with hearing aids.

1.1.1. Rationale of bilateral fitting

It is generally accepted that the use of two ears has a number advantages. With two ears it is easier to localize sounds and with two ears the spatial experience of the room acoustics is more natural than with one ear. Another advantage is better speech intelligibility in background noise. By the use of two ears we are able to separate speech and noise better than with one ear, especially when there is a spatial separation between
the sound sources. Finally, binaural hearing decreases the negative effect of reverberation on speech intelligibility. This effect is called binaural dereverberation.

Above-mentioned advantages are relevant for difficult acoustical situations that hearing-impaired people have to cope with. The most heard problem of the hearing-impaired person is that speech intelligibility is difficult in background noise and in reverberation. Unfortunately, this is a situation in which also the hearing aid usually provides only little benefit. Consequently, it is of the utmost importance to maintain or restore the function of binaural hearing in hearing-impaired listeners by a bilaterally hearing aid fitting (Markides, 1977). A systematic review of recent literature with respect to the benefits of bilateral hearing aids will be presented in Chapter 3.

1.1.2. Current criteria for reimbursement of a bilateral fitting with hearing aids

In clinical practice it is rather difficult to assess the improvement of bilateral hearing aid fittings in objective evaluation measurements, because in a one-to-one condition the binaural benefit is hardly present. In the Netherlands the official indication to get hearing aids (partly) reimbursed by the health insurance companies is based on parameters of speech intelligibility and localization.

Speech discrimination should be improved by at least 10 percent due to bilateral fitting. Localization should be restored to within 45 degrees using two hearing aids.

There is no guarantee that especially these parameters correlate well with ‘real-life’ improvements. In addition, the reliability of the speech intelligibility scores is only limited. Especially when words are presented ‘live’ (this is not unusual in clinical practice) the improvement to be obtained is of the same order of magnitude as measurement inaccuracy.

Also, there is no clear specification of the way the benefits in speech discrimination should be measured. The use of words or sentences (which is more related to real-life
conditions) has not been defined. It is unclear whether the speech material should be presented in quiet or in background noise (more realistic). Speech intelligibility will strongly depend on the possibility of lip reading and on the spatial positions of speaker and noise sources relative to the listener.

At the moment the improvement in horizontal localization is most frequently used as a motivation for the reimbursement of bilateral hearing aids. A problem with the current fence for reimbursement is that a number of candidates reach already localization ability within 45 degrees with only one hearing aid.

1.1.3. Rationale for the study on the benefits of bilateral hearing aids

As mentioned above, criteria for the (partial) reimbursement of hearing aids by the health insurance companies are poorly specified and lack a strong relationship with real-life situations. Besides this, the decision to choose for one or two hearing aids is not only dependent on speech intelligibility and localization. Therefore, we investigated retrospectively a large number of clinical files. We inventoried different aspects of current fitting practices in the Netherlands to retrieve more information about the anamnestic, audiometric, and the rehabilitation data. In addition, the hearing-impaired people included in this retrospective study were asked to fill in an extensive questionnaire to get additional information about the subjective results.

After this retrospective study still some questions needed a more detailed answer, especially on an individual basis. Therefore, we conducted a prospective study with focus on the following questions:

- Can we predict a positive effect of a bilateral hearing aid fitting?

  To investigate the possibility for predicting the effect of a bilaterally fitting using other information than parameters of the tone audiogram and the speech audiogram, we conducted a prospective study with "new" diagnostic tests. The diagnostic tests were especially based on the capacity of binaural interaction.
Does bilateral fitting with hearing aids work?
In this prospective study evaluation tests were included also, to get information about unilateral and bilateral results objectively.

Does bilateral fitting with hearing aids help?
Because it remains important to get information about the subjective results, questionnaires were used to retrieve subjective information about different situations with one and with two hearing aids.

1.2. Benefits of advanced signal processing in hearing aids

In this section some new features of modern hearing aids are outlined. Some of these features have been implemented already in analogue hearing aids (like multiple-channel processing). Others are specific for the use of digital hearing aids (like the feature of feedback reduction). For all features it can be stated that digital technology allowed more flexibility and/or an improved effectiveness. In addition digital technology stimulated the use of several features in the same hearing aid.

The most intriguing developments are in the field of noise reduction. The term “noise reduction” is used for different methods that aim to improve the balance between the wanted signal (called “speech”) and the unwanted signal (called “noise”). Noise reduction uses the differences between speech and noise.

- Noise reduction by multi-channel compression is based on spectral differences.
- Modulation-based noise reduction uses a combination of temporal and spectral differences.
- Noise reduction by directional microphones applies spatial differences between speech and noise sources.
1.2.1. The use of multiple programs

Since long it has been recognized that one fixed setting of a hearing aid is not optimal for the many different auditory tasks in many different acoustical situations. In fact, the hearing aid fitter is trying to find a compromise between optimal speech perception in quiet, optimal speech perception in noise, listening comfort, music perception, etc. The introduction of non-linear hearing aids facilitated the automatic adaptation of the hearing aids to different input levels (compression), but still there is a need for multiple programs by some hearing-aid users. In quiet situations most hearing-impaired subjects prefer their reference gain. When there is more background noise they like to have less gain in the lower frequencies. However, in high frequency background noise and possibly for listening to music, they prefer a flatter response than their reference response (Keidser et al., 1996).

