ASSESSMENT OF SPATIAL AND BINAURAL HEARING IN HEARING IMPAIRED LISTENERS

VRIJE UNIVERSITEIT

ASSESSMENT OF SPATIAL AND BINAURAL HEARING IN HEARING IMPAIRED LISTENERS

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Sietze Theodoor Goverts

geboren te Leiden

promotor: prof.dr.ir. T. Houtgast

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Cover:
In audiometry, red and blue are the colors that are used to indicate results for the right and left ear respectively. This thesis is concerned with the interaction of both ears. The picture is a schematic illustration of the test-configuration for the assessment of spatial hearing, which was originally designed by Joost Festen and is used in this thesis with his approval.

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1 Introduction

I IMPORTANCE AND COMPLEXITY OF SPATIAL HEARING

The human beings amazing capacity of spatial hearing is of great importance because it extends our perceptual world beyond the visual field. The combination of selective attention, sensitivity and localization accuracy provides us with an real time acoustical image of our surroundings. The availability of this auditory information is very important from a perspective of social and communicative interaction and safety.
Spatial hearing is a very complex phenomenon in which both peripheral and central, both monaural and binaural aspects are involved. The metaphor of the ‘Game on the edge of the lake’ that Bregman (1994) used to draw our notice to the difficulties in auditory scene analysis is very illustrative.

> *Imagine that you are on the edge of a lake and a friend challenges you to play a game. The game is this: Your friend digs two narrow channels up from the side of the lake. Each is a few feet long and a few inches wide and they are spaced a few feet apart. Halfway up each one, your friend stretches a handkerchief and fastens it to the sides of the channel. As waves reach the side of the lake they travel up the channels and cause the two handkerchiefs to go into motion. You are allowed to look only at the handkerchiefs and from their motions to answer a series of questions: How many boats are there on the lake and where are they? Which is the most powerful one? Which one is closer? Is the wind blowing? Has any large object been dropped suddenly into the lake?*

A schematic illustration of this game on the edge of the lake is given in Figure 1.1. The interpretation of the metaphor is clear. The lake represents the air surrounding us, the

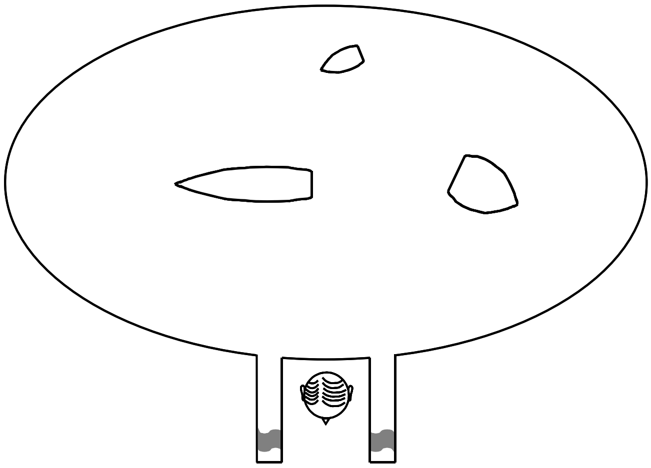


Figure 1.1 Schematic illustration of the ‘Game at the edge of the lake.’

channels are our ear channels and the handkerchiefs represent our eardrums. The questions auditory scene analysis has to deal with are in analogy to the game on the edge of the lake: How may people are talking ? Which is louder or closer? Is there a machine humming at the background? The only information available to our auditory system to perform auditory scene analysis is a bilateral vibration pattern. Bregman then concludes that ‘We are not surprised when our sense of hearing succeeds in answering these questions any more than we are when our eye, looking at the handkerchiefs, fails’. Though it is very illustrative, the Bregman metaphor is still an underestimation of the complexity of auditory scene analysis. With respect to our special interest in spatial hearing we might also add questions like ‘Do the boats move and in what direction?’ representing the capacity of our auditory system to perceive distance and movement of sound sources (for example Bronkhorst and Houtgast, 1999; Gatehouse and Noble, 2004). Furthermore, the metaphor lacks the aspects of multi source speech intelligibility, usually named the ‘cocktail party effect’ (Cherry, 1953; Bronkhorst, 2000).

The complexity of auditory perception is also acknowledged by Plomp (2002) in his monograph, going into speech perception in particular. He stresses the importance of combining research on psychophysical and cognitive aspects of sound perception, because ‘sounds have meaning, implying that their significance is much greater than the physical content of the signal’. In a Dutch text Plomp formulates very concisely ‘Perception is essentially recognition and identification’ (Plomp, 1998). Based on the physical properties of a sound stimulus, that can vary along a continuous scale, we recognize it and perceive it as one of the discrete possible auditory events. This is in line with Blauert (1997), who makes a distinction between the physical 'sound event' and the perceptual 'auditory event' that is formed. Blauert describes a signal-driven bottom-up stream in which the physical properties of the signal are led through external and middle ear to the cochlea. After spectro-temporal coding and monaural and binaural processing, a binaural activity pattern is created. From the other side there is a hypothesis-driven top-down stream in which cognition, mental aspects and non auditory information play a role. In higher stages of the central system hypotheses are set up and tested on the question what would be an appropriate percept in a certain situation. A slightly adapted version of the Blauert model is given in Figure 1.2. In addition to the Blauert model we add a 'mental effort' component in the final psychological stage in line with Kramer et al. (1997), stating that 'Effort seems to be an extra dimension which may account for disadvantages experienced by hearing impaired people in daily life'.

Acoustical cues for spatial and binaural hearing

Relevant acoustical cues in the signal-driven bottom-up stream are interaural time differences (ITD) and interaural level differences (ILD) as well as interaural phase cues and monaural and binaural spectral cues. An overview of the contribution of the cues is given by Blauert (1997) and Gilkey and Anderson (1997). In this thesis we are concerned with the assessment of spatial and binaural hearing in hearing impaired listeners and thus with the effect of hearing impairment on these cues.

II SPATIAL AND BINAURAL HEARING OF HEARING IMPAIRED LISTENERS

Given the complexity of spatial hearing as described in the preceding section it can be intuitively understood that the processes involved are easily disturbed by impairments of the auditory system. Because of the importance of spatial hearing in our 'communication with the outside world and with other human beings in particular' (Plomp 2002), we need a proper framework to describe the effects of hearing impairment on this capacity and the

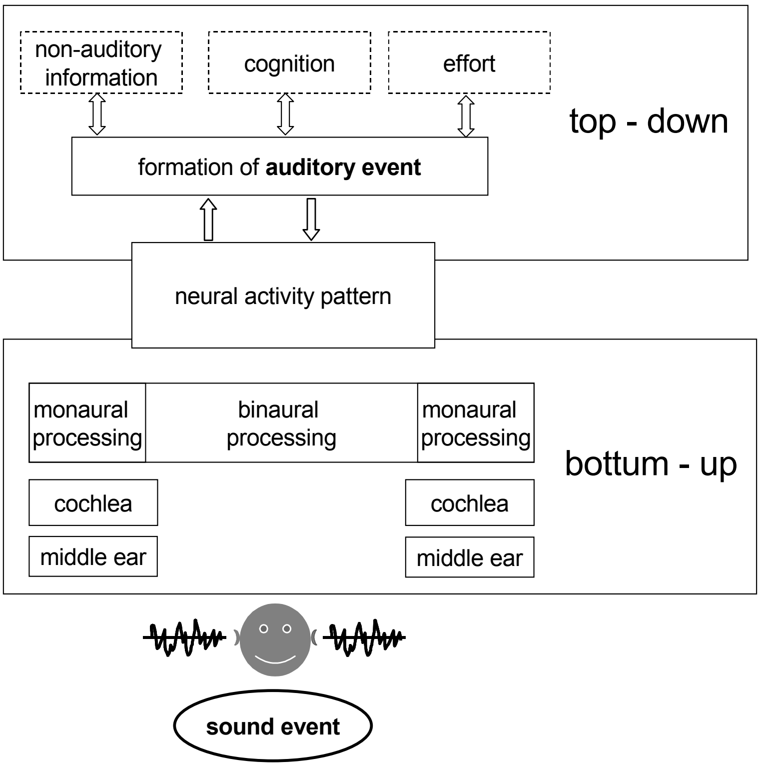


Figure 1.2 Schematic model of spatial hearing, comprising the physiological, psychophysical and psychological aspects of auditory information processing (adapted from Blauert, 1997).

consequences of these effects in everyday life.

Consequences of reduced spatial hearing in everyday life

Such a framework is formed by the classification system of the World Health Organization (WHO). In the 1980 version of the International Classification of Impairments, Diseases and Handicaps (ICIDH, WHO 1980) the terms ‘impairment’, ‘disability’ and ‘handicap’ were used to describe the effect of a disease or disorder. While these concepts are still widespread, recently (ICIDH-2, WHO 1998; ICF, 2001) the terms ‘body function and structure’, ‘activity’ and ‘participation’ are introduced. Additionally the relevance of contextual factors is acknowledged by including the concepts ‘environmental factors’ and ‘personal factors’. Major reasons for the revision were the intent to reflect more fully the consequences of the health status in the daily activity pattern and the effect on social participation (Beck, 2000) and to provide an integrative and universal model.

Kramer et al. (1995) investigated hearing disability, which would be named activity limitation nowadays, using the Amsterdam Inventory for Auditory Disability and Handicap (AIADH). Be performing factor analysis on the response of 274 hearing-impaired subjects, they found that five factors of hearing disability should be distinguished, namely 'distinction of sounds', 'intelligibility in noise', 'auditory localization', 'intelligibility in quiet' and 'detection of sounds'. Especially the factors 'intelligibility in noise' and 'auditory localization' are associated with spatial hearing. Recently, another self assessment scale was designed to measure a range of hearing disabilities in the domains of speech intelligibility, spatial hearing and hearing qualities, that is even more dedicated to the spatial hearing domain (Gatehouse and Noble, 2004). In another study (Kramer et al. 1998) it was found that the factors 'intelligibility in noise' and 'auditory localization' are the *most frequent disabilities* and that the handicap resulting from the disability of speech intelligibility in noise is *most strongly* felt. These results illustrate the relevance of understanding the effect of hearing impairment on spatial hearing. It will enhance our insights in auditory perception of course, but it will also provide guidelines for audiological rehabilitation.

Effect of hearing impairment on spatial hearing

The effect of hearing impairment on spatial and binaural hearing has been the subject of numerous studies. The first general result, found in most of these investigations is that overall performance of hearing-impaired listeners is usually poorer than performance of listeners with normal hearing. However, a wide range of performance for the hearing-impaired listeners is found, varying form close to normal to very deviant. A second general result is the limited relationship between spatial and binaural hearing performance and the pure-tone audiogram. Most studies in this domain are restricted to psychophysical assessment of impairment and activity limitation. They study for example the effect of hearing impairment on sound localization or spatial speech perception (e.g. Lorenzi et al., 1999; Noble et al. 1994; Bronkhorst and Plomp., 1989, 1990, 1992; Häusler et al., 1983). As binaural processing plays an important role in spatial hearing as illustrated by the Blauert model, a lot of research has been devoted also to the binaural hearing capacities of hearing impaired (e.g. Colburn, 1982; Gabriel et al., 1992).

III AIM AND APPROACH OF THIS THESIS

This thesis aims to contribute to appropriate and specific assessment of spatial and binaural hearing of the hearing impaired. Two ways of assessing spatial hearing are distinguished: (1) *overall assessment* of spatial hearing, in which all bottom-up and top-down processes (see Figure 1.2) are comprised and (2) the *specific assessment* of some of these underlying processes. We will first consider the correspondence between psychophysical data on overall assessment of disability in spatial hearing and results of self-assessment. The main goal of this thesis is to enhance insight in the effect of hearing impairment on the psychophysically established disability, by means of specific assessment.

Approach of the thesis

Our approach in the investigation of a valid psychophysical test on spatial hearing is rather straightforward. Using the results of self-assessment by a validated instrument (AIADH, Kramer et al., 1995) as a *gold standard*, we vary parameters in the psychophysical test seeking an optimal correspondence between both types of assessment.

In the specific assessment of the effects of hearing impairment on spatial hearing a combination of psychophysical, psychological and physiological approaches would be needed, as indicated by the adapted Blauert model (Figure 1.2). However, this model is too complex to make research with hearing-impaired subjects a realistic project, so restrictions and simplifications are necessary. This thesis is mainly restricted to the *psychophysical domain* with the exception of the use of self-assessment data. As a research framework, we use a simple model of spatial hearing, which consists of audibility, suprathreshold coding, binaural processing and central processing (see Figure 1.3). The latter is a gross term, which comprises cognition, intelligence, non-auditory processing etc. A further major simplification

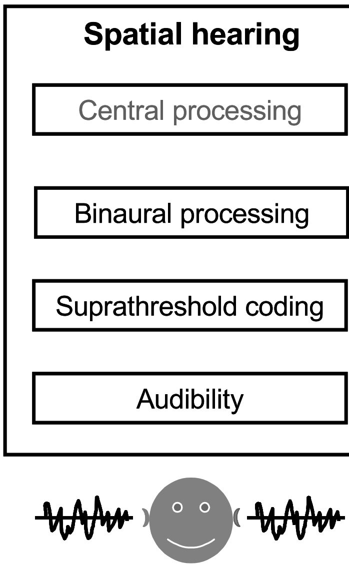


Figure 1.3 Simple model, serving as a research framework, representing four stages of auditory function that are distinguished in investigating the effect of hearing impairment on spatial hearing.

of our approach is that we do not consider the central processing term in this research, thus effectively assuming it to be constant over subjects and over listening conditions. We will investigate the binaural performance of hearing impaired and the role of reduced audibility and suprathreshold coding deficits in this binaural processing.
The general approach followed in the thesis is illustrated in Figure 1.4. In overall assessment, only *sound localization* was involved. The specific assessment of binaural hearing is restricted to the *precedence effect* and *binaural unmasking* phenomena. However, the results might be generalized to enhance our insight in the field of spatial and hearing of hearing impaired and provide a framework for appropriate clinical assessment.

The approaches for investigation of the effect of *reduced audibility* on binaural processing, used in this thesis, are rather straightforward. In a *level dependency* approach, binaural tests can be carried at lower sensation levels for normal-hearing listeners or, the other way around, at higher levels to provide better audibility for hearing-impaired listeners. Thus, the role of reduced audibility can be established. By comparing binaural performance of hearing impaired to that of normal hearing at equal sensation level, we can identify whether audibility problems account for all of the reduced binaural processing. For investigating the role of audibility also a *correlation study approach* can be used in which, for a group of hearing-impaired listeners, performance on a binaural task is correlated to pure-tone audiometric data.

A last approach is to exclude the audibility factor by measuring binaural performance with *optimized audibility*, i.e. for stimuli in the middle of the subject's dynamic range. Reduced binaural performance, while optimal audibility is ensured, should be caused at higher stages of auditory processing.

Examination of the effects of deficits in *suprathreshold coding* on binaural processing is less trivial. Correlation studies are common in this field (e.g. Kinkel, 1990; Gabriel et al. 1992). In this type of approach, binaural processing is assessed next to a battery of other tests, and the relation between the binaural performance and other auditory capacities is investigated. In this thesis we attempted to examine the influence of suprathreshold coding on binaural processing more directly. Therefore an approach is used that is based on estimating the effects of *perturbations of auditory stimuli* in a binaural task. This same approach was applied to study the effect of suprathreshold coding deficits in monaural speech reception in noise (van Schijndel et al. 2001a, 2001b). This type of approach is somewhat related to the weighting/perturbation analysis used for example by Zahorik (2002), investigating the use of acoustical cues in distance perception. In that technique the role of perceptual cues in the formation of a single stimulus percept is investigated by applying independent random perturbations to each of the physical parameters. The perceptual weights of psychophysical cues are estimated by examining the relation between responses and physical parameter perturbations. In the present study the stimulus perturbations are applied by using wavelet analysis and resynthesis of the binaural signals. Since the wavelet analysis to some extent

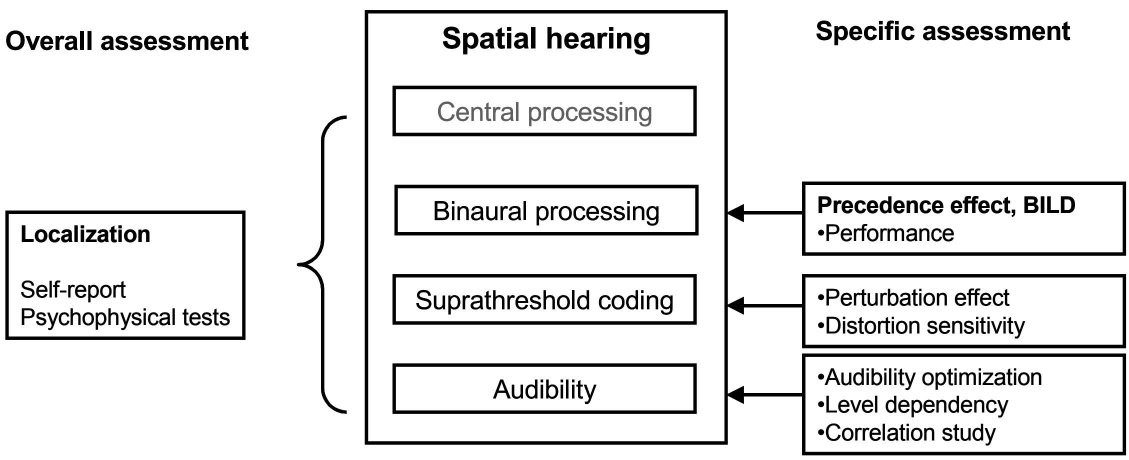


Figure 1.4 Illustration of the general approach of assessing spatial and binaural hearing of the hearing impaired, that is used in this thesis.

resembles peripheral auditory processing (van Schijndel, 2000), we can try to mimic various kinds of coding deficits or 'distortions', as induced by hearing loss. The underlying assumption of this *distortion sensitivity approach* (Houtgast, 1995; van Schijndel, 2001a, 2001b) is 'you won't notice the external distortion of a cue which is already distorted internally'. Stated more formally: 'when the auditory coding of a particular cue in sound is distorted for hearing-impaired listeners, they will be less sensitive to an artificial external distortion of this cue than normal hearing listeners' (van Schijndel, 2001a). The same type of wavelet coding perturbation is applied in order to compare the suprathreshold behaviour of two binaural phenomena, the Binaural Intelligibility Level Difference (BILD) and the Binaural Masking Level Difference (BMLD) with the *perturbation effect method*. The underlying assumption of this perturbation effect approach is 'if the processes underlying two phenomena use the same cues in a similar way, these processes will be very similar'.

In conclusion, we will first investigate the possibility of *overall assessment* of spatial hearing by a psychophysical test, the results of which correspond to self-assessment. The main goal of this thesis is the *specific psychophysical assessment* of the effect of hearing impairment on binaural processing. More precisely, we will focus on the role of reduced audibility and suprathreshold coding deficits in binaural processing of the hearing impaired.

IV OUTLINE OF THE THESIS

In the first place, in *chapter 2* the relation between psychophysical tests and self-assessment for sound localization, as an example of spatial hearing, is examined. This study leads to an optimal design for measuring sound localization. It also confirms the scattered relation between localization performance and pure-tone audiometric data, i.e. between overall auditory performance and audibility. Thus, this chapter is the point of departure for chapters 3 and 4 in which the role of audibility and suprathreshold coding in binaural processing is investigated in a more detailed way.

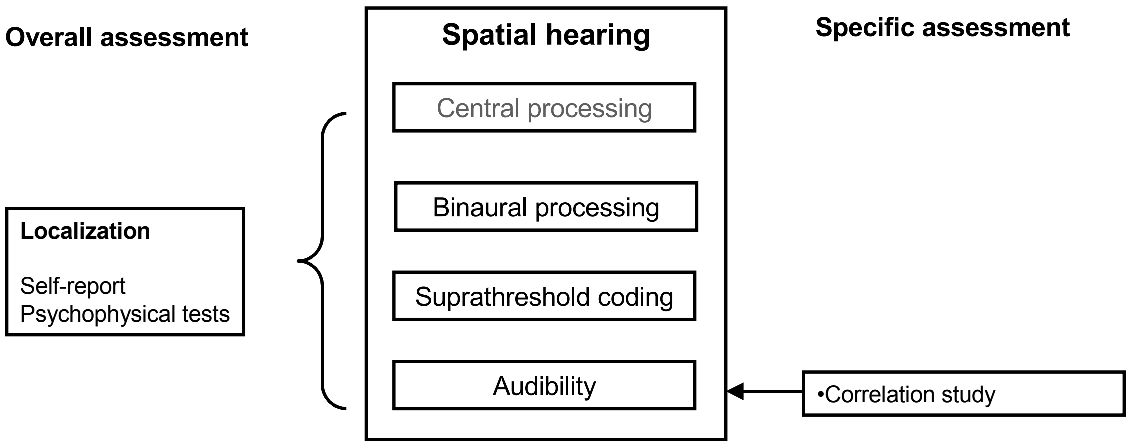
In *chapter 3* the role of audibility problems in binaural processing is further investigated, taking the phenomenon of the precedence effect as an example. A relatively new headphone-based design to quantify the precedence effect is used, aiming to minimize effects of a subject's skill and motivation that are usually inherent to this kind of measurements. The dependence of precedence effect on sensation level is measured for normal hearing in *section 3.1*. These data serve as a set of reference for *section 3.2*, where they are compared to data of six mild to moderate sensory neurally hearing-impaired listeners at either equal sound pressure level or equal sensation level. In *section 3.3* the correspondence between the results obtained using the new paradigm and those obtained using a classic paradigm is investigated. The latter is used to evaluate the precedence effect at low sensation levels for six normal-hearing listeners and the results are compared to those found in section 3.1.

In *chapter 4* the role of suprathreshold coding problems in binaural processing is investigated. While measurements of the precedence effect are time consuming and require much effort of the subject, even using the new design, a more 'friendly' binaural phenomenon is chosen, namely the Binaural Intelligibility Level Difference (BILD), based on speech intelligibility measurements. *Section 4.1* considers the question whether the underlying mechanism in binaural unmasking in speech intelligibility (BILD) is the same as in binaural unmasking in speech detection, the Binaural Masking Level Difference (BMLD). The perturbation effect approach is used. The relation between BMLD and BILD is examined for six normal-hearing listeners in several conditions of signal degradation. In *section 4.2* the role of suprathreshold coding in the BILD of 25 mild to moderate hearing-impaired listeners is examined while optimal audibility is assured. In this study the distortion sensitivity approach is used

In *chapter 5* a general conclusion is given, discussing the results of the successive chapters in relation to the central issue of this thesis, i.e. to enhance our understanding of the performance of hearing-impaired listeners on spatial and binaural hearing.

Finally, in *chapter 6* the implications of the results of this research for clinical practice are discussed, and a protocol for systematic evaluation of spatial and binaural hearing will be presented.

2 Sound Localization - the relation between psychophysical tests and self-assessment



This subplot of Figure 1.4 illustrates the main themes of this chapter in relation to the general approach of the thesis. The chapter deals with overall assessment of sound localization by means of self-report and psychophysical tests. The results are related to audiometric data to investigate the role of audibility in sound localization.

2 Sound Localization - the relation between psychophysical tests and self-assessment*

In this chapter an optimal relation between psychophysical tests and subjective assessment of auditory sound localization is sought. Self-assessment by a thoroughly validated instrument (AIADH, Kramer 1995) is taken as a point of departure. Kramer et al. (1996) found a relatively weak correlation between the self-assessed localization performance and data obtained by a listening test developed by Smoorenburg and Geurtsen (1990). Several parameters in this test-design are varied, resulting in ten different localization tests. The scores of these tests were correlated with the subjective assessment for a non-homogenous group of 39 hearing-impaired listeners, including a subgroup of 23 listeners who used no hearing aids. For the test with the highest correlation, the relation to self-assessment is further elaborated. A score-measure, using false response patterns in addition to the classic percentage correct score is defined. For the subgroup of non hearing aid wearers, this results in an improvement of the correlation with the subjective score from 0.63 to 0.76.
Finally, the relation between both the psychophysically and subjectively evaluated localization performance and the pure-tone audiogram is examined. Significant relations are found. owever, for a given degree of hearing loss a large range of localization scores can be found.

** This section is based on Goverts, S.T., Kramer, S.E., Houtgast, T. (**2004**). "**Sound Localization - the relation between psychophysical tests and self-assessment**", to be submitted to Ear and Hearing.*

INTRODUCTION

Spatial hearing is an important capacity of the auditory system, which is easily affected by hearing loss. Kramer et al. (1996) found that difficulties in speech intelligibility in noise and sound localization are the most frequently reported disabilities in a group of 239 hearing-impaired subjects. Noble et al. (1995) investigated disabilities and handicaps associated with impaired auditory localization in a sample of 104 hearing-impaired patients. They found a significantly higher self-reported disability for this sample, the majority of which had a symmetrical sensorineural hearing impairment. Sound localization of the hearing impaired has been subject of several studies (e.g. Lorenzi et al., 1999; Häusler et al., 1983; Noble et al., 1994; Colburn, 1982; Rakerd et al., 1998). The overall finding is a large variation in data, ranging from close to normal to very abnormal (Moore 1995). Furthermore, localization performance turns out to be only partly predictable from pure tone audiometric data (e.g. Lorenzi et al., 1999; Noble et al., 1994).

It is important that the results of psychophysical assessment of auditory functions are in accordance with self-assessment. The psychophysical test should reflect an underlying capacity and the subjects perception of this capacity is quantified by the self-assessment. To our knowledge, the relation between self-reported and psychophysical data on auditory function has been investigated seldom. Kramer et al. (1996) examined this relation for the five factors of auditory disability, they distinguished earlier, i.e. speech intelligibility in quiet, speech intelligibility in noise, detection of sounds, distinction of sounds, and sound localization. For each of the factors a laboratory performance test was chosen that aimed to specifically measure that factor. Self-perceived disability was assessed by means of the Amsterdam Inventory of Auditory Disability and Handicap (AIADH). For a non-homogenous group of 51 hearing- impaired subjects, a maximal correlation between laboratory performance test and self report of 0.58 (Pearson's r) was found. Subjecting all data including some audiometric measures to a multiple regression analysis, multiple correlation coefficients in the range of 0.60 - 0.70 were found. The weakest multiple correlation (0.60) was found for auditory sound localization. To measure sound localization performance, they used a test design that Smoorenburg and Geurtsen developed to measure localization of warning signals (1990). This design is based upon identification of one out of eight loudspeakers in a 360 degree configuration, producing a complex tone burst. Inspection of the relation between the psychophysical and subjective data yielded an asymmetric pattern scatterplot. There was an amount of subjects who had quite good scores on the psychophysical test, even though they reported considerable difficulties with localizing sounds in daily life. This was confirmed by informal comments of the subjects. However, none of the subjects with positive self-report had low scores on the psychophysical test.

From a scientific perspective the relation between the subjective and psychophysical modalities is looked upon in terms of correlations for groups of subjects. Hyde (2000) states that validity coefficients (Pearson's r values) usually take values in the range 0.2-0.6. Even outcome measures with a modest validity coefficient of about 0.3-0.4 can have substantial utility at the group level. Thus, a multiple correlation of about 0.60 can yield a sufficient utility at the group level. However, in clinical practice where capacities of an individual patient are assessed, the necessity of agreement between the results of assessment is more urgent. In the medico-legal context for example, self-assessment should be accompanied by a sufficiently valid psychophysical test, in order to reduce the subject's influence on the result of assessment.

In the present study we seek to improve the correlation between the results of subjective and psychophysical assessment of sound localization as reported by Kramer et al. (1996). The underlying reason for this sub-optimal correlation may be a lack of validity and/or reliability of the outcome measures used (Hyde, 2000). Hyde also states that attributes like validity and reliability are not intrinsic or invariant but depend on the application, purpose and context. Actualization of both the validity and reliability of the AIADH has been established in various studies involving different groups of hearing-impaired people (Kramer et al., 1995, 1996, 1998; Meijer et al., 2003; Joore et al., 2003). Lack of validity of the psychophysical test may be due to the limited relevance of laboratory measures. Those measures may not at all be representative for the situation as experienced in daily life and may therefore lack predictive validity. The bias of auditory research to make abstractions from dirty everyday conditions is identified by Plomp (2002), stressing the importance of making a contribution to 'a better insight into the perception of everyday sounds'. However, measuring a subject's ability to localize sounds in everyday settings like an office, a street and a shopping mall would maybe yield a valid, but certainly not a reliable test.

The aim of the current study is designing a psychophysical test for localization which is more representative for daily life. Validity and reliability of the AIADH are presumed to be fairly optimal. Therefore, in the present study the self-assessment by the AIADH is taken as a gold standard. We tried to optimize the agreement of the psychophysical test with this gold standard by varying several parameters of the test design (type of signal, type of masker, S/N ratio, reverberation). Finally, we examine to what extent results of both gold standard and optimized psychophysical test are correlated to the pure tone audiogram.

METHOD

Subjects

Thirty-nine hearing-impaired subjects (mean age 49 ranging from 17 to 67) participated in this study. Sixteen of them were hearing aid users. When subjects used hearing aids, both subjective assessment and laboratorium test score were based on their performance with hearing aids. While we assume that by the use of hearing aid some uncertainties are introduced in both outcome measures, most analyses will be done for the total group of 39 subjects and for the subgroup of 23 non-hearing aid wearers (mean age 49 ranging from 17 to 67). Both total group and subgroup were rather inhomogeneous with respect to type, degree, symmetry and configuration of the hearing loss. For the total group the mean threshold at 500, 1000 and 2000 Hz for both ears was 48 dB, ranging from 12 to 106 dB. For the subgroup the mean threshold was 39 dB, ranging from 12 to 78 dB. The mean difference between the ears was 19 dB, ranging from 0 to 105 dB (two subjects with a unilateral deaf ear were included). Fifteen normal-hearing subjects (mean age 36, ranging from 25 to 58) served as a control group for the psychophysical tests in the 10 conditions. Their pure tone air conduction thresholds did not exceed 15 dB HL at any octave frequency from 250 to 8000 Hz.

Psychophysical tests

In all psychophysical tests the subject was surrounded by eight loudspeakers at a distance of 1.5 m in a large room ($200m^3$) as schematically illustrated in Figure 2.1. The reverberation

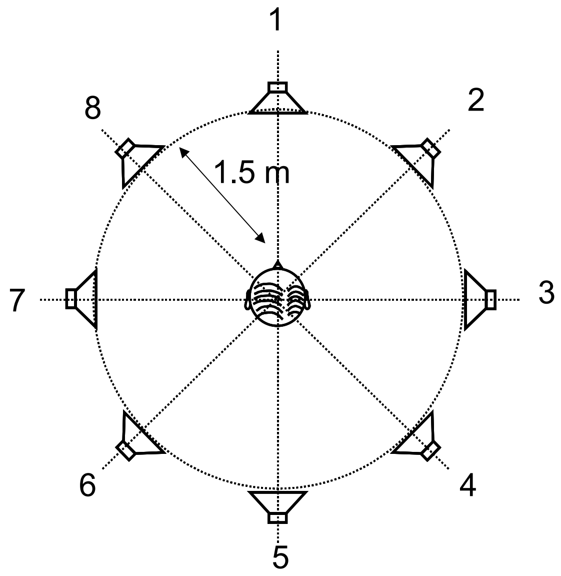


Figure 2.1 Schematic illustration of test-setup. A ninth speaker was positioned 1.5 m above the subject's head.

time of the room could be varied from 0.5 to 2.3 s. A ninth loudspeaker was positioned 1.5 m above the subjects head in order to create a non direction-specific sound field. Signals (S) were presented randomly through one the eight surrounding loudspeakers, while masking noise (N) was presented by the ninth one. The subjects task was to indicate the perceived direction of the signal, using a response box.

The initial test-configuration was adapted from Smoorenburg and Geurtsen (1990), who investigated the influence of wearing earplugs and an earmuff on sound localization of warning signals in subjects with near normal hearing and subjects with a noise-induced hearing loss. In that design, signals were complex tones, consisting of ten harmonics added in sine phase, with amplitudes decreasing with 6 dB/octave. The duration of the tone bursts was 300 ms. The masking noise was pink noise. The levels of signal and noise were 85 and 70 dB(A) respectively, chosen in accordance with the initial goal of the test, i.e. localization of warning signals. The reverberation time was 0.5 s. Signals were presented in a random sequence over the eight loudspeakers and eight fundamental frequencies of the harmonic complex. There was a fixed time interval between the subject's response and the next stimulus presentation, so the subjects could expect the stimulus. The test consisted of 64 presentations. The percentage correctly localized signals was taken as the outcome measure. This condition being the point of departure of the present study, for nine subsequent conditions the parameters type of signal, type of masker, signal-to-noise ratio, amount of reverberation and time-interval between response and next stimulus are varied. While we assumed that a major cause for the lack of correlation between self-assessment and psychophysical assessment was ‘the abstraction from “dirty” everyday conditions’ (Plomp, 2002) of the laboratory test, we aimed to make it more realistic by introducing changes in the domains of signal, masker, signal to noise ratio and time-interval between stimuli. Everyday sounds were chosen as signal and masker: a barking dog, big-ben, birds, a telephone, sirens, a laughing child, a guitar and pouring water. The duration of the everyday sounds was between 1 and 2 s. The telephone was selected as the target signal. The masking noise was created by mixing the seven other sounds and generating them in a random succession, while each sound was generated about 0.5 s after onset of the preceding one. The resulting masking noise was presented through the ninth loudspeaker. Also the fixed time interval between a subject’s response and the next stimulus presentation was considered to be not-representative for everyday life. Therefore a ‘surprise effect’ was introduced by means of alternating the target signal, presented through one of the eight loudspeakers, randomly by one of the seven masking sounds. The test consisted of 64 presentations, 32 of which contained the target signal, equally balanced over the eight loudspeaker positions. The signal-to-noise ratio (S/N) was varied over -15, 0 and +15 dB, by changing the signal level and keeping the level of the masking noise constant at about 70 dB(A).

Table 2.1 List of ten conditions as used in this study.

Condition	Target Signal (S)	Masking noise (N)	S/N [dB]	Reverberation [s]
1	Complex tones	Pink noise	+15	0.5
2	Complex tones	Pink noise	0	0.5
3	Telephone	Everyday sounds	+15	2.3
4	Telephone	Everyday sounds	+15	0.5
5	Telephone	Pink noise	+15	0.5
6	Telephone	Everyday sounds	0	2.3
7	Telephone	Everyday sounds	0	0.5
8	Telephone	Pink noise	0	0.5
9	Telephone	Everyday sounds	-15	2.3
10	Telephone	Everyday sounds	-15	0.5

For some patients that were not able to detect the signals at a level of 55 dB(A), the level of the masking noise was raised to 75 dB(A). The reverberation time was varied over the values 0.5 and 2.3 s, again to mimic several relevant daily life conditions.
Thus, we had two types of signal, two types of masker, three values for S/N ratio, two values for reverberation time. A complete design to evaluate the value of these parameters in improving the relation with self-assessment would consist of 2*2*3*2 = 24 conditions. For pragmatic reasons (mainly time of the subject's) it was chosen to perform measurements in a subset of ten conditions was chosen, as listed in Table 2.1, without retest.

Subjective evaluation

The subscale for 'Auditory localization' of the Amsterdam Inventory of Hearing Disability and Handicap (AIADH, Kramer et al. 1995) is used as gold standard, to optimize the configuration of the psychophysical test. The AIADH consists of 30 questions addressing activity limitations in everyday listening situations. Each question is accompanied by a graphical illustration to enhance the validity of the instrument. A four - point scale is used with categories 'almost never', 'occasionally', 'frequently' and 'almost always', coded as 3, 2, 1, 0 respectively. Five subscales are identified, one of which is auditory localization. Kramer et al. (1995) found a Cronbach alpha value of 0.88 for this subscale. The five questions constituting this subscale are listed in Table 2.2.

Table 2.2 The five questions constituting the "Auditory localization" with factor loadings (Kramer et al. 1995).

Item	Question	Factor Loading
3	Do you immediately hear from what direction a car is approaching when you are outside?	0.81
9	Can you hear from what corner of a lecture room someone is asking a question during a meeting?	0.65
15	Do you immediately look in the right direction when somebody calls you in the street?	0.87
21	Can you hear from what corner of a room someone is talking to you being in a quiet house?	0.66
27	Do you hear from what direction a car horn is coming?	0.8

Table 2.3 Overview of Cronbach alpha values for self-assessment and psychophysical scores for the total group of hearing impaired and both subgroups. The maximal correlation to be observed, that can be estimated on the basis of the values, is also given.

	Total group N = 39	Hearing aid N = 16	No hearing aid N = 23
Reliability of self-assessment score (Cronbach alpha)	0.85	0.72	0.91
Reliability of psychophysical score (approximated Cronbach alpha)	0.94	0.93	0.94
Maximal correlation to be observed (if validity were optimal)	0.89	0.81	0.92

Statistics

To answer the question which combination of parameters in the psychophysical test yields the best correspondence with the gold standard, Pearson's correlation coefficients are calculated for each condition in section Results A. For all variables the normal-distribution hypothesis was tested by a Kolmogorov-Smirnov test. To interpret these correlations, we relate them to an estimation of the optimal correlation to be expected. Both inventory and psychophysical test have their reliability and validity (Johnson and Danhauer, 2002; Hyde, 2000). The true correlation (r_{true}) between psychophysical test and the self-assessment will be attenuated by the non optimal reliability of both measures. If the reduced reliablity is caused by measurement error, the relation between the observed correlation ($r_{observed}$) and the true correlation is given by

$$r_{observed}^2 = \alpha_{m1} * \alpha_{m2} * r_{true}^2 \qquad (1)$$

where α_{m1} and α_{m2} indicate the reliability of the outcome measures, expressed by the Cronbach alpha values for example (e.g. Guilford, 1954; Nunnally, 1967).

The reliability of the self-assessment can be indicated by calculating Cronbach alpha using the five questions constituting the localization subscale. Because no test-retest data for the psychophysical test are available, data of two highly correlated conditions are used to calculate Cronbach alpha, namely conditions 4 and 7 (Pearson's correlation coefficient 0.90). Based on formula (1) we can estimate maximal values of the observed correlation, assuming a true correlation of 1, i.e. optimal validity of both outcome measures (see Table 2.3). These values are the upper limits of the real observed correlations that will be reported in the results section. All statistics were done using SPSS.

RESULTS

The results section is divided in two sections. Section A is devoted to the central question of the optimal relation between psychophysical tests and self-assessment of sound localization. Having thus established a listening test to measure sound localization in a valid way, we investigate the relation between the pure tone audiogram and sound localization assessed either subjectively or psychophysically.

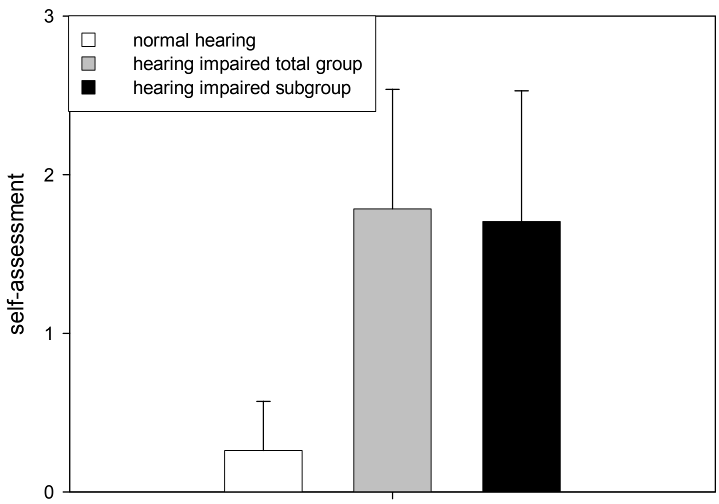


Figure 2.2 Average results of self-assessment scores, determined by the mean score on the five questions on auditory localization (Table 2.2) without factor loading for the three groups as indicated. The error bars represent standard deviations.

A Optimal relation between psychophysical tests and self-assessment of auditory sound localization

A.1 Finding the 'best test'.

In Figure 2.2 average results and standard deviations of self-assessment scores on auditory localization are given for the total group of 39 hearing impaired and the more homogenous subgroup without a hearing aid. Normal hearing data are given as a reference. In Figure 2.3a an overview of the eight by eight stimulus-response matrix is given. *Correct responses* are identified by a black circle. Figure 2.3b shows a typical response pattern for condition 7, in which the surface of the bullets represents the percentage of a certain stimulus-response combination. Correct responses can be observed, more for frontal and lateral stimulus directions (1, 2, 3, 7, 8) than for the backward directions (4, 5, 6). Figure 2.4 presents average percentage correct scores on the psychophysical test in the ten conditions for the total group of 39 hearing impaired and the more homogenous subgroup without a hearing aid. Normal hearing data are given as a reference. From this Figure, it is clear that the hearing impaired performance is worse than normal hearing performance in all conditions. Normal hearing reach maximal scores of more than 80% in conditions 4 and 7, i.e. using everyday sounds as signal and masker without reverberation, with S/N ratios of +15 and 0 dB respectively. Scores

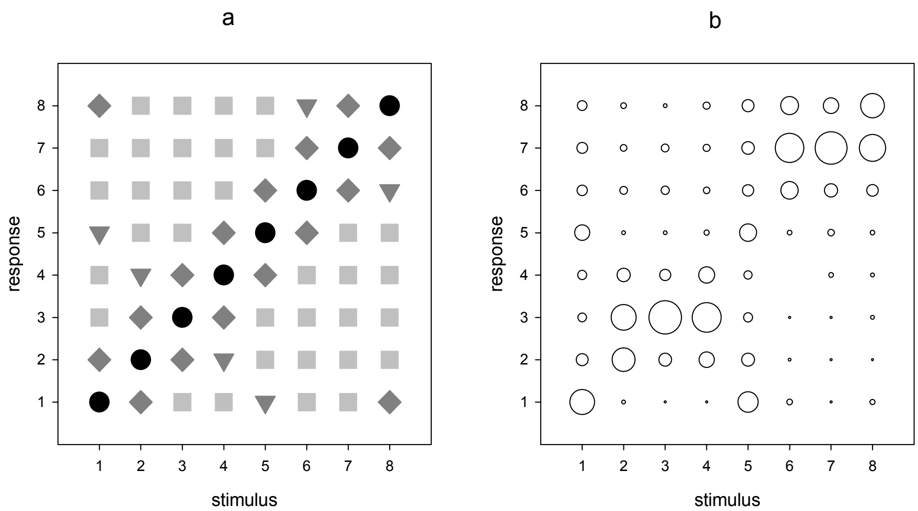


Figure 2.3 In panel a an overview of the eight by eight stimulus-response matrix is given. Correct responses are represented by a bullet. localization inaccuracies in which the response is a direction directly adjacent to the stimulus direction, named *dispersions* are indicated by grey diamonds and confusions that are mirroring the direction of presentation in the frontal plane, named *front-back confusions* are indicated by diamonds. In panel b, as an illustration the mean response pattern for condition 7 (see Table 2.1) is given. The surface of the bullets represents the percentage of a certain stimulus-response combination.

decrease by adding reverberation (conditions 3 and 6) or by replacing the everyday sounds by pink noise (conditions 5 and 8). The latter is not the case for the hearing-impaired listeners. Normal-hearing listeners seem to benefit form masker fluctuation as in speech perception, while the hearing-impaired listeners lack this advantage of a fluctuating over a steady masker (e.g. Festen and Plomp, 1990; Bronkhorst and Plomp, 1990).

In Figure 2.5 the correlation between the as gold standard adopted self-assessed localization score and the percentage correctly localizing sounds is plotted for all test conditions and for two groups, namely all hearing impaired and the subgroup of hearing impaired without hearing aid. For the total group and the subgroup, largest correlations of -0.52 and -0.63 respectively are found for condition 7. It can be intuitively understood that this condition 7 in which target signal and masker consist of everyday sounds that are presented at the same level represents a realistic acoustic situation .