Not every hearing-impaired person wants to have a multiple-program hearing aid. This depends on the number of different acoustical situations and the degree of differences between those acoustical situations. It is also important that the possible range of variation between the programs is large enough, given the requirements for gain and output of the individual subject. For example, a subject with a ski-slope audiogram and near-normal thresholds for the low frequencies is usually fitted with an ear mould with large vent. In this case, the range for adjusting the lower frequencies is very small, and the hearing-impaired listener will not hear much difference between different programs.

In addition, not every one can operate the different programs when the programs have to be switched manually. A solution for this problem could be hearing aids that switch between programs automatically, depending on the amount of background noise. On the other hand, the automatic switching between programs is not always pleasant because sometimes listeners choose to optimize listening comfort instead of intelligibility. In that case, subjects usually prefer to switch between programs manually.
In digital hearing aids the possibilities for automatic adaptation of the hearing aid characteristics improved markedly. This could reduce the need for multiple-program hearing aids. On the other hand, some of the features in digital hearing aids will only be relevant in specific situations (e.g. directivity). This increased the need for multiple-program hearing aids (Dillon, 2001).

1.2.2. Signal processing in multiple channels

For most hearing aids a frequency-dependent gain characteristic is required. For relatively regular hearing losses (flat losses or losses with a uniform sloping character) single-channel hearing aids can do the job, in combination with the filtering characteristics of the ear mould, determined by an appropriate choice of vent and tubing.

For the fitting of subjects with more irregular audiograms, it is easier and more precise to compensate the hearing loss with a multiple-channel hearing aid. In a multiple-channel hearing aid the input signal will be split in different frequency channels and then it is possible to adjust the gain for each specific frequency channel independently (multiple-channel equalizer). In some subjects, also the dynamic range of hearing (the range between threshold and uncomfortable loudness level) is frequency-dependent. In that case multiple-channel hearing aids can be applied in which each channel contains its own compressor. So the compression can be adjusted for each frequency band independently. If the different channels in a multiple-channel hearing aid are equipped with compression limiting, we can also apply multiple-channel compression aids for frequency-dependent output limitation.

Another advantage for multiple-channel hearing aids is the possibility to exploit the differences in energy in different frequency regions between noise signals and speech. With a single-channel compression hearing aid the gain in all frequencies will be
reduced even if the energy in only one frequency becomes too high, causing loss of information. With a multiple-channel compression hearing aid the gain in the lower frequencies can be decreased when there is a lot of low-frequency background noise, while the gain in the higher frequencies is maintained. This is called noise reduction based on spectral differences. Noise reduction should increase comfort (Kuk et al., 1990) and theoretically also the amount of upward spread of masking can be reduced (Cook et al., 1997; Van Tasell, 1992).

1.2.3. The use of modulation-based noise reduction

Speech can be distinguished from noise by spectral and temporal characteristics. The range of speech frequencies is roughly between 100 to 4000 Hz, but the most important frequencies for speech intelligibility are between 1000 and 2000 Hz. Speech is not a continuous signal. For a single speaker there are temporal fluctuations caused by pauses between words and sentences and by differences in energy belonging to different phonemes. Therefore, the envelope of the speech shows a characteristic temporal behaviour and shows characteristic temporal modulations. The average speaking rate is about 2.5 words per second. This corresponds to about 5 syllables per second and 12 phonemes per second. Consequently, the most dominant modulation frequencies in speech are between 2 and 8 Hz (Plomp, 1984).

The modulation spectrum for noise differs from speech. Noise often shows higher modulation frequencies than speech and has smaller modulation depths. These differences can be used to discriminate between speech and noise. In order to exploit these differences for modulation-based noise reduction in digital hearing aids the envelopes of the signals will be analysed in different frequency channels. If the signal in a specific channel is classified as “speech”, the gain and compression characteristics in that specific frequency channel band will be adjusted according the requirements of the hearing loss. If the modulation spectrum of the signal is classified as “noise”, the gain in
that specific frequency channel will be reduced. A recent review of the results obtained is presented by Alcántara et al., 2003.

### 1.2.4. The use of directional microphones

The signal pick-up of the microphone largely determines the signal-to-noise ratio. The best way to reduce background noise is to move the microphone to the speaker, but this is not always practical. Noise reduction by directional microphones is based on differences in the direction of incidence of the speech and the background noise signal. The traditional directional microphone has a front and a rear port. The front port in the microphone should be directed towards the speaker (0° azimuth) and the rear port is directed to the back (180° azimuth). For a noise source at the back, the sound is detected first in the rear port and the signal will be delayed in the hearing aid (‘internal delay’) for the same duration it takes to travel from the rear port to the front port (‘external delay’). As a result the sounds from both ports will reach the microphone membrane simultaneously, but from different sides and the signals will cancel each other. In contrast, the signals from the frontal direction will pass in a normal way and will not be cancelled.