A.2 Optimizing the localization score-measure for the 'best test'

The data presented in Table 2.3 indicate that, given the reliability of both outcome measures, the maximal correlation that could be expected for the total group and the subgroup without hearing aid is 0.89 and 0.92 respectively. Comparing these data to the observed correlations (Figure 2.5) yields that, having chosen condition 7 as the 'best test', there is still room for a higher correlation. An attempt is made to further improve the correspondence between the

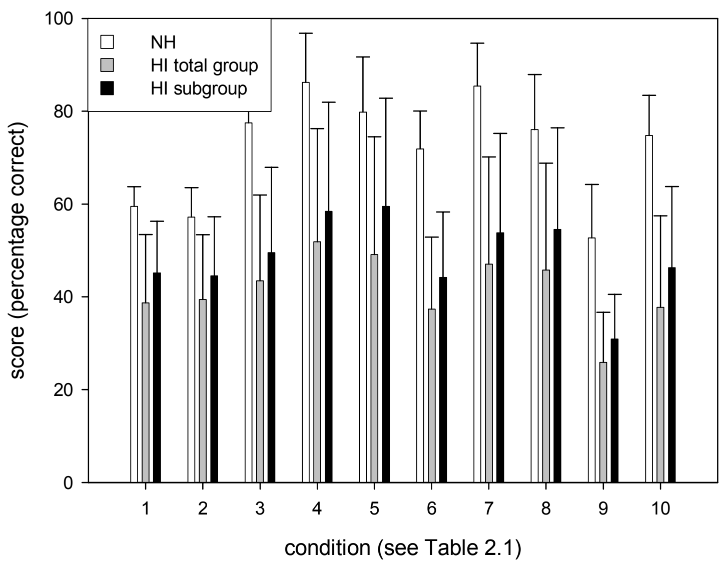


Figure 2.4 Average percentage-correct scores and standard deviations for the psychophysical tests in the ten conditions for the groups as indicated.

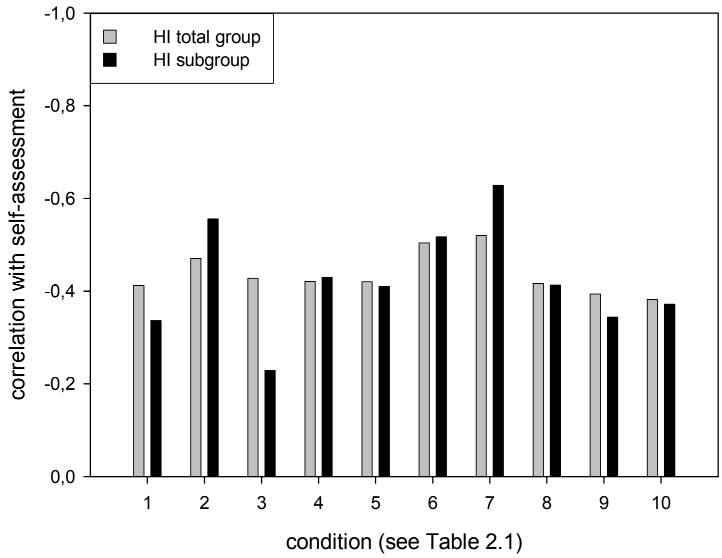


Figure 2.5 The correlation (Pearsons correlation coefficient) between the as gold standard adopted self-assessed localization score and the psychophysical tests in the ten conditions for the groups as indicated.

psychophysical test and the self-assessment by defining another score-measure, based on the response pattern (see Figure 2.3a).

A straightforward approach is to give an individual weighting factor to all 64 cells of stimulus-response combinations or to subgroups of the 64 cells and optimize those factors in order to reach optimal correlation between subjective and psychophysical data for the 39 subjects. Applying a split-half approach to the results of this approach showed that this method lead to non-generic solutions. A correspondence analysis (Greenacre, 1993) was performed to look for structures in the data. This basically did lead to two types of relevant stimulus-response combinations in addition to the correct responses: localization inaccuracies in which the response is a direction directly adjacent to the stimulus direction, named *dispersions* (indicated by grey diamonds in Figure 2.3a), and confusions that are mirroring the direction of presentation in the frontal plane, named *front-back confusions* (indicated by triangles in Figure 2.3a). Subjecting the scores on those false-response patterns to a regression analysis with the gold standard as dependent variable results in an increase of correlation. Applying the coefficients of the regression model to the false pattern scores yields a new score-measure, the false response pattern for condition 7 (FRP7).

Table 2.4 Pearson's correlation coefficients between the gold standard and several score-measures for the total group and the subgroup of hearing impaired. Significance at the $p<0.05$ and $p<0.01$ level is indicated by * and ** respectively.

Correlation with self-assessment score	Total group N = 39	No hearing aid N = 23
C1 (correct score condition 1)	-0.41**	-0.34
C7 (correct score condition 7)	-0.52**	-0.63**
DISP7 (dispersions condition 7)	0.44**	0.67**
FB7 (Front-back confusions condition 7)	0.24	0.20
FRP7 (combination of DISP7 and FB7 using coefficients of regression model)	0.54**	0.76**

Table 2.4 presents the Pearson's correlation coefficients between the gold standard and several score-measures for the total group and the subgroup of hearing impaired. The correlation between the gold standard and the new score-measure FRP7 amounts 0.54 for the total group and 0.76 for the homogeneous subgroup. For the interpretation of the correlation between the false pattern based score-measure and the self-assessment, it is related to the estimated maximal correlation to be observed, given the reliability of subjective and psychophysical tests. The reliability of the self-assessment data is given in the Method section. For FRP7, it is estimated by calculating Cronbach alpha of FRP7 and FRP4 (derived by applying the coefficients of the regression model on the data of condition 4). This results in a maximum of $r = 0.84$ for both the total group and the subgroup. This implies that for the subgroup the results of the newly established measure correspond fairly well to the gold standard. The difference between the results for the total group and the subgroup without hearing aid is an illustration of Hyde's (2000) remark on the dependence of reliability and validity of outcome measures on the application, purpose and context.

B The relation between the results of sound localization assessment and the pure tone audiogram.

As a sideline, the relation between audiometric thresholds and the subjective and psychophysical outcome measures is examined. This is of interest in order to understand the effect of hearing impairment on activity limitation in the domain of sound localization. Inspecting the Pearsons's correlation coefficients between the subjective and psychophysical outcome measures and audiometric thresholds at the octave frequencies 500, 1000, 2000, 4000 and 8000Hz shows that all outcome measures are mainly correlated to low frequencies. Therefore, the mean threshold over 500, 1000 and 2000 Hz, often referred to as Pure Tone Average (PTA) is taken as audiometric measure, averaged over both ears, for the better and the poorer ear, respectively. Correlation coefficients (Pearson's r) are listed in Tables 2.5 and 2.6 respectively and illustrative scatter plots are given in Figure 2.6.
This Figure indicates that results of psychophysical and subjective assessment vary from normal to very deviant (more than 2 times the standard deviation, see also Figures 2.2 and 2.4) for patients with a mild to moderate hearing loss. An overall finding is that the mean threshold of the *better* ear is not correlated to any of the outcome measures. Subjective data show a limited correlation of about 0.60 with the mean threshold averaged over *both* ears. This correlation is not improved by taking the poorer ear. No large difference between the total group and subgroup data is found. For the correlation with the psychophysical data based on the percentage correct response (C7), the choice of the poorer ear's thresholds yields a substantial improvement compared to the average of both ears.

Table 2.5 Pearson's correlation coefficients between the self-assessment scores and several mean threshold values for the total group and the subgroup of hearing impaired. Significance at the $p<0.05$ and $p<0.01$ level is indicated by * and ** respectively.

Correlation of mean threshold (500, 1000, 2000 Hz) with self-assessment score	Total group N = 39	No hearing aid N = 23
Average of both ears	0.56**	0.60**
Better ear	0.35*	0.29
Poorer ear	0.57**	0.53**

Table 2.6 Pearson's correlation coefficients between the psychophysical data based on the percentage correct response (C7) as well as based on the false response patterns (FRP7) and several mean threshold values for the total group and the subgroup of hearing impaired. Significance at the $p<0.05$ and $p<0.01$ level is indicated by * and ** respectively.

Correlation of mean threshold (500, 1000, 2000 Hz) with psychophysical score	Total group N = 39		No hearing aid N = 23	
	C7	FRP7	C7	FRP7
Average of both ears	-0.59**	0.35*	-0.57**	0.48*
Better ear	-0.18	0.16	0.17	0.13
Poorer ear	-0.74**	0.39*	-0.75**	0.48*

Again, no large difference between the total group and subgroup data is found. The correlation of the psychophysical outcome measures based on the false response patterns (FRP7) are in the same range as the results for the subjective measure. Here the choice of the poorer ear doesn't improve the correlation, while the subgroup shows a better correlation than the total group.

DISCUSSION AND CONCLUSION

In this study an optimal relation between psychophysical and subjective assessment of sound localization is sought, taking the self-assessment by means of the AIADH for a non-homogenous group of 39 hearing-impaired subjects, as a point of departure. This type of research on validation in audiology has not been reported often, to our knowledge. The correlation between subjective assessment and psychophysical test can be improved from 0.40 to 0.52, by choosing a design corresponding to a realistic everyday acoustic situation. A major difference between this design and the test developed by Smoorenburg and Geurtsen (1990) are the choice for everyday sounds as target signal and masking noise in stead of the more artificial complex tone bursts and pink noise. Also the S/N ratio of 0 dB and the

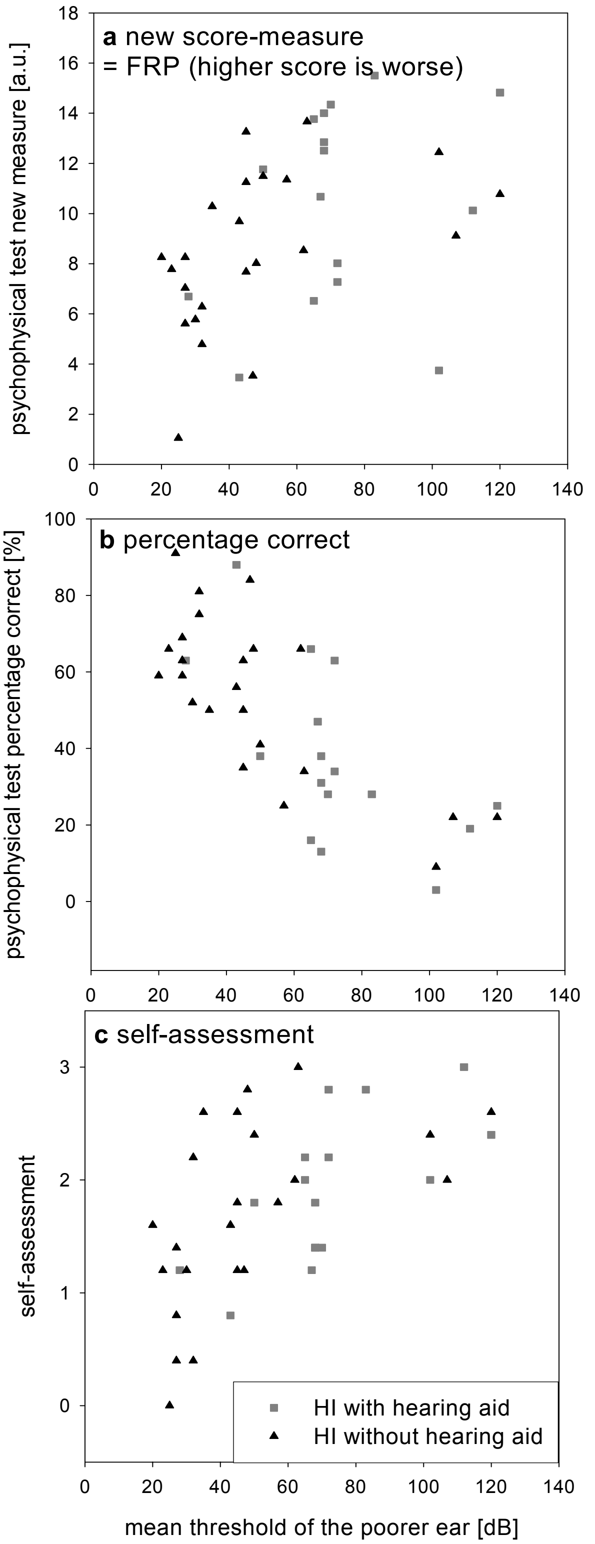


Figure 2.6 Scatter plots of subjective and psychophysical data on sound localization versus the mean hearing threshold at 500, 1000 and 2000 Hz of the worse ear for the 39 hearing-impaired listeners.

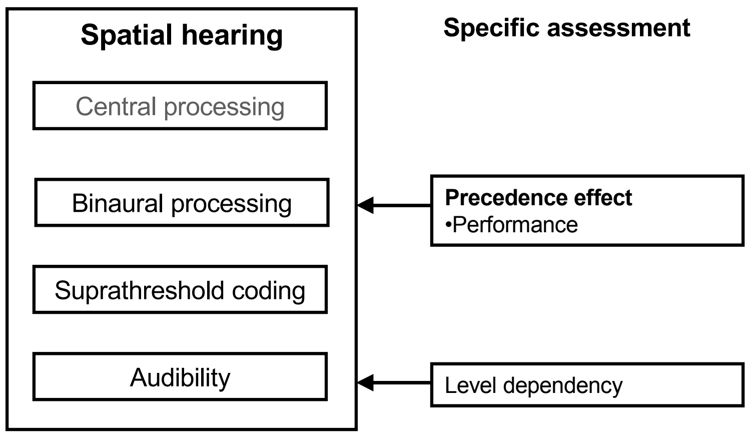
introduction of uncertainty in the time-interval between two target signals are more typical for daily life situations. For the more homogenous subgroup the absolute value of the correlation is improved from 0.38 to 0.63 by choosing this design. Using an alternative score-measure based on dispersions and front-back confusions, the correlation is optimized to 0.54 for the total group and even to 0.76 for the subgroup. Comparing this to the maximum correlation, given the attenuation of the true correlation by the reliability of the tests, shows that a sufficient validity has been established for the subgroup. Thus, it turned out to be possible to define a listening test to assess overall sound localization performance, in reasonable agreement with self-assessment scores. Subjective localization data are correlated with the mean hearing thresholds of both ears over 500, 1000 and 2000 Hz (0.56). The psychophysical localization data based on percentage correct response, are correlated with the mean hearing thresholds of both ears over 500, 1000 and 2000 Hz (0.57). Taking only the thresholds for the poorer ear yields a substantial improvement (0.74), while taking only the better ear's thresholds yields no correlation at all. It is remarkable that C7 is more correlated to the audiometric measure than the new score-measure, FRP7. This suggests that FRP7 in stead of C7 might be of additional value to the Pure Tone Average. Only for the total group this is confirmed by a regression analysis with the subjective score as the dependent and the PTA and FRP7 as explanatory variables, yielding a multiple correlation coefficient of 0.66, which is an improvement (see Tables 2.4 and 2.5).

Despite the correlations that are listed in Tables 2.5 and 2.6, given a certain degree of hearing loss a large range of localization scores can be found. The limited relation between psychophysical test and audiometric data might be caused by suprathreshold problems or central problems as was also suggested e.g. by Lorenzi et al. (1999) and Noble et al. (1994). In clinical practice pure tone audiometry should be accompanied by subjective assessment of sound localization. The test-design and score-measure presented in this article are shown to be fairly valid and can contribute in those cases where psychophysical data are needed.

ACKNOWLEDGEMENTS

The authors thank Gaston Hilkhuysen for contributing to statistical considerations and Hans van Beek for technical assistance.

3 The role of audibility in the precedence effect



This subplot of Figure 1.4 illustrates the main themes of this chapter in relation to the general approach of the thesis. The chapter deals with specific assessment of the precedence effect. The role of audibility in the precedence effect is investigated by means of the level-dependency approach.

3.1 The precedence effect for lateralization at low sensation levels*

Using dichotic signals presented by headphone, stimulus onset dominance (the precedence effect) for lateralization at low sensation levels was investigated for five normal hearing listeners. Stimuli were based on 2400-Hz lowpass filtered 5-ms noise bursts. We used the paradigm, as described by Aoki and Houtgast (1992) and Houtgast and Aoki (1994), in which the stimulus is divided into a leading and a lagging part with opposite lateralization cues (i.e. an interaural time delay of 0.2 ms). The *occurrence* of onset dominance was investigated by measuring lateral perception of the stimulus, with fixed equal duration of leading and lagging part, while decreasing absolute signal level or adding a filtered white noise with the signal level set at 65 dBA. The dominance of the leading part was *quantified* by measuring the perceived lateral position of the stimulus as a function of the relative duration of the leading (and thus the lagging) part. This was done at about 45 dB SL without masking noise and also at a signal-to-noise ratio resulting in a sensation level of 10 dB. The occurrence and strength of the precedence effect was found to depend on sensation level, which was decreased either by lowering the signal level or by adding noise. With the present paradigm, besides a decreased lateralization accuracy, a decrease in the precedence effect was found for sensation levels below about 30-40 dB. In daily-life conditions with a sensation level in noise of typically 10 dB the onset dominance was still manifest, albeit degraded to some extent.

* *This section is based on Goverts, S.T., Houtgast, T. and Beek, J.H.M. van (2000). "**The precedence effect for lateralization at low sensation levels**." Hearing Research 148, 88-94*

I INTRODUCTION

The precedence effect, in short, the dominance of the first arriving acoustical information in signal perception, plays an important role in spatial hearing. Wallach et al. (1948) found that the localization of two brief stimuli that are perceived as one fused sound is determined largely by the location of the first sound. Haas (1951) investigated the influence of a single echo on the intelligibility of speech. He found that echo's with a delay of 1-30 ms are suppressed; their intensity needs to exceed that of the primary sound by 10 dB before it is perceived separately. Blauert (1997) gives a review on the precedence effect including both classic data and more recent work on the build-up phenomenon. Recently Litovsky et al. (1999) published an extensive review of psycho-acoustical and physiological data on this topic with the primary goal to 'carefully delineate between studies that claim to measure "the precedence effect."' They distinguish three phenomena: fusion, localization dominance, and lag discrimination suppression, and give an overview of free field and headphone studies. The present study is related mostly to localization dominance: the every day experience that the directional impression of a sound source is little influenced by its reflections. This phenomenon of the precedence effect is operationalized in the laboratory by simulating a source and its reflection and investigating the localization of the fused image. Localization dominance is strongest at delays in the range between 1 millisecond and echo threshold. Zurek (1980,1987) found that the sensitivity to interaural differences in both time and intensity follows a non-monotonic course after the abrupt onset of a sound. He found that the sensitivity is reduced for a period of approximately 0.5 to 10 ms after onset. Zurek hypothesizes that this temporary lapse of interaural sensitivity serves to avoid interaural ambiguities. He proposes a model in which dichotic information is multiplied by an inhibition function I(t), which is triggered by an onset detector. Aoki and Houtgast (1992), and Houtgast and Aoki (1994) investigated the perceptual weight of dichotic information as a function of time-after-signal-onset. For three types of interaural cues, i.e. interaural time-delay, level difference and cross-correlation, they found weighting functions with the same overall shape. The shape of these functions shows a peak for the first few milliseconds, followed by a period of about 20 milliseconds of reduced weight. The shape of this weighting function is in line with the inhibition function in the model of Zurek. The paradigm used by Aoki and Houtgast

(1992), and Houtgast and Aoki (1994) is also used in the present study (see Method section). Using noise stimuli presented by headphones it provides a method to quantify the underlying mechanism of localization dominance (or rather lateralization dominance) avoiding the subjective nature that is usually inherent to measurements of this effect of precedence (Litovsky et al, 1999).

With the future intent to investigate the precedence effect for hearing-impaired subjects, the goal of this study is to determine the influence of sensation level on the precedence effect for normal hearing listeners. From the studies mentioned above the relevance of abrupt onsets for the precedence effect is evident. It also arose from Abel and Kunov (1983), who investigated the effects of shape of rise/decay and amplitude on the lateralization of pure tones at levels ranging from 60 to 80 dB. Furthermore, Rakerd and Hartman (1986) investigated the influence of onset rate on the precedence effect for localization using tones with different amplitude envelopes at steady state levels of 40 and 65 dBA. They concluded that onset rate, rather than the absolute signal level, determines the onset dominance. Shinn-Cunningham et al. (1993) reported a relatively minor effect of changing the overall stimulus level from 80 to 110 dB on localization dominance and lag discrimination suppression for noise bursts with different interaural lags.

In the present study a first approach is made to answer the question how the precedence effect depends on absolute signal level and on signal-to-noise ratio for sensation levels between 0 and 63 dB. This approach is limited to localization dominance, especially to temporal lateralization cues for brief noise stimuli presented by headphones.

II METHOD

Subjects

Five normal-hearing subjects participated in the study. They all had hearing thresholds below 20 dB in the frequency range 0.25 to 4 kHz. The range of age was 27-36. Two subjects, referred to as N1 and N3, were not familiar with psycho-acoustic experiments. The other subjects had some experience with this kind of tests. One of them (N4) was the first author.

No systematic difference in the performance was observed between the less experienced and the more experienced groups.

Stimuli

In this study we used a test design as described by Houtgast and Aoki (1994). In this design, the signal is based on lowpass filtered white noise bursts, with a cutoff frequency of 2400 Hz and a duration T of 5 ms. For each trial a different noise burst is drawn randomly from the digitally stored lowpass filtered noise. The masking noise is lowpass filtered white noise with the same cutoff frequency. A single *signal presentation* consists of a dichotic presentation of this noise burst (see Figure 3.1.1). The 5-ms signal presentation is subdivided in two parts, one part before Ts (leading part) and one after Ts (lagging part). In the leading part the noise burst is delayed 0.2 ms in one ear relative to the other. In the lagging part this delay is introduced in the other ear.[1] Thus the signal contains two opposite temporal cues for lateralization, C1 and C2. The crucial parameter is the switch time Ts, i.e. the time after signal onset at which cue C1 turns into cue C2. This switch time will be expressed relative to the

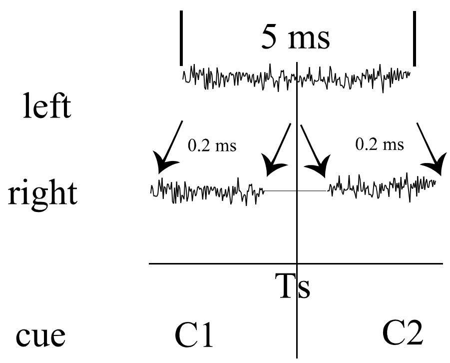


Figure 3.1.1 Schematic representation of a single signal presentation, consisting of a dichotic presentation of a low pass filtered white noise burst. The burst is subdivided in two parts, containing opposite temporal cues for lateralization, C1 and C2. The cue was always an interaural time difference of 0.2 ms. The crucial parameter is the relative switch time Ts, i.e. the time after signal onset at which cue C1 turns into cue C2, expressed relative to the total duration T. In this example Ts = 0.5, C1 = ‘right leading’, C2 = ‘left leading’.

[1]By introducing the interaural time differences, for one of the ears a small 0.4 ms gap is introduced, which is below gap detection threshold (e.g. Snell et al. 1994). This gap is enlarged on the schematic representation in Figure 3.1.1.

total duration T (i.e. Ts between 0 and 1). The brief 5-ms signal presentation is perceived as one fused sound image with a specific lateral position. This position depends on Ts. For Ts values of 0 or 1, i.e. at the beginning or at the end of the signal, there is no switching of cues during the signal presentation, resulting in a single unambiguous cue for lateralization. By varying the value of Ts between 0 and 1 the ratio between the duration of cue C1 and the duration of cue C2 changes. The working hypothesis is that when the perceptual weight of the parts before and after Ts is equal, the opposite lateralization cues C1 and C2 will cancel each other. For that particular value of Ts, which is named Tequal (Teq), the fused image is not lateralized, i.e. it is perceived in the middle. If there was no precedence effect, or other perceptual weighting, we would expect Teq to be 0.5. Thus, any systematic effect of lateralization for Ts = 0.5 must be due to some perceptual dominance.

Each *trial* contains two signal presentations with opposite values for C1 and C2 as indicated in Figure 3.1.2 for two signals with Ts = 0.2. The subject's task is to indicate in a forced choice procedure the lateral position of the second signal relative to that of the first signal, i.e. 'left' or 'right'. No feedback is given to the subject. This wouldn't make sense in this paradigm, while there is no 'correct' response. In conditions with masking noise, this masker is presented in each trial from 1 s before the first signal presentation until 1 s after the second signal presentation.

Procedure

Two types of experiments were performed. With the first type we determined the *occurrence* of the precedence effect at various sensation levels. For stimuli with Ts = 0.5 we measured the proportion of responses in accordance with C1, thus, the cue in the first half of the signal.

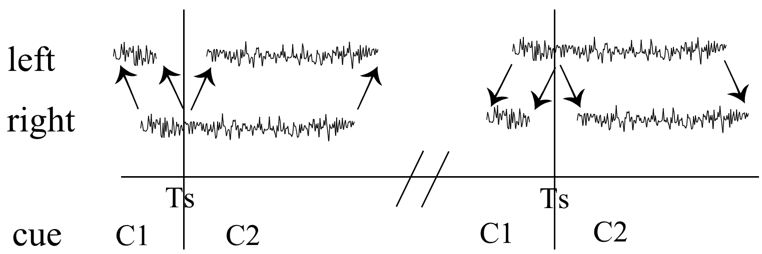


Figure 3.1.2 Schematic representation of a single trial, consisting of two signal presentations with the same value for Ts (in this example about 0.2) and opposite values for C1 and C2. In the first signal presentation C1 = 'left leading' and C2 = 'right leading', in the second signal presentation the reverse is true.

A 50 % response would indicate either that the target was not detected or that there was no systematic preference for lateralization in accordance with C1 or C2. Any response percentage significantly larger than 50 % would indicate an effect of onset dominance. With the second type of experiment we aimed at *quantifying* the precedence effect for a given sensation level by determining that value of Ts at which the responses are at the 50 % chance level (thus, the value of Teq as discussed above).

An experimental series for investigating the occurrence of onset dominance consisted of 220 trials which were balanced over 11 values of signal or masker-noise level, while keeping Ts constant at 0.5. An experimental series for quantifying onset dominance consisted of 220 trials which were balanced over 11 values of Ts in a range of [0, 0.1,...,1], while keeping the level of signal and masker-noise constant. By curve fitting and interpolation, Teq was determined as the value of Ts for which the score was 50 %, indicating that the two stimuli in a trial could not be systematically discriminated in terms of lateral position. The experimental paradigm ensured that the measurement of Teq was not biased by motivation, experimental skills, subjective aspects of judgement, or learning effects during a series. A series of 220 trials consisted of ten repeated blocks of 22 trials. Every block contained two essentially identical trials for each of the 11 parametric values; in these two trials the order of the two signals was simply reversed. Each *curve* that will be presented in the results section is based on two series, thus every point of the individual curves reflects 2*2*10 = 40 trials.

Hardware

Stimuli were digitally generated on an OROS-AU22 sound processing card, which was placed in an 286 computer. They were presented to the listener via Beyer DT48 headphones. Levels were adjusted by an in house produced programmable attenuator. Experiments were performed in a sound insulated room.

III RESULTS

Threshold

The average absolute thresholds for the 5 ms stimuli and the continuous masking noise, measured using a simple Bekesy paradigm, were 20 and 10 dBA (calibrated using an artificial ear and a flat plate coupler), respectively. Masked threshold for the signal in 75-dBA noise was 65 dBA. Little variation between the subjects was observed. Signal levels (Ls) are expressed in dB Sensation Level relative to the *mean* threshold. Signal-to-noise-ratios are expressed in dB Sensation Level relative to the thresholds in noise.

Occurrence of the precedence effect as a function of signal level

The influence of signal level on the occurrence of the precedence effect was investigated, while Ts was held constant at 0.5 and the signal level was varied over a grid of 11 equally distributed values between 3 and 63 dB SL (in 6 dB intervals). Results are given in Figure 3.1.3, which presents the proportion of responses in accordance with C1 (the cue in the first half of the signal) versus the signal level Ls. Besides median data, the individual curves for the five subjects are given. The overall performance is similar for all subjects. A decrease in onset dominance for signal levels above 50 dB SL, however, is most clearly found for subject

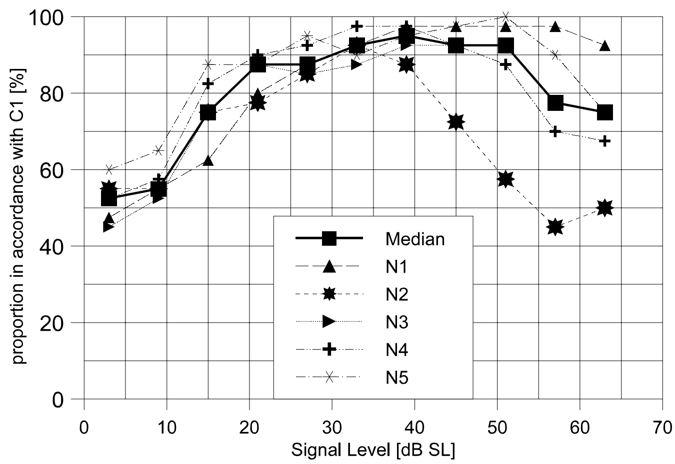


Figure 3.1.3 Proportion of responses in accordance with C1 versus signal level for stimuli without masking noise with Ts = 0.5. Next to the individual data for 5 subjects, median data are presented.

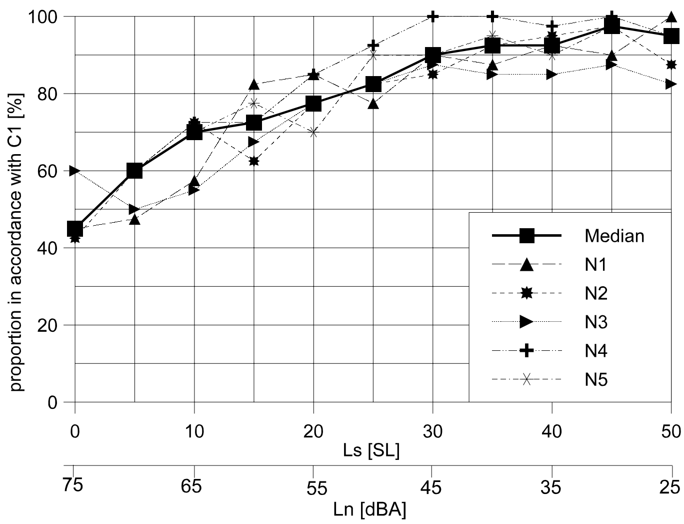


Figure 3.1.4 Proportion of responses in accordance with C1 versus noise level for stimuli with Ts = 0.5 and at a fixed signal level of 65 dBA (i.e. 45 dB SL without masking noise). The noise level is varied from 25 to 75 dBA, which implies a variation of the signal sensation level in noise of 50 to 0 dB. Next to the individual data for 5 subjects, median data are presented.

N5. All subjects except N1 show some measure of decrease above 50 dB SL. This decrease was observed in both series.

Occurrence of precedence effect as a function of signal-to-noise ratio

The influence of signal-to-noise-ratio on the occurrence of the precedence effect was investigated using the same experimental design. In this experiment, Ts and Ls were held constant at 0.5 and 65 dBA (i.e. 45 dB SL without masking noise), respectively. The level of the masking noise, Ln, was varied over a grid of 11 equally distributed values between 25 and 75 dBA (5 dB intervals), which means that we actually varied the sensation level of the signal over a range of 50 to 0 dB[2]. Results are given in Figure 3.1.4, which presents the proportion of responses in accordance with C1 versus Ln. Besides median data, the individual curves for the five subjects are given. The overall performance is similar for all subjects.

[2]Actually, the detection threshold of the signal in noise (0 dB SL) was only measured at a level of the masking noise of 75 d.b.a.; probably, the sensation level will not be exactly 50 at a 25 d.b.a. masking noise level.

Quantification of precedence effect at 45 dB SL

To quantify the precedence effect at 45 dB SL the proportion of responses in accordance with the first cue, C, was determined as a function of the switch time Ts. Results are given in Figure 3.1.5. Besides median data, individual curves for the five subjects are given. The overall performance is similar for all subjects. At Ts = 0 and 1 subjects respond 100% in accordance with C2 or C1, respectively. This is in accordance with our expectations, because these are pure lateralization conditions. The point of equal perceptual weight Teq (i.e. the value of Ts at which 50 % of responses is in accordance with C1) lies for all subjects at a relative switch time between 0.1 and 0.2. For each individual curve an estimation of Teq was made on the basis of a third order polynomial fit. These results are presented in the left-most column of Table 3.1.1.[3]

Table 3.1.1 Teq determined on the basis of a third order polynomial fit of data for lateralization in accordance with the first cue in the condition without noise (Ls = 65 dBA, = 45 dB SL) and with a sensation level of 10 dB (Ls = 65 dBA and Ln = 65 dBA).

subject	Teq rel T Ls = 65 dBA no noise 45 dB SL	Teq rel T Ls = 65 dBA Ln = 65 dBA 10 dB SL
N1	0.17	0.28
N2	0.21	0.34
N3	0.20	0.26
N4	0.15	0.33
N5	0.15	0.56
Average	0.17	0.35
Std	0.02	0.10

[3]It is remarkable that the decrease in score in accordance with C1, which is observed at 45 dB SL in Figure 3.1.3 for subject N2, is not manifest in the corresponding data point in the current data (Ts = 0.5). We have no explanation for this.

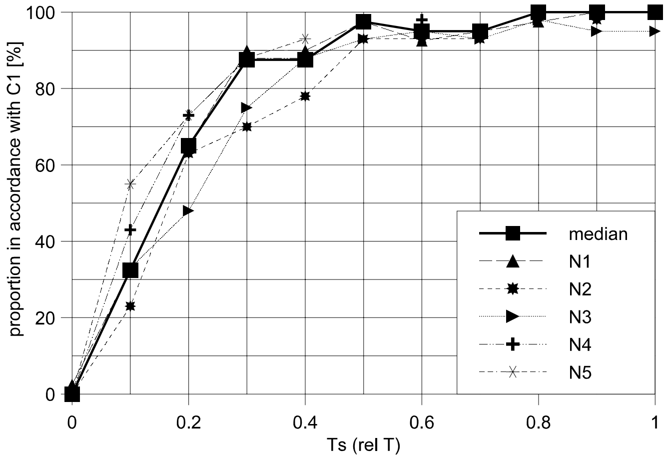


Figure 3.1.5 Proportion of responses in accordance with C1 versus the switch time Ts without masking noise with a fixed signal level of 65 dBA, (i.e. 45 dB SL). Next to the individual data for 5 subjects, median data are presented.

Quantification of precedence effect in noise at a sensation level of 10 dB

To quantify the precedence effect at a lower sensation level, the dominance of C1 as a function of Ts was determined at a signal-to-noise ratio of +10 dB (Ls = 65 dBA and Ln = 65 dBA). Results are given in Figure 3.1.6. Besides median data, individual curves for the five subjects are given. The overall performance is similar for all subjects. For each individual

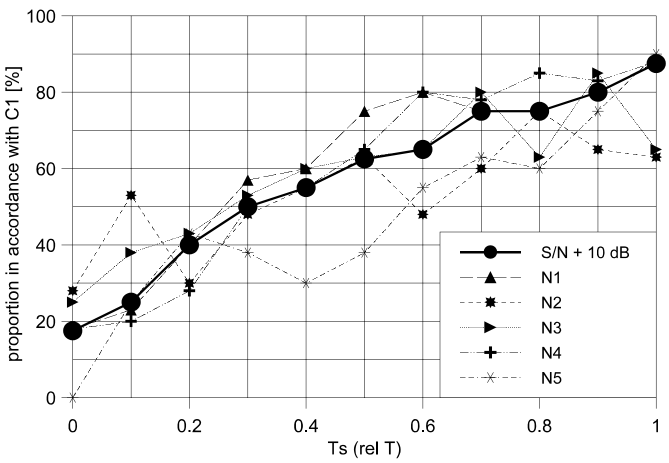


Figure 3.1.6 Proportion of responses in accordance with C1 versus the switch time Ts in noise with a fixed sensation level of 10 dB (i.e. signal level 65 d.b.a., and noise level 65 d.b.a.). Next to the individual data for 5 subjects, median data are presented.

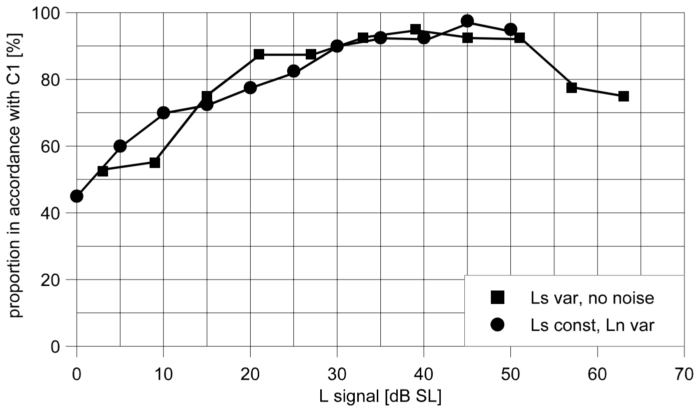


Figure 3.1.7 Compilation of median data of Figure 3 and 4. Proportion of responses in accordance with C1 versus sensation level, either by varying absolute signal level without masking noise or by varying noise level with a fixed signal level (65 dBA).

curve an estimation of Teq was made on the basis of a third order polynomial fit. The results are presented in the right-most column of Table 3.1.1. Addition of the masking noise increased the average value of Teq from 0.17 to 0.35. This implies that without noise the perceptual weight of the first (0.17*5) 0.85 ms of the signal is equal to the perceptual weight of the last (0.83*5) 4.15 ms, while at a signal-to-noise ratio of +10 dB the weight of the first 1.75 ms is equal to that of the last 3.25 ms.

IV DISCUSSION

Occurrence of precedence effect as a function of sensation level

In Figure 3.1.7 median results for the two experiments, in which sensation level is varied, are compared in order to investigate whether a general effect of sensation level on onset dominance can be established. It appears that the effects on onset dominance of varying sensation levels, either by decreasing the absolute signal levels or by raising masking-noise levels, are very similar. From 30 to 50 dB SL stimuli are mainly lateralized in accordance with the first cue, C1. Below 30 dB SL this dominance gradually diminishes to the chance performance (i.e. no dominance of either cue) as expected at 0 dB SL. This result cannot be

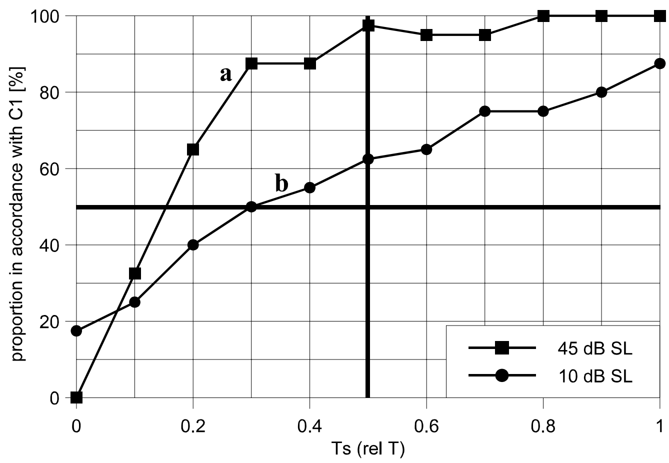


Figure 3.1.8 Proportion of responses in accordance with C1 versus Ts. Curve *a* is a replot of the median data of Figure 3.1.5 (45 dB SL). Curve *b* is a replot of the median data of Figure 3.1.6 (10 dB SL).

compared directly to the findings of Abel and Kunov (1983), Rakerd and Hartman (1986), and Shinn-Cunningham et al. (1993), as mentioned in the introduction. Besides differences in experimental setup and stimuli, the main discrepancy lies in the signal levels used in those studies, e.g. 60-80 dB SPL (Abel and Kunov), 40-65 dBA (Rakerd and Hartman) 80-110 dB SPL (Shinn-Cunningham et al.) versus 23-83 dBA in our study.

The decrease in the responses in accordance with C1 that is observed for signal levels above 50 dB SL (see Figure 3.1.3) is caused mainly by one subject's performance, though four of the five subjects show some measure of decrease above 50 dB SL. Further research will be needed to determine whether there is a general effect of high sensation signal levels on the precedence effect.

Lateralization accuracy and precedence effect

The data for the signal with Ts = 0.5, as shown in Figure 3.1.7 (i.e. the decrease of the score in accordance with C1 for sensation levels below 30 dB) do not necessarily imply a decrease of the precedence effect for lateralization at low sensation levels; these data may also simply reflect a decrease of lateralization accuracy towards low sensation levels. This issue can be clarified by considering the relation between lateralization score and Ts, as given in Figure 3.1.8 Considering the scores at Ts is 0 or 1, there is a clear difference between 45 dB SL and

10 dB SL. This must be attributed to decrease of lateralization accuracy at 10 dB SL as compared to 45 dB SL. Using a 500-Hz octave band filtered white noise signal, Houtgast and Plomp (1968) showed that the lateralization accuracy depends on signal-to-noise ratio and signal duration.

For the shortest signal duration they investigated (10 ms) the accuracy remained constant from the condition without noise until a signal-to-noise ratio of about 30 dB. From S/N = 30 downwards the accuracy decreases until the lateralization is limited by detectability. This is interpreted as reflecting the statistical fluctuations in the Inter Aural Time Delay cue caused by the relative magnitude of the internal and external noise. It is reasonable to suppose a similar effect of level on lateralization accuracy for the 5-ms stimuli used in the present study, resulting in the data as observed at Ts is 0 and 1.

However, it must be stressed that the decrease in lateralization accuracy does not explain the difference in Teq for the 45 dB SL and 10 dB SL curves. When accuracy would be the only issue, a 50% score for 45 dB SL would remain 50% at 10 dB SL. The shift of Teq from about 0.15 to 0.30 must imply that the dominance of the initial part of the signal is diminished, i.e., that the precedence effect at 10 dB SL is smaller that at 45 dB SL. Hafter (1983a, 1983b, 1988) investigated detection of interaural time and intensity differences in trains of 4000-Hz clicks by varying the inter-click interval and the number of clicks in a train. He explained the results by means of a rate-dependent saturation model, in which the evoked neural activity moves from a tonic to a phasic response as the click rate increases. These results might be applied to our data set: at low sensation levels the neural firing rate is low and consequently the neural activity is tonic, while at moderate sensation levels (about 45 dB SL) the responses turn to phasic, i.e. a reduced neural activity beyond the signal onset.

The above suggests that the decrease of scores with level at Ts = 0.5 is due to both decreased lateralization accuracy and to a decreased precedence effect. At 10 dB SL the unambiguous signal (Ts = 1 or 0) leads to a score of only about 80 %. This implies that the lateralization accuracy has decreased substantially at low sensation levels, which is in accordance with the results of Houtgast and Plomp (1968). The point at which there is no dominance of either cue, Teq, moves from 0.17 to 0.35. This reflects a decrease of the precedence effect. It can thus be concluded that, at a sensation level of 10 dB, there is still an onset dominance, albeit less strong. Following this reasoning for the mean data of onset dominance as a function of sensation level with a fixed value of Ts (Figure 3.1.7), it can be carefully concluded that the decrease of scores in accordance with C1 for sensation levels lower than 30 dB reflects both a decrease of precedence effect and an decrease of lateralization accuracy.

Results of the current study stress the importance of audibility and sensation level in the precedence effect. This is important for correctly interpreting aspects of binaural performance in hearing-impaired listeners.

ACKNOWLEDGEMENTS

We thank Joost M. Festen, Johannes Lyzenga and three anonymous reviewers for comments on an earlier version of this paper.

3.2 The precedence effect for lateralization for the mild sensory neural hearing impaired*

Using dichotic signals presented by headphone, stimulus onset dominance (the precedence effect for lateralization) was investigated for six sensorineural hearing-impaired listeners. Stimuli were based on 2400-Hz lowpass-filtered 5-ms noise bursts. We used the paradigm, as described by Goverts et al. (2000), in which a single noise burst is divided into leading and lagging parts, with opposite lateralization cues (viz. an interaural time delay of 0.2 ms). The *occurrence* of onset dominance was investigated by measuring the lateral perception of the stimulus (‘left’ or right’) with fixed, equal durations of leading and lagging parts, while decreasing the absolute signal level or adding a filtered white noise. The dominance of the leading part was *quantified* by measuring the lateral perception of the stimulus as a function of the relative duration of the leading (and thus the lagging) part. This was done at about 40 dB SL in quiet and in filtered white noise, at a signal-to-noise ratio resulting in a sensation level of about 6 dB. Results are compared to normal hearing reference data at various sensation levels. Hearing impaired data show a large variance and overall a decreased precedence effect in terms of both occurrence and quantification, which cannot be explained on basis of reduced audibility. Mean performance of the hearing-impaired subjects at 40 dB in quiet was similar to normal hearing performance in masking noise at a signal-to-noise ratio of 0 dB.