In digital hearing aids often two omni-directional microphones are used instead of a directional microphone with two ports. With this so-called dual microphone technique one microphone is directed towards the speaker (front microphone) and one directed backwards (rear microphone). The principle is the same as in the directional microphone but now the internal delay between the microphones can be varied electronically. The variation of the ratio between the (electronic) internal delay and the external delay determines the directivity pattern of the dual-microphone combination (see Ricketts et al., 1999a, 1999b; 2000a, 2000b; Csermak, 2000). Another feature is that the delay can be varied adaptively depending on the direction of the (most dominant) noise source. The adaptive dual microphone technique switches
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automatically between different directivity patterns in order to reduce the most
 dominant noise source. So, the adaptive directional microphone varies its directivity
 pattern that way that a so-called notch is directed towards the most dominant noise
 source.

Recently, a hearing aid with three-microphone directionality has been introduced. A
 hearing aid with three omni-directional microphones in a horizontal line, three delay
 units and three subtraction units. With three microphones it is possible to implement a
 second-order directivity, which gives an even better directivity pattern than with a first
 order directivity. However the frequency response of a three-microphone system has an
 increased low frequency cut and this results in a reduction of sensitivity in the
 frequency range below 1000 Hz. As expected the sound quality of the extreme low
 frequency cut in a three-microphone system is not always acceptable. This can be
 compensated by a higher gain in the lower frequencies, but then the microphone noise
 will be increased too. For that reason two microphones are used for the lower
 frequencies and the three microphone processing is used for the higher frequencies only
 (> 1400 Hz).

1.2.5. The use of feedback reduction

A major problem in hearing aids is feedback oscillation. The output signal of the
 hearing aid partly leaks to the input of the microphone again. This means that the
 amplified output signal makes a complete loop again, and will be amplified more and
 more if the loop gain is larger than unity gain. Feedback is inevitable, but if the
damping for the leakage at a specific frequency is less than the gain in the forward
direction, feedback oscillation occurs and the hearing aid starts to “whistle”.

A traditional method to avoid feedback problems is to make the fitting of the ear mould
very tight. But even then feedback problems may be present. The simplest way to
reduce the feedback is to turn down the volume wheel in order to reduce the gain, but
then the hearing-aids user misses a lot of information. Another solution is to give only less high frequency gain, but those frequencies are important for speech intelligibility. So both options are not desirable.

A better option is to reduce the gain at those frequencies where feedback occurs. The more frequency bands the more precisely the gain reduction can be reduced locally. Often feedback occurs in specific situations for example when the volume control is higher than the usual setting, or if wide dynamic range compression causes relatively high gain values for low input levels. It is desirable to reduce the maximum gain to a safe value for each frequency region. This can be done by the clinician him/her self or by an in-situ feedback test (in which the fitting system raises the gain automatically until feedback occurs). The problem is that the frequency of the feedback oscillation can vary. When the gain has to be reduced at all those frequencies, a lot of information will be lost again.

With a digital feedback reduction system the hearing aid generates by purpose the same signal as the feedback signal, but now out of phase. The two signals will sum up to zero and cancel the feedback. Another method is feedback reduction with an adaptive filter. The filter will be active when there is a continuous signal at a special frequency for a certain amount of time. A disadvantage of this adaptive feedback is that other signals than feedback signals (of a special frequency for a certain amount of time) will be cancelled too.

Thanks to the increased possibilities of digital feedback systems hearing aids can also be prescribed for hearing-impaired listeners who need a very open ear mould, because of medical reasons or because of occlusion problems.
1.2.6. *Rationale for the evaluation of advanced signal processing in hearing aids*

As mentioned before hearing-impaired listeners do have a lot of problems in noisy environments. Since the introduction of digital hearing aids several improvements have been claimed. But at the same time a lot of questions came forward:

- What is the experimental evidence that should be the basis for objective information for the hearing-aid users?
- Do the benefits in daily life correspond to the claims of the manufacturer?
- More specific: which developments lead to improved speech intelligibility in noise and which developments lead to subjective benefits?
- Are the benefits valid for every hearing-aid user in every situation?

To answer these questions, this thesis reports about some field tests and laboratory studies to investigate the advantages of the noise reduction and dual microphone technique on speech perception in different acoustical situations.

For the laboratory studies, we conducted different tests to obtain knowledge about the effects of the hearing aid algorithms under study in clinical practice. We used speech perception tests in different background noises to measure the performance with the different hearing aid settings objectively. For the hearing aids with the directional microphones (fixed or adaptive) we also used a Just Follow Conversation (JFC) test with noises coming from different sides. To get more information about the effect of the different microphones on localization, a localization test was performed. Paired comparisons were used to evaluate the subjective preference for different hearing aid settings in different background noises. For the subjective evaluation we used questionnaires. With the results of those studies we could verify the claims of the manufacturer.