** This section is based on Goverts, S.T., Houtgast, T. and van Beek, J.H.M.(2002). “**The precedence effect for lateralization for the mild sensory neural hearing impaired,**” Hearing Research 163, 82-92*

I INTRODUCTION

Reduced understanding of speech in noise and sound localization are major problems experienced by hearing impaired (for example: Kollmeier, 1997; Koehnke and Besing 1997; Kramer, 1998a). In both auditory functions spatial and binaural hearing play a significant role. A large amount of experimental data on both spatial and binaural hearing reveal a reduced overall performance of hearing-impaired subjects compared to normal hearing. In most of the studies however a large variation between subjects is found ranging from close to normal to very abnormal, which cannot be explained on base of audiometric data (Durlach, Thompson and Colburn, 1981; Colburn 1982; Koehnke and Besing, 1997; Kramer et al. 1996).

Three hypotheses for decreased performance of hearing-impaired subjects on binaural and spatial hearing tasks can be formulated. Firstly, reduced *audibility* possibly results in a loss of information available to the auditory system to perform the task. It is clear that attempts to correlate binaural performance to audiometric data are based on this hypothesis. Second reduced *supra threshold processing* in the temporal or spectral domain possibly results in less precise coding of relevant monaural features available to both ears, providing the binaural system with inadequate information for the task. The third hypothesis is that the *binaural signal processing itself* is less accurate. This framework of hypotheses results in *three types of impairment of hearing*. Careful investigation of binaural and spatial hearing of hearing impaired should take into account these aspects and try to differentiate between them.

For example, in another domain of auditory perception, this kind of approach was followed by Noordhoek et al. (2000), who investigated speech intelligibility. They tried to differentiate between audibility and supra threshold processing, which was subdivided in temporal resolution and frequency discrimination, spectral resolution, and peripheral compression as a cause for reduced speech intelligibility. For 23 out of 34 hearing-impaired subjects the data in the intelligibility tests could not be explained on basis of audibility. In a second experiment they found that two factors, reduced spectral resolution and reduced temporal resolution/frequency discrimination, determine suprathreshold impairment for speech intelligibility. There was no need for cognitive factors. It should be noted that in the domain of binaural and spatial hearing central and top-down processing aspects do play a significant role (Blauert, 1997). Involving these aspects could lead to at least a fourth hypothesis. In our current approach in investigating of binaural and spatial hearing of the hearing impaired these aspects are left aside, however.

The present study is a first approach to investigate the effect of mild symmetrical sensory neural hearing loss on the *precedence effect*, regarding the above mentioned hypotheses, which to our knowledge has not been investigated so far. The precedence effect, in short, the dominance of the first arriving acoustical information in signal perception, plays an important role in spatial hearing. For a general survey on the precedence effect the reader is referred to the review of Litovsky et al. (1999). They distinguish three phenomena, e.g., fusion, localization dominance and lag discrimination suppression and give an overview of free field and headphone studies. The present study is related to localization dominance: the every day experience that the directional impression of a sound source is dominated by the first sound impression. This phenomenon of the precedence effect is usually operationalized in the laboratory by simulating a source and its reflection and investigating the localization of the fused image. In the present study the design described by Goverts et al. (2000) is used to measure occurrence and quantification of the precedence effect. Using noise stimuli presented by headphone this design gives the possibility to quantify localization dominance, avoiding the subjective nature that is usually inherent to measurements of this effect of precedence (Litovsky, 1999). For six subjects with mild symmetrical sensorineural hearing impairment (Fletcher Index between 30 and 50 dB), we measured the occurrence of the precedence effect for various absolute signal levels and signal-to-noise ratios. In two conditions the precedence effect is also quantified. Overall results will be compared to normal hearing data for various sensation levels, which we collected in an earlier study (Goverts et al., 2000). Comparing results of normal hearing and hearing-impaired subjects at the same sensation level leads to conclusions about the contribution of the audibility hypothesis in interpreting group and individual data.

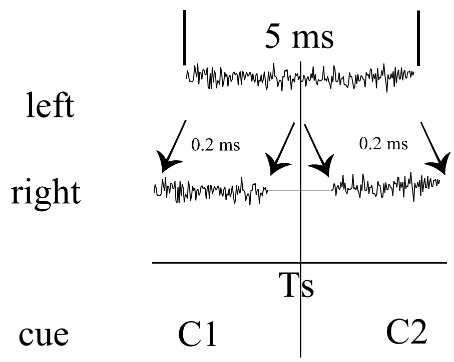


Figure 3.2.1 Schematic representation of a single signal presentation, consisting of a dichotic presentation of a low pass filtered white noise burst. The burst is subdivided in two parts, containing opposite temporal cues for lateralization, C1 and C2. The cue was always an interaural time difference of 0.2 ms. The crucial parameter is the relative switch time Ts, i.e. the time after signal onset at which cue C1 turns into cue C2, expressed relative to the total duration T. In this example Ts = 0.5, C1 = 'right leading', C2 = 'left leading'.

II MATERIALS AND METHOD

Subjects

In this study six subjects participated all of whom had a bilateral symmetric mild sensory neural hearing loss. Fletcher indices varied between 30 and 50 dB. No conductive components were involved. Individual audiograms are included in Figures. 3.2.8a and 3.2.8b. The range of age was 43-68 (mean age 55). None of the subjects was familiar with psycho-acoustic experiments.

Stimuli

In this study we used the test design described by Goverts et al. (2000). In this design, the signal is based on lowpass filtered white noise bursts, with a cutoff frequency of 2400 Hz and a duration T of 5 ms. For each trial a different noise burst is drawn randomly from the digitally stored, lowpass filtered noise. The masking noise is also lowpass filtered white noise with the same cutoff frequency, which is uncorrelated at the two ears. A single *signal* *presentation* consists of a dichotic presentation of this noise burst (see Figure 3.2.1.). The 5-ms signal presentation is subdivided into two parts, one part before Ts (leading part) and one after Ts (lagging part). In the leading part the noise burst is delayed 0.2 ms in one ear relative to the other. In the lagging part this delay is introduced in the other ear.[1] Thus the signal contains two opposite temporal cues for lateralization, C1 and C2. The crucial parameter is the switch time Ts, i.e. the time after signal onset at which cue C1 turns into cue C2. This switch time will be expressed relative to the total duration T (i.e. Ts between 0 and 1). The brief 5-ms signal presentation is perceived *as one fused sound image* with a specific lateral position. This position depends on Ts. For Ts values of 0 or 1, i.e. at the beginning or at the end of the signal, there is no switching of cues during the signal presentation, resulting in a single unambiguous cue for lateralization. By varying the value of Ts between 0 and 1 the ratio between the duration of cue C1and the duration of cue C2 changes. The working hypothesis is that when the perceptual weight of the parts before and after Ts is equal, the opposite lateralization cues C1 and C2 will cancel each other. For that particular value of Ts, which is named Tequal (Teq), the fused image is not lateralized, i.e. it is perceived in the middle. If there was no precedence effect, or other weighting, we would expect Teq to be 0.5. Thus, any systematic effect of lateralization for Ts = 0.5 must be due to some perceptual dominance.

Each *trial* contains two signal presentations with opposite values forC1and C2 as indicated in Figure 3.2.2 for two signals with Ts = 0.2. The subject's task is to indicate in a forced choice procedure the lateral position of the second signal relative to that of the first signal, i.e. 'left'

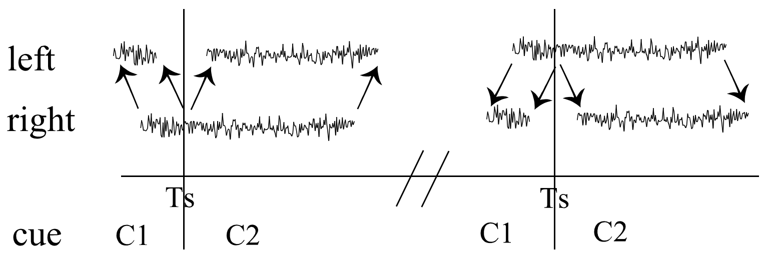


Figure 3.2.2 Schematic representation of a single trial, consisting of two signal presentations with the same value for Ts (in this example about 0.2) and opposite values for C1 and C2. In the first signal presentationC1= 'left leading' and C2 = 'right leading', in the second signal presentation the reverse is true.

[1]By introducing the interaural time differences , for one of the ears a small 0.4 ms gap is introduced, which is below gap detection threshold (e.g. Snell et al. 1994). This gap is enlarged on the schematic representation in Figure 3.2.1.

or ‘right’. No feedback is given to the subject. This wouldn’t make sense in this paradigm, since there is no ‘correct’ response. In conditions with masking noise, this masker is presented in each trial from 1 s before the first signal presentation until 1 s after the second signal presentation.

Procedure

Two types of experiments were performed. With the first type we determined the *occurrence* of the precedence effect at various sensation levels. For signals with Ts = 0.5 we measured the proportion of responses in accordance with C1, thus, the cue in the first half of the signal. A 50 % response would indicate either that the target was not detected or that there was no systematic preference for lateralization in accordance with C1 or C2. Any response percentage significantly larger than 50 % would indicate an effect of onset dominance. With the second type of experiment we aimed at *quantifying* the precedence effect for a given sensation level by determining that value of Ts at which the responses are at the 50 % chance level (thus, the value of Teq as discussed above).

An experimental series for investigating the occurrence of onset dominance consisted of 220 trials which were balanced over 11 values of signal or masker-noise level (equally spaced in dB), while keeping Ts constant at 0.5. An experimental series for quantifying onset dominance consisted of 220 trials which were balanced over 11 values of Ts in a range of [0, 0.1,...,1], while keeping the level of signal and masker-noise constant. By third-order polynomial curve fitting and interpolation, Teq was determined as the value of Ts for which the score was 50 %, indicating that the two stimuli in a trial had no systematic difference in terms of lateral position. The experimental paradigm ensured that the measurement of Teq was not biased by motivation, experimental skills, subjective aspects of judgement, or learning effects during a series. A series of 220 trials consisted of ten repeated blocks of 22 trials. Every block contained two essentially identical trials for each of the 11 parametric values; in these two trials the order of the two signals was simply reversed. Each *curve* that will be presented in the results section is based on two series, thus every point of the individual curves reflects 2*2*10 = 40 trials. All observed test-retest variability can be explained by statistical variation as a result of the limited number of trials per series.

Presentation levels

Due to hardware constraints the maximal signal level (Ls) that was presented was 83 dBA. For investigating the occurrence of precedence effect in quiet the signal level was varied over 11 values equally divided (in dB) between 83 dBA and the subject's individual threshold. For measuring occurrence of precedence effect in noise, for each subject signals were presented at a fixed level corresponding with about 40 dB SL in quiet, while varying the noise level (Ln) resulting in signal-to-noise ratios varying over 11 values between -10 to 40 dB (equally spaced in dB). This corresponds to sensation levels of about -4 and 46 dB SL respectively.[2] For the quantification measurements in quiet signals were presented for each subject at a fixed level of about 40 dB SL. The quantification measurement in noise was done at the same signal level with a signal-to-noise ratio of 0 dB corresponding to a *mean* sensation level of 6 dB, varying between 4 and 8 dB SL for the 6 subjects.[3]

Hardware

Stimuli were digitally generated on an OROS-AU22 sound processing card, which was placed in an 286 computer. They were presented to the listener by a Beyer DT48 headphone. Levels were adjusted by an in-house-produced programmable attenuator. Experiments were performed in a sound isolated room.

III OVERALL RESULTS

Thresholds

The absolute threshold for the 5 ms stimuli measured using a simple Bekesy paradigm varied between 35 and 56 dBA (calibrated using an artificial ear and a flat plate coupler). Masked

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[2]The same signal-to-noise ratios were chosen as used for normal hearing in our earlier study. For normal hearing these signal-to-noise ratios correspond to sensation levels of 0 to 50 dB.

[3]For subject 1 the presentation level for the measurement of occurrence in noise and both quantification measurements was 30 dB SL In order to achieve appropriate presentation levels for the measurement of occurrence in noise and both quantification measurements for subjects 5 and 6, an additional 10 dB amplification was applied leading to a signal level (Ls) of 93 dBA.

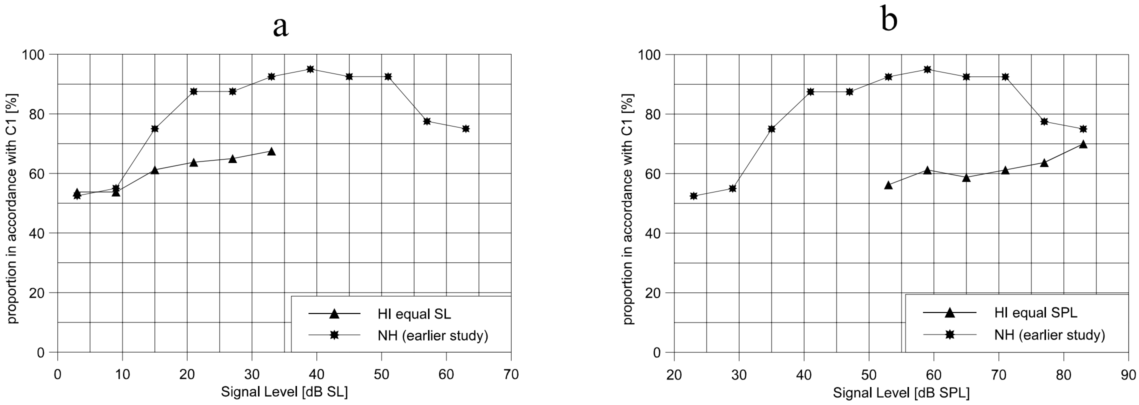


Figure 3.2.3 Proportion of responses in accordance with C1 versus signal level for stimuli in quiet with Ts = 0.5. As a reference median data for 5 normal hearing are given. Median data for the six hearing-impaired subjects for six corresponding Sensation Levels (+/- 2 dB) are given in Figure 3.2.3a and for six corresponding Sound Pressure Levels (+/- 2 dB) in Figure 3.2.3b.

thresholds for the signal in 75-dBA noise varied between 67 and 71 dBA. For normal hearing as measured in the earlier study these values are 20 and 65 dBA respectively.

Occurrence of the precedence effect as a function of signal level

The influence of signal level on the occurrence of the precedence effect in quiet was investigated. In this experiment Ts was held constant at 0.5 and the signal level was varied over a grid of 11 values equally distributed (in dB) between each subject's individual threshold and the maximal value of 83 dBA, determined by hardware constraints. Figures 3.2.3a and 3.2.3b present the proportion of responses in accordance with C1 versus Ls, expressed in absolute and sensation level, respectively. Median results for the six subjects are presented for six overlapping dB SL values and for six overlapping dB SPL levels (+/- 2 dB). As a reference, median data for five normal-hearing subjects are replotted from our earlier study.

Occurrence of precedence effect as a function of signal-to-noise ratio

The influence of signal-to-noise-ratio on the occurrence of the precedence effect was investigated using the same experimental design. In this experiment Ts was held constant at 0.5 and the signal level Ls was held at a sound pressure level corresponding to about 40 dB SL in quiet. Consequently, the absolute sound pressure level varied over subjects. The level of the masking noise, Ln, was varied over a grid of 11 equally distributed values (in dB) resulting in signal-to-noise ratios between -10 and 40 dB, corresponding to a variation of the sensation level of the signal over a range from about -4 to about 46 dB.[4] Median results are given in Figure 3.2.4, which presents the proportion of responses in accordance with C1 versus signal-to-noise ratio. As a reference, median data for five normal hearing listeners are replotted from our earlier study for the same signal to noise ratios.

Quantification of precedence effect at 40 dB SL

To quantify the precedence effect at 40 dB SL in quiet the proportion of responses in accordance with the first cue, C1, was determined as a function of the switch time Ts. Constant sensation level implies that, the absolute sound pressure level varied over subjects. Median results are given in Figure 3.2.5, which presents the proportion of responses in accordance with C1 versus Ts. As a reference, median data for five normal hearing listeners are replotted from our earlier study for about the same sensation level.

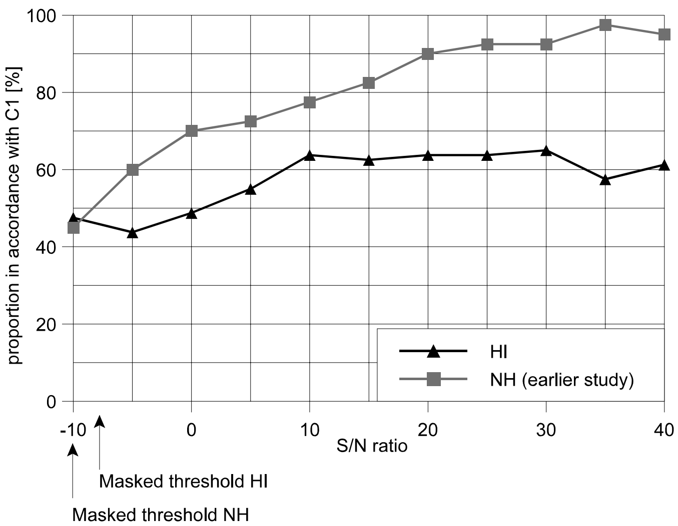


Figure 3.2.4 Proportion of responses in accordance with C1 versus noise level for signals with Ts = 0.5 and at a fixed signal level (corresponding to about 40 SL in quiet for each subject) while the signal-to-noise ratio was varied from -10 to 40 dB. Median data for the six hearing-impaired subjects are presented. As a reference median data for five normal hearing are given for the same range of signal-to-noise ratios. Average masked thresholds in 75 dBA noise are indicated for the hearing-impaired subjects and for the normal-hearing subjects.

[4]The sensation level of the signal in noise was not measured for al subsequent values of Ln but only at a level of the masking noise of 75 dBA. This means that for all other values of Ln the sensation level is not exactly known.

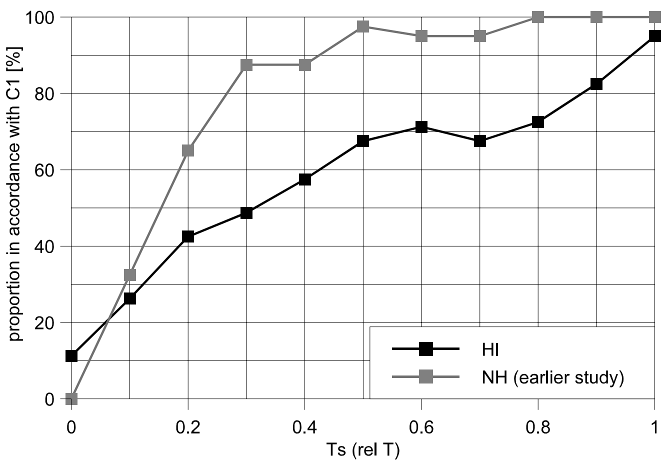


Figure 3.2.5 Proportion of responses in accordance with C1 versus the switch time Ts in quiet for a fixed signal level of about 40 dB SL for each subject. Median data for the six hearing-impaired subjects are presented. Teq is about 0.3 in this graph. As a reference median data for five normal hearing are given at a signal level of about 45 SL. Teq is about 0.15 in this graph.

Quantification of precedence effect in noise at a sensation level of about 6 dB

To quantify the precedence effect at a lower sensation level the dominance of C1 as a function of Ts was determined at a signal-to-noise ratio of 0 dB.[5] For each subject the signal levels corresponded to about 40 dB SL in quiet. The mean sensation level of the signal in noise was 6 dB, varying between 4 and 8 dB SL for the 6 subjects. The absolute sound pressure levels of signal and noise varied over subjects. Results are given in Figure 3.2.6, which presents the score in accordance with C1 versus Ts.

[5] The same signal-to-noise ratio was chosen as used in our earlier study. For normal hearing this signal-to-noise ratio corresponds to a sensation level of 10 dB.

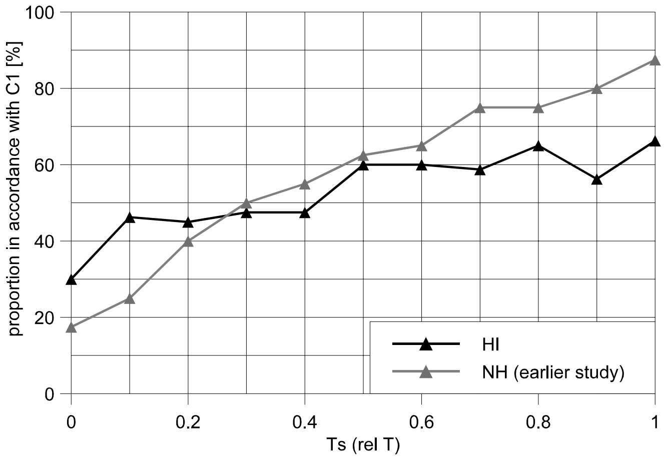


Figure 3.2.6 Proportion of responses in accordance with C1 versus the switch time Ts in noise with a fixed signal-to-noise ratio of 0 dB. Median data for the six hearing-impaired subjects are presented. As a reference median data for five normal hearing are given at a signal level of about 45 SL. Teq is about 0.35 in both graphs.

IV DISCUSSION OF OVERALL DATA

Occurrence of precedence effect

A reduced occurrence of precedence effect was found for the group of sensory neural impaired subjects as compared to normal hearing data (Figures 3.2.3a, 3.2.3b and 3.2.4). For Ts = 0.5 where the cue intervals for lateralization are equal and would have equal impact, provided there were no perceptual weighting, their maximal scores in accordance with the first cue C1 are about 70 and 65% without and with masking noise respectively.
Whereas the hearing-impaired subjects never show scores higher than 70%, normal-hearing subjects perceive the laterality of almost 100% of the signals in accordance with C1 in these conditions. Although audibility seems to be a factor, in that the percentage agreement increases with level, it is certainly not the only reason for the difference in performance, since the maximal score is well below values reached at comparable sensation levels for normal hearing listeners. Although the normal hearing group was younger than the hearing-impaired subjects (range 27-36 years versus range 43-68), an age difference of this order is not likely to be a factor.

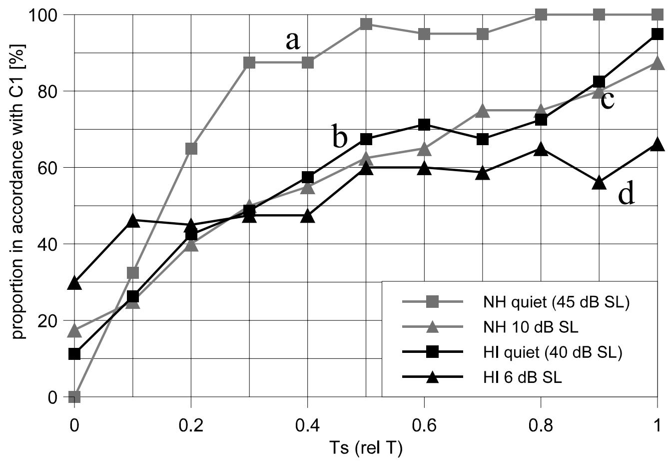


Figure 3.2.7 Proportion of responses in accordance with C1 versus Ts. Curve *a*) is a replot of the normal hearing score in accordance with C1 as a function of Ts at 45 dB SL, *b*) the normal hearing data measured at a signal-to-noise ratio of 0 dB with the same signal level as for curve a) resulting in a sensation level of 10 dB, *c*) the hearing impaired data measured at 40 dB SL and *d*) the hearing impaired data measured at a signal-to-noise ratio of 0 dB measured with the same signal level as for curve c), resulting in a sensation level of about 6 dB.

Quantification of precedence effect

In our earlier study we distinguished two possible explanations for a decrease of the score in accordance with C1 at Ts = 0.5 for normal hearing towards low sensation levels, i.e. decreased lateralization accuracy and decreased precedence effect. This distinction will also be made for the decreased performance of hearing impaired compared to normal hearing data. We inspect the relation between lateralization score and Ts for both normal hearing (earlier study) and hearing impaired. Figure 3.2.7 shows a replot of *a*) the normal hearing score (from Figure 3.2.5) in accordance with C1 as a function of Ts at 45 dB SL as a point of departure, *b*) the normal hearing data measured at a signal-to-noise ratio of 0 dB with the same signal level as for curve a) resulting in a sensation level of 10 dB (from Figure 3.2.6), *c*) the hearing impaired data measured at 40 dB SL in quiet and *d*) the hearing impaired data measured at a signal-to-noise ratio of 0 dB measured with the same signal level as for curve c), resulting in a sensation level of about 6 dB. For each curve next to the overall shape two values will be looked at. At first Tequal (Teq), the value of Ts at which the cues C1 and C2 cancel each other; this value gives an indication of the strength of precedence effect. The other value is the *lateralization error*, the proportion of wrong lateralization at Ts = 0 and Ts = 1. It is defined as the mean of the scores at Ts = 0 and (100%-score) at Ts = 1. This value gives an indication of the lateralization performance, because in these conditions there is only a single unambiguous cue for lateralization. Note that the conditions Ts = 0 and Ts = 1 are technically equivalent; any difference between the score at Ts = 0 and (100 - score) at Ts = 1 must be due to statistical variations as a result of the limited number of trials. The lateralization error is the converse of the *lateralization accuracy*.

Inspecting Figure 3.2.7, the first observation is a great similarity between curve b) and c): the median performance of the hearing impaired in quiet can be compared to the normal hearing performance in noise with a signal-to-noise ratio of 0 dB. Considering the scores at Ts is 0 or 1, there is a clear difference between curve a) versus curves b) and c), which must be attributed to a decrease of lateralization accuracy. However, it must be stressed that the decrease in lateralization accuracy does not explain the difference in Teq between curve a) versus curves b) and c). If accuracy were the only issue, the 50% score would be at the same value of Ts for all three curves. The shift of Teq from about 0.15 to 0.30 implies that the dominance of the initial part of the signal is diminished, i.e., that the precedence effect at 10 dB SL for normal hearing and at 40 dB SL for hearing impaired is smaller than at 45 dB SL for normal-hearing subjects. The effect of noise on the overall result for these hearing-impaired subjects appears as a decrease of lateralization accuracy (curve d). All scores at values of Ts for which lateralization information is available are affected, but the overall value of Teq remains unaffected. It should be noted that, at this signal-to-noise ratio with corresponding sensation levels of 4 to 8 dB, detectability of the signal could be a factor.

Table 3.2.1 summarizes the data of the six hearing-impaired subjects compared to the normal hearing results of our earlier study. In quiet the *lateralization error* is substantially larger for the hearing impaired as well as the *spread* in the individual data. For both normal hearing and hearing impaired, the lateralization performance is influenced by adding of masking noise. For normal-hearing subjects, lateralization error in noise is in the same range as the hearing impaired lateralization error without noise. This holds also for the spread in the individual data. In noise both groups have equal standard deviation of lateralization. The value of Teq (obtained by a third order polynomial fit) shows a reduced *precedence effect* for the hearing-impaired subjects, the first 1.60 ms (Teq * total duration = 0.32 * 5) of a signal have the same perceptual weight as the remaining 3.40 ms (0.68 * 5). For normal hearing the values are 0.85 ms (0.17 * 5) and 4.15 ms (0.83 * 5). Hearing impaired data again show a substantially larger standard deviation over listeners. Normal hearing performance is more influenced by the noise; the average strength of precedence effect in noise as well as the spread in the data is of equal magnitude for both groups.

Table 3.2.1 Values of Teq and lateralization error for the six subjects. Average data for normal hearing are provided as a reference (earlier study). For some subjects reliable values for Teq cannot be established. (Subject 2 in quiet and in noise, Subject 5 in noise).

	Teq (rel T)		**lateralization error [%]**	
	quiet	noise	quiet	noise
subject				
S1	0.31	0.39	9	21
S2	no reliable value	no reliable value	39	53
S3	0.19	0.16	3	31
S4	0.40	0.45	8	30
S5	0.22	no reliable value	15	48
S6	0.49	0.47	11	30
avg	0.32	0.37	14.2	35.5
std	0.13	0.12	11.7	11.2
normal hearing				
avg	0.17	0.35	0.8	19.6
std	0.02	0.1	1.2	10.4

A reduced precedence effect for hearing-impaired subjects is in line with findings of Cranford et al. (1993) who investigated the influence of hearing loss on localization dominance using an essentially different design. They examined performance on a free-field sound localization task for four groups of young and elderly subjects, matched with respect to age and the presence or absence of a sloping sensory neural hearing loss. They found reduced performance on the precedence effect task for both increased age and hearing loss and tentatively conclude that ‘hearing loss may have a relatively greater effect on the performance of at least some elderly subjects than it does on younger subjects.’

Hypotheses for degradation of performance

In the introduction three hypotheses were mentioned to account for decreased performance on binaural tasks. It is obvious that audibility, which comprises the first hypothesis, is always a factor for auditory tasks; however Figures 3.2.3 and 3.2.4 indicate that we cannot explain the overall hearing impaired performance by this hypothesis. At equal sensation levels there remains a substantial difference between listeners with normal and impaired hearing in occurrence of precedence effect both in quiet and in noise. The decrease of lateralization accuracy and the decrease in precedence effect as observed in Figure 3.2.5 must be due to supra threshold processing impairment or directly by binaural processing impairment. It is remarkable that the degree of both decrease in lateralization accuracy and decrease of precedence effect for the hearing impaired are similar to the effect of adding noise for normal hearing-subjects (Figure 3.2.7). A possible explanation could be that sensory neural hearing loss, like additional noise for normal hearing, causes *noisy temporal coding* in both cochleas, providing the binaural system with fluctuating temporal information, resulting in a decreased lateralization accuracy. This can be brought in line with the causes we suggested for the decrease of precedence effect and lateralization accuracy for normal hearing (Goverts et al., 2000). The decrease of lateralization for normal hearing at 10 dB SL was interpreted as reflecting the statistical fluctuations in the Interaural Time Delay cue caused by the relative magnitude of the internal and external noise at low sensation levels, in line with Houtgast and Plomp (1968). A plausible hypothesis is that hearing-impaired subjects have a form of increased internal noise, which even at a sensation level of 40 dB introduces statistical fluctuations of the same amount as normal hearing at 10 dB SL.

V INDIVIDUAL DATA

In this section individual results for the six subjects are presented. Figures 3.2.8a and 3.2.8b, for each subject, age, level of uncomfortable loudness (UCL) and the maximal speech discrimination for monosyllabic words. In addition, the audiometric configuration for both ears is plotted and the curves that represent the quantification of the precedence effect at a sensation level of 40 dB in quiet and with the same signal level in noise resulting in a signal-to-noise ratio of 0 dB. As a reference median data for five normal-hearing subjects (earlier study) are plotted. The data in Table 3.2.1 were derived from these individual plots. Individual results will be discussed below in comparison with normal hearing results in terms of decrease of lateralization accuracy (derived from the scores at Ts = 0 and Ts = 1) and of precedence effect (in terms of Teq).

Subject 3 has a flat and very mild hearing loss. Her performance in quiet is rather similar to that of the normal hearing; this holds for both the data for lateralization and Teq in quiet. Adding noise results in *a pure decrease in lateralization accuracy* for this subject. Although the value of Teq remains approximately constant, all the scores tend to converge towards 50%.

Subjects 2 and 4 both have a more or less steep high frequency hearing loss. Subject 2 shows a strongly reduced *lateralization accuracy* in quiet, compared to normal hearing. The value of the lateralization error is about 40%, which makes it impossible to find a reliable value for Teq by curve-fitting. Adding noise results in *minimal lateralization accuracy* for this subject: for all values of Ts the score is about 50%. It is again not possible to find a reliable value for Teq in this condition. Subject 4 shows a *strong decrease of precedence effect* in quiet. She has roughly normal values for lateralization but a minimal precedence effect with Teq approximately 0.4. Adding noise results in a *combined effect* of a rather large decrease of lateralization accuracy (lateralization error increases from 8 to 30%) and a small further decrease of precedence effect (Teq from 0.40 to 0.45).Subjects 1, 5 and 6 all have more or less the same audiometric configuration: a symmetric flat hearing loss of about 50 dB. Subject 1 shows, like subject 4, a *strong decrease of precedence effect* in quiet. Adding noise gives a *combined effect* of a little decrease of lateralization accuracy (lateralization error from 9 to 21%) and a small further decrease of precedence effect (Teq from 0.31 to 0.39) resulting in scores very similar to normal hearing data.

Subject 5 shows in quiet a *combined effect* of lateralization inaccuracy and decreased precedence effect. Adding noise results, like Subject 2, in *minimal lateralization accuracy* for this subject: for all values of Ts the score is about 50%. It is hence not possible to find a value for Teq in this condition. Subject 6 shows a *total absence of precedence effect* (Teq = 0.5) and rather normal lateralization accuracy in quiet. Adding noise gives a decrease in lateralization accuracy (lateralization error from 11 to 30%).

The individual results are summarized in a polar representation in Figure 3.2.9. This polar representation was chosen because it reflects the fact that Teq cannot be determined reliably for low values of lateralization accuracy.

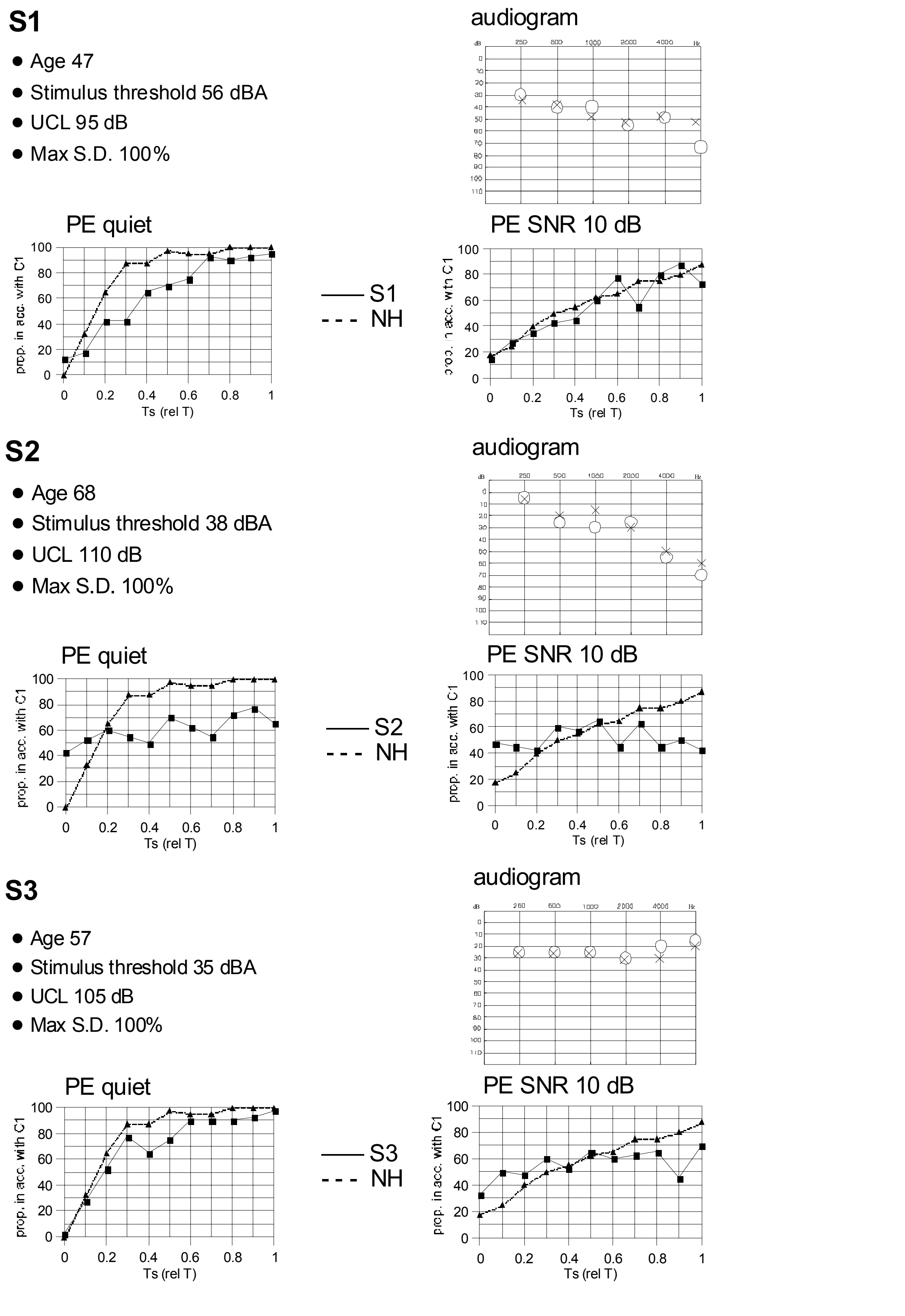
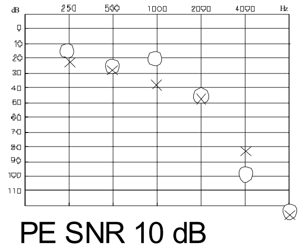


Figure 3.2.8a Individual data for the hearing-impaired subjects, S1, S2 and S3. For each subject the proportion of responses in accordance with C1 versus the switch time Ts is given with a fixed signal level of about 40 dB SL in quiet and at the same signal level at a signal-to-noise ratio of 0 dB. Median normal hearing results are given as a reference. Besides age, stimulus threshold, UCL level and maximum speech discrimination scores are given.

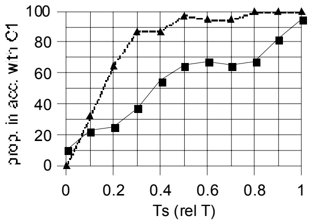
S4

- Age 60
- Stimulus threshold 47 dBA
- UCL 90
- Max S.D. 73%

audiogram

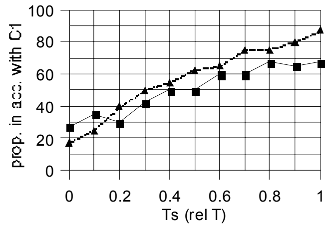


PE quiet



—— S4
- - - NH

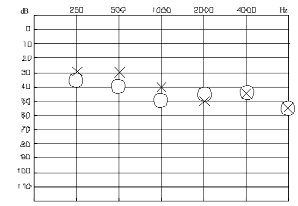
PE SNR 10 dB



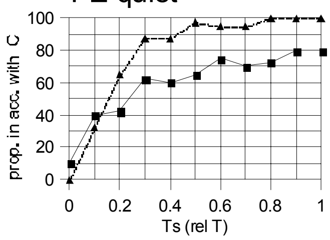
S5

- Age 43
- Stimulus threshold 53 dBA
- UCL 90 dB
- Max S.D. 100%

audiogram

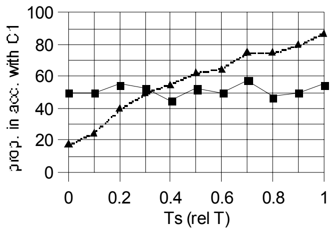


PE quiet



—— S5
- - - NH

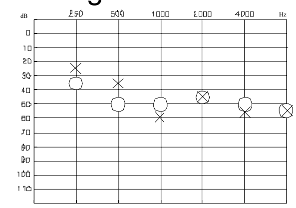
PE SNR 10 dB



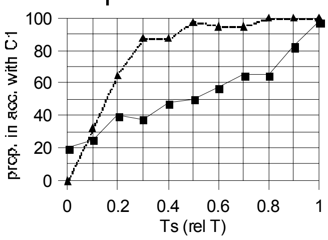
S6

- Age 54
- Stimulus threshold 56 dBA
- UCL 90 dB
- Max S.D. 100%

audiogram



PE quiet



—— S6
- - - NH

PE SNR 10 dB

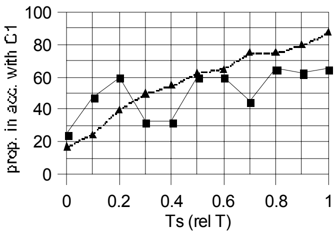


Figure 3.2.8b As Figure 3.2.8a, representing data for hearing impaired subjects S4, S5, and S6.

For each subject the shift of lateralization accuracy and Teq is plotted for the two conditions, i.e. 40 dB SL in quiet and the same signal level at a signal-to-noise ratio of 0 dB. Normal hearing data are plotted as a reference. For the noise condition a 'normal hearing area' is bounded by +/- 1 standard deviation. The influence of masking noise for the normal hearing is characterized by an increase of Teq and a decrease of lateralization accuracy, and also by an increase of the standard-deviation area. The data for subject S2 are not included in this Figure, because Teq could not be established for this subject, neither in quiet nor in noise. Three subjects (S1, S4 and S5) have a shift orientation more or less similar to normal hearing. Subject S3 turns out to have a normal Teq value, which is not influenced by noise. There is no relationship between the audiometrical configuration and the results on both lateralization accuracy and precedence effect, most clearly illustrated by the different patterns of subjects S1, S5, S6, who have nearly identical audiograms. Also the other factors (age, maximum speech discrimination, UCL) have no apparent influence.

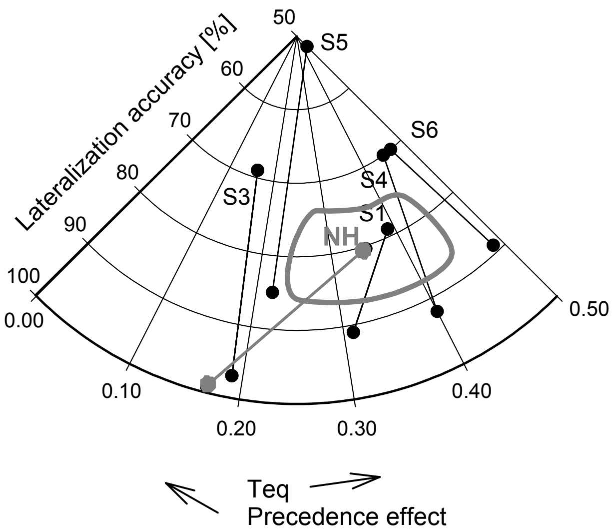


Figure 3.2.9 Scores for lateralization error and Teq are represented in a polar plot in two conditions i.e. a fixed signal level of about 40 dB SL and at the same signal level at a signal-to-noise ratio of 0 dB. Besides data for each hearing-impaired subject, average normal hearing results are given as a reference. For both conditions (i.e., quiet and noise) a normal hearing data are given as well as a 'normal hearing area' is given, bounded by +/- 1 standard deviation for the noise condition.

VI GENERAL DISCUSSION AND CONCLUSION

For the six hearing-impaired subjects with a mild sensory neural hearing loss, overall results show a reduced precedence effect for lateralization, as well as a reduced lateralization accuracy. Mean performance of the hearing-impaired subjects in quiet is similar to normal hearing performance in masking noise at a signal-to-noise ratio of 0 dB. There is a broad range of individual result patterns in terms of strength of precedence effect and lateralization accuracy. These results cannot be explained on basis of reduced audibility alone. Measurements of cochlear function (frequency resolution, temporal resolution) should be performed to distinguish between possible effects of sub-optimal cochlear processing (e.g. noisy temporal coding) or binaural processing impairment, as a cause for the reduced binaural performance.

3.3 The precedence effect at low sensation levels: non-classic and classic design compared*

The precedence effect at low sensation levels is investigated using a classic free field design, in which two loudspeakers positioned at plus and minus 45 degrees, are driven by identical stimuli while one is variably delayed. Six normal-hearing subjects participated in the experiment. Performance at a sensation level of 45 dB is compared to that at 10 dB, established either by lowering the signal level or by adding masking noise. A decreased precedence effect at low sensation levels is found. The results turn out to be in line with our earlier study (Goverts et al., 2000), in which a non-classic headphone-based design was used. Thus, this headphone based paradigm has turned out to be useful in investigating aspects of localization dominance.

Results of both studies stress the importance of audibility and sensation level in the precedence effect. This is important for correctly interpreting aspects of binaural performance in hearing-impaired listeners.

** This section is based on Goverts, S.T., Houtgast, T, Festen, J.M. (**2001**). "**The precedence effect at low sensation levels**," presented at ARO midwinter meeting.*

I INTRODUCTION

The precedence effect, in short, the dominance of the first arriving acoustical information in signal perception, plays an important role in spatial hearing (Blauert, 1997; Litovsky et al. 1999; Wallach et al., 1949) . Litovsky et al. distinguish three phenomena: fusion, localization dominance, and lag discrimination suppression. The present research is related mostly to localization dominance: the every day experience that the directional impression of a sound source is little influenced by its reflections. Localization dominance is strongest at delays in the range between 1 millisecond and echo threshold (typically about 5 ms, depending on the type of sound).

With the future intent to investigate the precedence effect for hearing-impaired subjects, we aimed at determining the influence of sensation level on the precedence effect for normal hearing listeners. Results reported earlier (Goverts et al., 2000) showed a decrease of precedence effect for sensation levels below 30–40 dB.

However, a *non-classic design* was used, firstly introduced by Houtgast (1994). Using noise stimuli presented by headphones this method provides a method to quantify the underlying mechanism of localization dominance (or rather lateralization dominance) aiming to minimize the effects of a subject's skill and motivation that are usually inherent to measurements of this effect of precedence. In an attempt to evaluate the use of the non-classic design, in the present study we re-investigated the influence of sensation level on localization dominance, using a classic paradigm (Blauert, 1997; Litovsky, 1999), in which two loudspeakers at plus and minus 45 degrees in the azimuthal plane are stimulated by the same click stimulus, while one is delayed.

In this experiment, like in the earlier study, the sensation level is varied either by changing the absolute signal level or by adding a masking noise with the signal level fixed. Results with the classic design are compared to results found earlier in three conditions: 45 dB SL, 10 dB SL and 10 dB SL in noise.

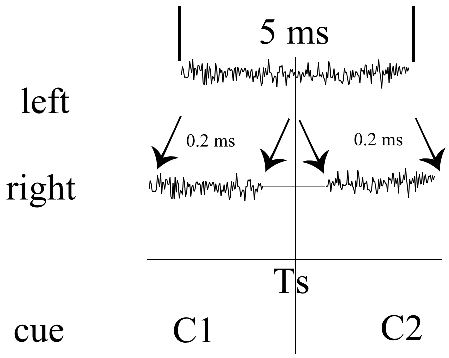


Figure 3.3.1 Schematic representation of a single signal presentation, consisting of a dichotic presentation of a low pass filtered white noise burst. The burst is subdivided in two parts, containing opposite temporal cues for lateralization, C1 and C2. The cue was always an interaural time difference of 0.2 ms. The crucial parameter is the relative switch time Ts, i.e. the time after signal onset at which cue C1 turns into cue C2, expressed relative to the total duration T. In this example Ts = 0.5, C1 = 'right leading', C2 = 'left leading'.

II NON-CLASSIC DESIGN

A brief review of methodological aspects and results of the earlier research (as reported in section 3.1.1) is given below.

Method

In this paradigm the signal is based on lowpass filtered white noise bursts, with a cutoff frequency of 2400 Hz and a duration T of 5 ms as indicated by Figure 3.3.1. The signal is subdivided in two parts, one part before Ts (leading part) and one after Ts (lagging part). In the leading part the noise burst is delayed 0.2 ms in one ear relative to the other. In the lagging part this delay is introduced in the other ear.[1] Thus the signal contains two opposite temporal cues for lateralization, C1 and C2. The crucial parameter is the switch time Ts, i.e. the time after signal onset at which cue C1 turns into cue C2. Ts is expressed relative to the

[1]By introducing the interaural time differences, for one of the ears a small 0.4 ms gap is introduced, which is below gap detection threshold (e.g. Snell et al. 1994). This gap is enlarged on the schematic representation in Figure 3.3.1.

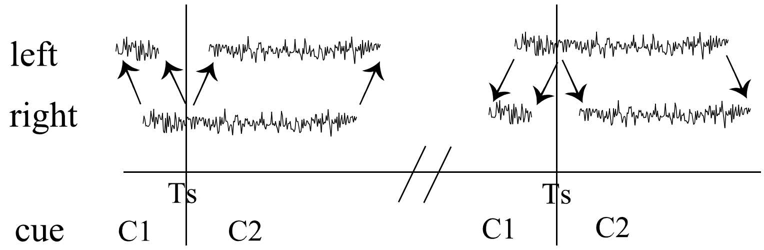


Figure 3.2.2 Schematic representation of a single trial, consisting of two signal presentations with the same value for Ts (in this example about 0.2) and opposite values for C1 and C2. In the first signal presentation C1 = 'left leading' and C2 = 'right leading', in the second signal presentation the reverse is true.

total duration T. Each *trial* contains two signal presentations with opposite values for C1 and C2, as indicated in Figure 3.3.2.

The subject's task is to indicate in a forced choice procedure the lateral position of the second signal relative to that of the first signal, i.e. 'left' or 'right'.
The perceived lateral position of the stimulus depends on Ts. This is illustrated using a plot of normal hearing lateralization scores in accordance with C1 as a function of Ts (see Figure 3.3.3). For Ts values of 0 or 1, i.e. at the beginning or at the end of the signal, there is no switching of cues during the signal presentation, resulting in a single unambiguous cue for lateralization. By varying the value of Ts between 0 and 1, the ratio between the duration of cue C1 and the duration of cue C2 changes. The *working hypothesis* of this design is that when the perceptual weights of the parts before and after Ts are equal, the opposite

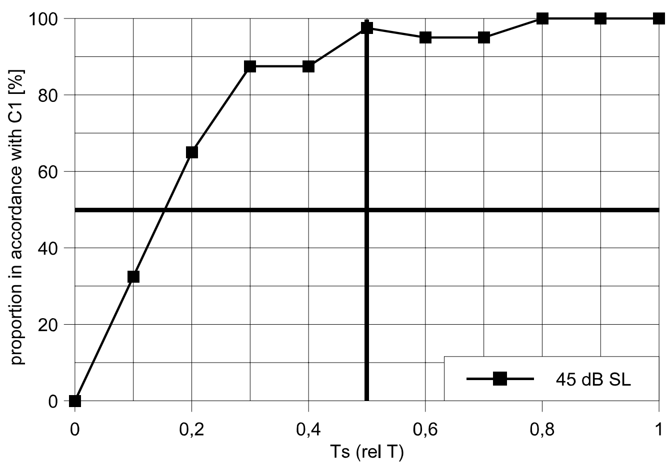


Figure 3.3.3 Illustrational curve giving the proportion of responses in accordance with C1 versus the switch time Ts at a signal level of 45 dB SL

lateralization cues C1 and C2 will cancel each other. For that particular value of Ts, which is named Tequal (Teq), the fused image is not lateralized, i.e. it is perceived in the middle, more precisely: there is no systematic difference between the perceived lateral positions of the two signals in a trial leading to a 50% score. The measurement of Teq is not biased by motivation, experimental skills, subjective aspects of judgement, or learning effects. If there was no precedence effect we would expect Teq to be 0.5. Thus, any systematic effect of lateralization for Ts = 0.5 must be due to some perceptual dominance.

Occurrence and Quantification

Two types of experiments were performed. With the first type the occurrence of the precedence effect was determined at various sensation levels. For stimuli with Ts = 0.5 we measured the proportion of responses in accordance with C1, thus, the cue in the first half of the signal. A 50 % response would indicate either that the target was not detected or that there was no systematic preference for lateralization in accordance with C1 or C2. Any response percentage significantly larger than 50 % would indicate an effect of onset dominance. With the second type of experiment it was aimed to quantify the precedence effect for a given sensation level by determining that value of Ts at which the responses are at the 50 % chance level (thus determining Teq).

Results

In Figure 3.3.4 overall results for five normal-hearing subjects are given on the occurrence of precedence affect as a function of sensation level. The effects on onset dominance of varying

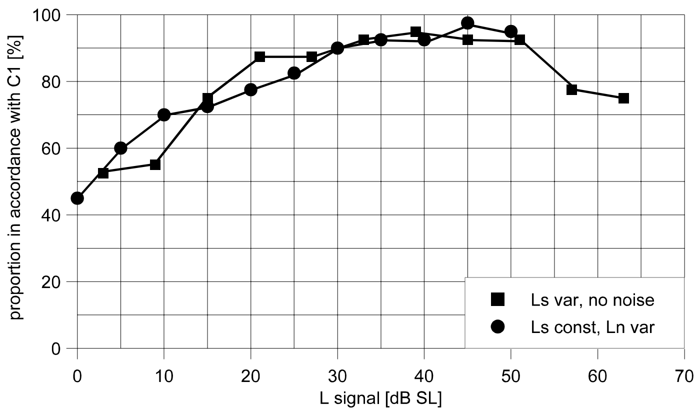


Figure 3.3.4 Proportion of responses in accordance with C1 versus sensation level, either by varying absolute signal level without masking noise or by varying noise level with a fixed signal level (65 dBA).

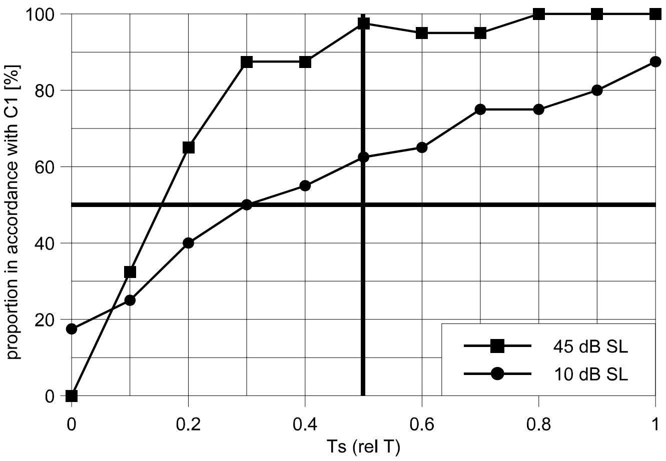


Figure 3.3.5 Proportion of responses in accordance with C1 versus the switch time Ts at a signal level of 45 dB SL and at a sensation level of 10 dB in noise.

sensation levels, either by decreasing the absolute signal levels or by raising masking-noise levels, are very similar. From 30 to 50 dB SL stimuli are mainly lateralized in accordance with the first cue, C1. Below 30 dB SL this dominance gradually diminishes to the chance performance (i.e. no dominance of either cue) as expected at 0 dB SL. The 45 dB SL and 10 dB SL curves (Figure 3.3.5) indicate that besides a decrease in lateralization accuracy there is indeed a decrease in precedence effect. When accuracy would be the only issue, a 50% score for 45 dB SL would remain 50% at 10 dB SL. The shift of Teq from about 0.15 to 0.30 implies that the dominance of the initial part of the signal is diminished, i.e., that the precedence effect at 10 dB SL is smaller than at 45 dB SL.

III CLASSIC DESIGN

Method

In this paradigm the stimulus configuration consists of 9 loudspeakers positioned in a sound insulated room as illustrated by Figure 3.3.6. This design is in accordance with the classic setup as described in literature (e.g. Blauert, 1997; Litovsky et al., 1999). Two loudspeakers (nr 2 and 8), positioned at plus and minus 45 degrees, are driven by identical stimuli while

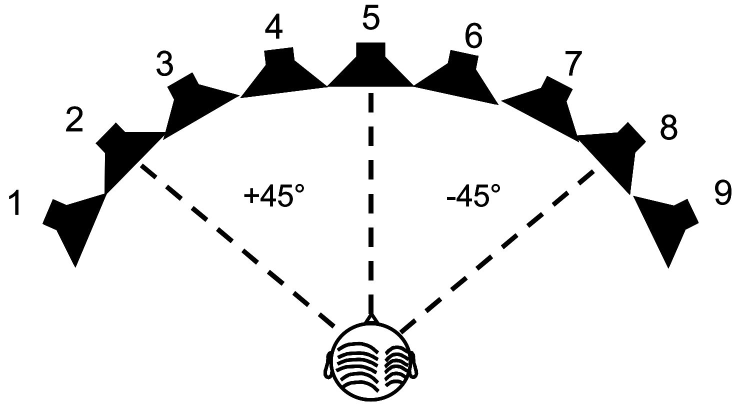


Figure 3.3.6. Schematic diagram of stimulus configuration in the classic design

one is variably delayed. For this configuration the relationship between the perceived location of the sound image as a function of delay is well known for clicks as illustrated by Figure 3.3.7 adopted from Litovsky et al.(1999). With no delay, one fused sound image is heard, ideally in the middle. If the delay shifts between 0 and 1 ms the image shifts toward the leading speaker summing localization). From 1 ms to echo threshold (when a second image appears) the image is heard at the lead location (localization dominance). The *working hypothesis* is that a decrease in precedence effect will appear as a delayed transition from summing localization to localization dominance.

Two stimuli were used: a Dutch sentence (duration about 2 s) and lowpass filtered white noise bursts, with a cutoff frequency of 2400 Hz and a duration T of 5 ms resembling the stimuli used in the non-classic design. Localization was measured as a function of delay in three conditions: at 45 dB SL, at 10 dB SL and at 10 dB SL in noise.

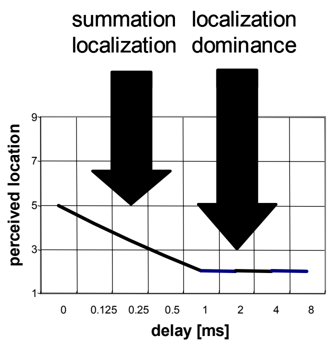


Figure 3.3.7 Theoretical curve. Perceived location versus stimulus delay (adopted from Litovsky et al., 1999)

Results

In Figure 3.3.8 overall response distributions of six normal-hearing subjects are given for the two types of stimuli and the three conditions. In Figure 3.3.9 mean responses are given. For noise stimuli the transition from summing localization to localization dominance is shifted from 1-2 ms to about 8 ms for both 10 dB SL conditions. This also holds for speech at 10 dB SL in noise. For speech stimuli at 10 dB SL the localization dominance stage is not reached. The distribution patterns indicate that there is no large decrease in localization accuracy.

IV DISCUSSION AND CONCLUSION

Both experiments reveal a dependence of the precedence effect on sensation level, varied either by lowering the signal level or adding masking noise.

Besides a decreased accuracy in lateralization (non-classic design) and localization (classic design) a decrease in the precedence effect towards low sensation levels was found.

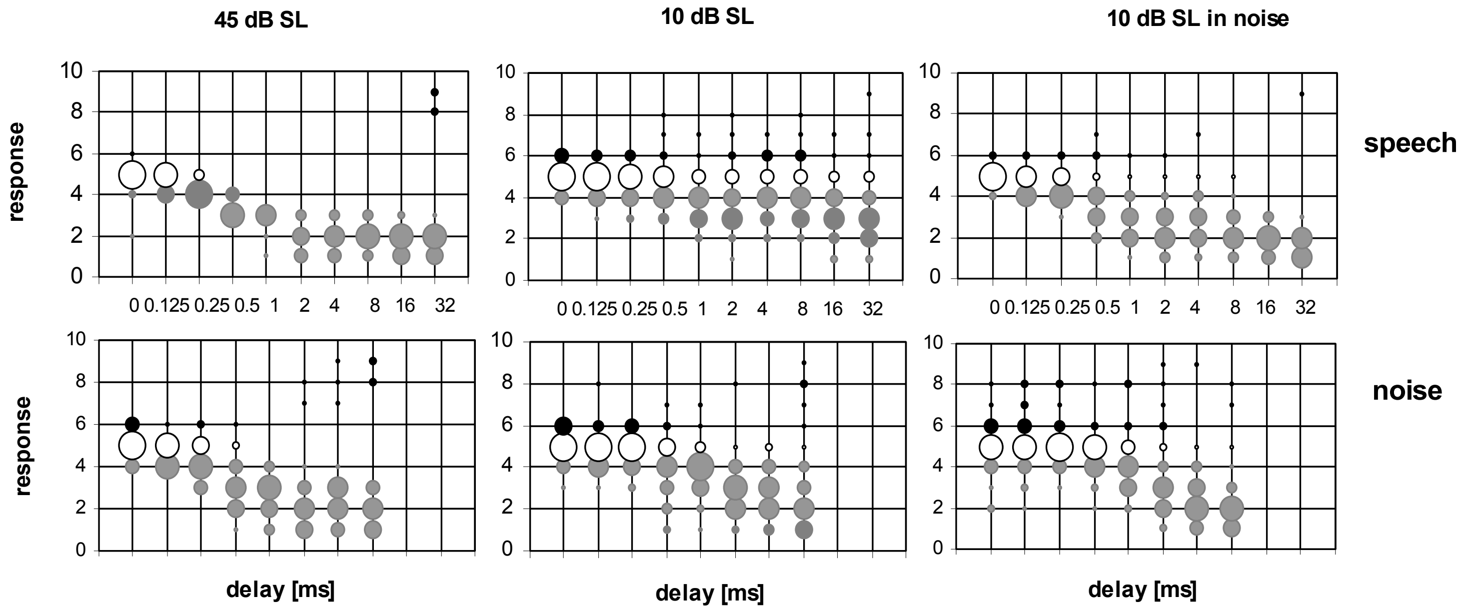


Figure 3.3.8. Distribution of responses for two stimuli in three conditions. For each delay the responses in the 9 directions are represented by the circle area. Left- and right leading stimuli are taken together. Response in the lead, middle and lag direction are represented grey, white and black respectively. Overall data for six normal hearing subjects are given.

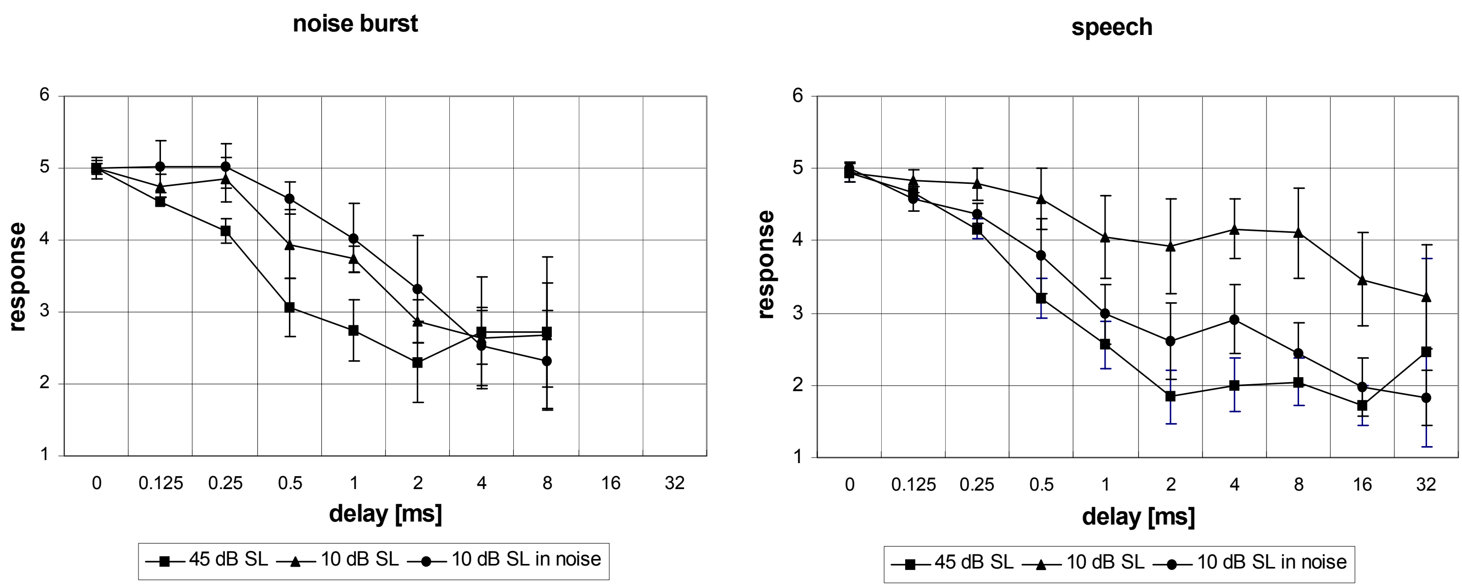
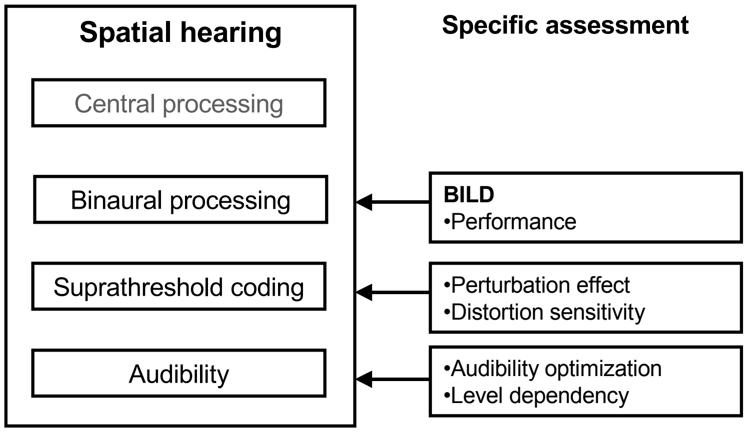


Figure 3.3.9 Mean responses for two stimuli in three conditions and standard deviation for six normal-hearing subjects. The error bars represent the standard-deviation.

The results stress the importance of audibility and sensation level in the precedence effect. This is important for correctly interpreting aspects of binaural performance in hearing-impaired listeners.

The non-classic headphone based paradigm has turned out to be useful in investigating aspects of localization dominance, now that the results of our earlier study are confirmed by the present study. Therefore, there is substantial reason to adopt this design as classic (Litovsky 2001, personal communication). The additional value of the design is that it ensures that the measurement is not biased by motivation, experimental skills, and subjective aspects of judgement.

4 The role of supra threshold coding in the Binaural Intelligibility Level Difference



This subplot of Figure 1.4 illustrates the main themes of this chapter in relation to the general approach of the thesis. The chapter deals with specific assessment of the BILD. The similarity in suprathreshold behavior of the BILD and BMLD is investigated by means of the perturbation effect approach. The role of audibility in the BILD is investigated by means of the level-dependency approach and audibility optimization. The role of suprathreshold coding is investigated by the distortion sensitivity approach..

4.1 The relation between binaural unmasking in speech detection and speech intelligibility*

We investigated the similarity in suprathreshold behaviour of binaural unmasking in speech intelligibility and speech detection (BILD and BMLD respectively) for six normal-hearing listeners. Speech reception thresholds (SRT) and speech detection thresholds (SDT) were determined in a N0S0 and a N0Sπ presentation mode by headphones. Both speech and noise signals were subjected to ear-independent random perturbations in several domains resulting in one undistorted condition and four additional distorted conditions. Correlations between BILD, BMLD and diotic measures were inspected. The results suggest a close relationship between unmasking in speech detection and speech intelligibility. While their absolute difference can be explained in terms of audibility, as demonstrated years ago, BMLD and BILD show a similar pattern of sensitivity for suprathreshold perturbations, suggesting a close similarity between the underlying processes.

** This section is based on Goverts, S.T. ,Houtgast, T. (**2004**). "**The relation between binaural unmasking in speech detection and speech intelligibility**," submitted as a research note to the Special Issue on Spatial and Binaural Hearing of ACTA Acustica.*

I INTRODUCTION

The daily life observation that a desired signal S is less effectively masked by an undesired noise N coming from a different direction is based on binaural unmasking, i.e. the ability of the auditory system to make use of the available interaural information (a.o. Blauert, 1997). Binaural unmasking can be made measurable in a headphone experiment by taking the difference between diotic and dichotic listening. Often a design using a N0S0 presentation versus a N0Sπ presentation is used, but also other configurations are known with different degree of phase shift (N0Sϕ) or with homophasic signal and interaural phase shift in the noise (NϕS0).

The majority of research on binaural unmasking, starting with Hirsh (1948), is concerned with the Binaural Masking Level Difference (BMLD), the difference in masked thresholds of tones in noise for diotic and dichotic presentation of stimuli. The major relevance of binaural unmasking, however, is its contribution to spatial speech perception in noisy environments, the so called cocktail party effect (Cherry, 1953; Bronkhorst, 2000). For speech as the target signal S, binaural unmasking occurs not only in detection yielding a BMLD for speech, but also in intelligibility. The difference in speech reception threshold in noise for diotic and dichotic presentation of stimuli is named the Binaural Intelligibillity Difference (BILD). This BILD was first described by Licklider (1948).

For both speech detection and speech intelligibility, the underlying process can be thought to be described by a basic model of dichotic hearing constituted by diotic hearing and unmasking, as illustrated in Figure 4.1.1.

Often it is presumed that the process of unmasking is essentially the same for speech intelligibility and speech detection. Levitt and Rabiner (1967b) have presented a procedure to predict both the BILD and BMLD in broadband Gaussian noise, assuming the same unmasking mechanism in both phenomena. They assume that the unmasking process can be represented by an equivalent reduction of the noise, based on the frequency specific BMLD for tones. For the prediction of speech intelligibility they use the Articulation Index (AI, French and Steinberg, 1947; Kryter, 1962). For the prediction of speech detection a signal-to-noise (S/N) criterion based on the long term rms speech spectrum is used (Kryter, 1962). The results of this prediction procedure are fairly well in line with experimental results the authors

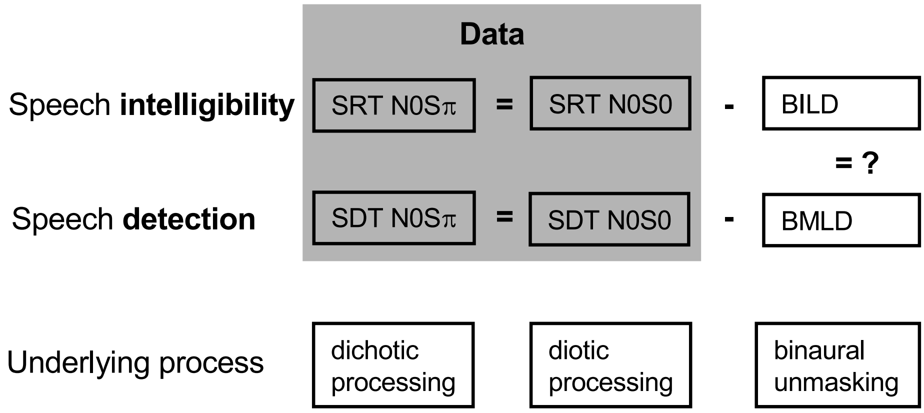


Figure 4.1.1 Schematic overview of the relation between data on dichotic and diotic processing in speech detection and speech intelligibility and binaural unmasking, as well and the underlying process they possibly have in common.

published earlier (Levitt and Rabiner, 1967a). They conclude that the BILD decreases with increasing AI. The dependence of BILD on AI is confirmed recently by Johansson and Arlinger (2002) who investigated unmasking in speech detection and speech intelligibility for Swedish spondaic words. Wilson et al. (1982) studied the homogeneity of the unmasking in detection and speech intelligibility for 36 CID W-1 spondaic words establishing psychometric curves as a function of S/N ratio for the N0S0 and N0Sπ presentation modes. They found a dependence of unmasking on S/N ratio, more so for intelligibility than for detection.

While the above mentioned literature is in line with a similarity between binaural unmasking in speech detection and speech intelligibility for normal hearing from an audibility point of view, the present study considers the similarity between BMLD and BILD from a suprathreshold perspective. Aiming to investigate the BILD for sensory-neurally hearing-impaired subjects using a distortion sensitivity approach (van Schijndel et al., 2001a; van Schijndel et al. 2001b; Houtgast, 1995; Goverts et al., 2004b) the question arose whether its underlying mechanism is affected similarly by suprathreshold deficits as the BMLD for speech signals.

Audibility and suprathreshold coding play a role in speech intelligibility and speech detection. The differences are obvious. For speech detection the role of audibility can be described by a simple energy detection model. For speech intelligibility the role of audibility is described by the articulation index model (e.g. French and Steinberg, 1947; Pavlovic, 1984) basically amounting to an importance weighting of frequency bands. Turner et al. (1992) measured psychometric curves for the detection and recognition of consonants in noise for normal hearing and hearing-impaired subjects. They found normal results on speech detection but poorer recognition scores for the hearing-impaired subjects and concluded that they suffer from a specific deficit in utilizing speech cues. For speech intelligibility more subtle suprathreshold processing is required because of e.g. discrimination, segmentation and segregation. Based on these differences it is not obvious that BMLD and BILD for speech reflect the same underlying mechanism.

In the present research speech detection and speech intelligibility are measured in both the N0S0 and N0Sπ presentation mode while speech and noise signals are degraded in several domains in a way aiming to mimic the possible detrimental effect of a cochlear deficit. Signals are perturbed with respect to their representation in the domains of phase, frequency, time and intensity. Thus introducing degradation in four domains for the six subjects, in fact an additional number of 4*6 = 24 subjects with an 'artificial hearing impairment' is introduced. This results in 30 data points for unmasking in speech detection (BMLD) and speech intelligibility (BILD).

The leading assumption is that a similarity of the processes underlying the BMLD and the BILD will lead to a similar pattern of effects for the various types of degradation, resulting in a high correlation between the 30 data points of both measures.

II INTRODUCTION OF SIGNAL PERTURBATION

A wavelet decomposition scheme was used to introduce perturbations in the domains of phase, frequency, time or intensity in the speech and noise material. The main advantage of the wavelet decomposition is its resemblance to auditory processing because of its proportional spectro-temporal analysis window based on a 'mother wavelet'. A brief overview of the approach will be given below. An extensive overview on wavelet analysis is given by e.g. Rioul and Vetterli (1991), Vetterli and Kovačević (1995), Strang and Nguyen

(1996). Details and examples on the application of this analysis tool in auditory research are given by van Schijndel, 2000 and van Schijndel et al., 2001a;2001b, investigating the influence of distorted auditory coding on monaural speech perception.
In accordance with van Schijndel et al. a Gaussian-windowed sinusoid was chosen as mother wavelet:

$$s(t) = \sqrt{\alpha f_0}\ \exp(i2\pi f_0 t)\ \exp(-\pi(\alpha f_0 t)^2)$$

in which f_0 is the carrier frequency, α the so called *shape factor* and $\sqrt{\alpha f_0}$ normalizes the energy of the analysis function. The effective bandwidth of this time-frequency window is given by αf_0 and its effective duration by $1/(\alpha f_0)$. Van Schijndel et al. chose a shape factor of 0.1735, resulting in an effective bandwidth of 1/4 octave, which roughly corresponds to the auditory critical bandwidth (e.g. Scharf, 1970; Moore, 1995).
A basis for signal decomposition is constructed with scaled versions of this mother wavelet. As a basis for wavelet analysis should be orthonormal, wavelets ought to have a compact support. Because this is not the case for such a Gaussian wavelet the range between the points that are 25 dB down the peak is taken as the significant range of the wavelet. Van Schijndel et al. demonstrated that for adequate sampling in frequency and time a basis consisting of eight wavelets per octave along the spectral axis and one wavelet per three periods of the wavelet carrier frequency is needed. All the signals were bandwidth limited between 250 and 4000 Hz. This results in 33 wavelets along the spectral axis and depending on the carrier frequency, between 83 and 1333 wavelets per second along the temporal axis. Thus, a speech signal of one second can after decomposition on this base be described by about $16 \cdot 10^3$ wavelet coefficients.

Degradation of stimuli

The process of degradation of signal and noise is essentially as follows

1 Selection of sentence file and speech shaped noise file.

2 Decomposition of both sentence and noise on the above described base, resulting in a *coefficient file.*

3 Statistical perturbation of all wavelets constituting the base in one of the four domains (phase, frequency, time or intensity) by a random factor R . This factor is chosen from a uniform distribution characterized by a perturbation value (PV), determining the ranges from minus PV/2 to plus PV/2. The perturbation process results in a *perturbed base for the signal and noise to be presented to the left ear*. Details about the perturbation process are given below.

4 Recomposition of sentence and noise file with coefficients from the *coefficient file* based on the *perturbed base* for the left ear stimuli and normalization of energy of the stimuli.

5 A similar independent processing as described in steps 3 and 4 for the right ear stimuli.

Perturbation in four domains

The perturbation values for the four domains are listed in Table 4.1.1 Perturbations in the phase domain were introduced by random temporal displacement of each wavelet as a whole, both envelope and fine structure. The perturbation value (PV) was 0.038 wavelet, so the maximal displacement was plus or minus 0.019 wavelets, corresponding to 0.11π. Perturbations in the frequency domain were introduced by shifting the carrier frequency of the wavelet, thus shifting it up or down along the spectral axis. The perturbation value (PV) was 0.75 octave, so the maximal displacement was plus or minus 0.38 octaves. Perturbations in the time domain were introduced by shifting the temporal position of the envelope, while keeping the fine structure constant by extrapolation.

Table 4.1.1 Overview of the amount of perturbation that is introduced in the four domains expressed by the value of the Perturbation Value (PV). Each wavelet is shifted by a factor R, randomly chosen from an uniform distribution ranging from minus PV/2 to plus PV/2.

condition	Perturbation Value (PV)	maximal perturbation
no degradation	no	no
phase	0.038 wavelet	0.11π
frequency	0.75 octave	0.38 octave
time	7 wavelets	20 local periods
intensity	20 dB	10 dB

The perturbation value (PV) was 7 wavelets, so the maximal displacement was plus or minus 3.5 times the effective wavelet duration. This equals to about 20 local periods.
Perturbations in the intensity domain were introduced by multiplying each wavelet coefficient by a random factor. The perturbation value (PV) was 20 dB, so the maximal perturbation yielded plus or minus 10 dB.
In interpreting the results, it should be noted that these four types of perturbations are not really completely mutually independent. For instance, the frequency-domain perturbations will also affect the momentary inter-aural phase relations, and so will the time-domain perturbations. We could not find a way to completely avoid such interactions among the various types of perturbations.

III METHOD

Subjects

Six normal-hearing subjects, aged 26 to 58 years with a mean age of 34, participated in the experiment. Pure tone air conduction thresholds of the normal hearing listeners did not exceed 15 dB HL at any octave frequency from 250 to 4000 Hz.

Procedures and conditions

Speech Detection Thresholds (SDT) and Speech Reception Thresholds (SRT) were measured in the N0S0 and N0Sπ presentation mode in five conditions as listed in Table 4.1.1. Speech Reception Thresholds were measured in an adaptive procedure as described by Plomp and Mimpen (1979). However, to avoid unwanted high presentation levels, the speech signal was held constant while the noise level was varied.

Speech Detection Thresholds were measured by means of a common 2AFC procedure (2 up, 1 down-4 dB steps), leading to an estimate of the 71% correct score (Levitt, 1971). For each condition and subject a retest was done. The order of conditions was fixed, because this research was part of a larger project in which also hearing-impaired subjects participated and it was aimed to compare subjects (Goverts et al. 2004b). The speech material as described by Versfeld et al. (2000) was used, consisting of lists of 13 everyday Dutch sentences of eight to nine syllables read by a male voice. The combination of list and condition was held constant and was the same for SRT and SDT measurements.

Stimulus presentation during SDT and SRT measurements

During the SDT and SRT measurements the speech and noise signals that were degraded in accordance with the method described in section 2 were added in the appropriate signal-to-noise ratio. The main reason for this procedure in stead of preparing al stimuli in all possible S/N ratio’s was practically driven by considerations of calculation time and storage capacity. In N0Sπ conditions a phase shift was introduced by inverting the stimulus waveform. In order to make the results comparable to hearing impaired data (not included in this chapter), the stimuli were filtered and amplified such that the long term speech spectrum was in the middle of the normal dynamic range.

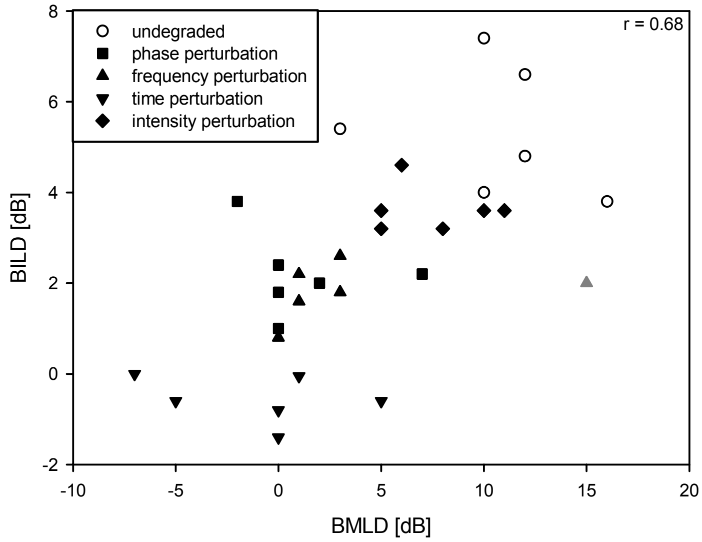


Figure 4.1.2 Binaural unmasking in speech intelligibility versus binaural unmasking in speech detection for the six normal-hearing subjects with undegraded speech and noise signals and with the four types of signal perturbation. One subject, having an exceptionally large difference in test-retest data for $SDT_{N0S\pi}$ in the frequency distortion condition, is indicated by a grey triangle.

IV RESULTS AND DISCUSSION

In Figure 4.1.2 binaural unmasking in speech intelligibility is plotted versus binaural unmasking in speech detection. As a reference also the SDT and SRT data for diotic processing are given in Figure 4.1.3. An overview of correlations coefficients between data on diotic (N0S0) and dichotic (N0Sπ) processing in speech detection and speech intelligibility, as well as the calculated values of unmasking is given in Table 4.1.2 While one subject showed an exceptionally large difference in test-retest data for $SDT_{N0S\pi}$ in the frequency distortion condition these data are not incluced in the analysis below. The corresponding data point is indicated by a grey triangle in Figures 4.1.2 and 4.1.3. From Table 4.1.2 and Figure 4.1.2 a correlation between the BMLD and BILD data can be observed indicating that binaural unmasking in speech detection and speech intelligibility is affected similarly by the perturbations and thus suggesting that BMLD and BILD are based on similar cues (see Figure 4.1.1). No such relation is observed for the diotic (N0S0) SDT and SRT,

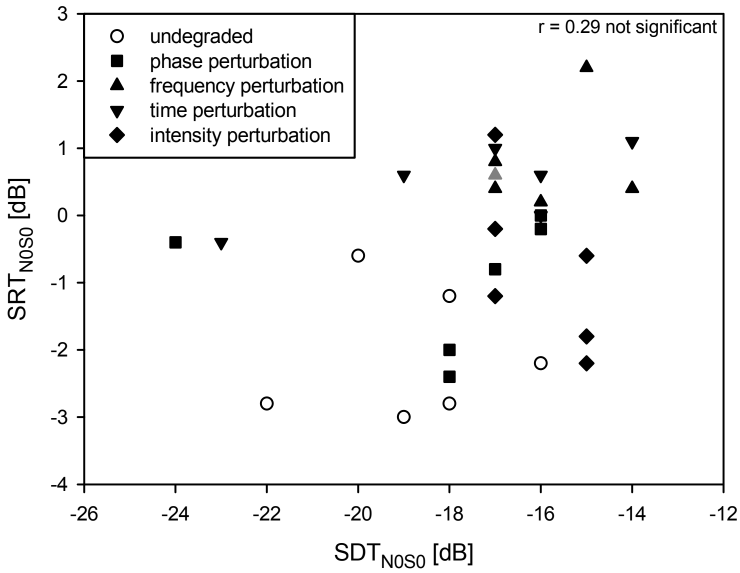


Figure 4.1.3 Diotic speech reception threshold (SRT) versus diotic speech detection threshold for the six normal-hearing subjects with undegraded speech and noise signals and with the four types of signal perturbation. One subject, having an exceptionally large difference in test-retest data for $SDT_{N0S\pi}$ in the frequency distortion condition, is indicated by a grey triangle.

indicating that diotic processing in speech detection and speech intelligibility is not affected similarly by the perturbations. This suggests, not surprisingly, that diotic detection and intelligibility of speech are not based on the same cues. Interestingly, the diotic SDT data are not correlated to any of the speech intelligibility data, while the dichotic (N0Sπ) SDT and BMLD data are correlated to all of the speech intelligibility measures. This suggests that $SDT_{N0S\pi}$, BMLD, SRT_{N0S0} $SRT_{N0S\pi}$ and BILD somehow make use of the same cues.

Table 4.1.2 Correlation coefficients between data of diotic hearing and unmasking in the domain of speech detection and speech intelligibility for the six subjects in the five conditions. While one data point is left out (see text) the correlation coefficients are calculated for 29 data points. For the sake of completeness also data for the N0Sπ presentation mode are given. Significant correlations at the $p < 0.01$ level are plotted bold.

correlation N = 29		intelligibility		
		diotic(SRT_{N0S0})	dichotic ($SRT_{N0S\pi}$)	unmasking (BILD)
detection	diotic (SDT_{N0S0})	0.29	0.32	-0.28
	dichotic ($SDT_{N0S\pi}$)	**0.62**	**0.83**	**-0.80**
	unmasking (BMLD)	**-0.49**	**-0.69**	**0.68**

The maximal correlation to be expected

The true correlation between two measures, m1 and m2, is attenuated by their accuracy. If this inaccuracy is caused by measurement error, the true correlation (r_{true}) between two measures, m1 and m2, can be estimated by

$$r_{true}^2 = r_{observed}^2/(\alpha_{m1} * \alpha_{m2}) \qquad (1)$$

where α_{m1} and α_{m2} represent the accuracy of m1 and m2, expressed for example by the Cronbach alpha (e.g. Nunnally, 1967; Guilford, 1954). It should be noted that the thus established value for r_{true} is only an estimation with a confidence interval, which depends on the number of observations, the value of α_{m1} and α_{m2} and the actual value of $r_{observed}$..
Based on this formula, using the test and retest data to calculate the values of Cronbach alpha, we can estimate the true underlying correlations. The results of this procedure are shown in Table 4.1.3. The limited accuracy of the tests turns out to have attenuated all correlations. Correcting for this attenuation reduces the differences in the coherence of $SDT_{N0S\pi}$, BMLD on the one hand and SRT_{N0S0} $SRT_{N0S\pi}$ and BILD on the other hand.

V CONCLUSION

We determined the effect of various types of suprathreshold stimulus degradations on diotic and dichotic speech perception in noise (Speech Detection Threshold and Speech Reception Threshold), in order to examine the relation between binaural unmasking in speech detection and speech intelligibility, i.e. BMLD and BILD. The leading assumption is that if there is a common underlying process in unmasking, BMLD and BILD will be affected in a related way by the various types of degradation, resulting in a correlation between both measures. This leads to the following conclusions.
Unmasking in speech detection and speech intelligibility is closely related. While their absolute difference can be explained in terms of audibility as demonstrated by Levitt and Rabiner (1967b), BMLD and BILD are similarly affected by suprathreshold perturbations.

Table 4.1.3 As Table 4.1.2, indicating the estimated true correlation coefficients.

correlation N = 29		intelligibility		
		diotic (SRT_{N0S0})	dichotic ($SRT_{N0S\pi}$)	unmasking (BILD)
detection	diotic (SDT_{N0S0})	0.61	0.46	-0.44
	dichotic ($SDT_{N0S\pi}$)	**1.13**	**1.03**	**-1.08**
	unmasking (BMLD)	**-1.00**	**-0.95**	**1.02**

This strongly suggests that the underlying processes are based on a similar set of stimulus cues. The same analysis also indicates that diotic speech detection is the outlier: the low correlation suggests that this process utilizes different cues than the processes involved in speech intelligibility and binaural unmasking.

ACKNOWLEDGEMENTS

The authors thank Tino Trahiotis for drawing our attention to the relation between detection and intelligibility in binaural unmasking of speech. Furthermore, they are grateful to Hans van Beek for technical assistance and to Erwin George for assistance in conducting the experiments.

4.2 The BILD of hearing-impaired listeners - the role of suprathreshold coding*

A reduced binaural performance of hearing-impaired listeners may be caused by the raised hearing threshold (reduced audibility), but also by suprathreshold coding deficits of essential signal cues. In this study the binaural performance is operationalized by the Binaural Intelligibility Level Difference (BILD), being the improvement of the Speech Reception Threshold for the N0Sπ presentation mode relative to the diotic N0S0 condition. The BILD is examined for 25 mild to moderate sensorineural hearing-impaired listeners, while optimal audibility is assured. A distortion sensitivity approach is used, investigating the sensitivity of a subject's BILD to external stimulus perturbations in the domains of phase, frequency, time or intensity. The underlying assumption of this approach is that an auditory coding deficit of a cue in a particular domain will result in a low sensitivity to external perturbations applied in that same domain. Only eight listeners appear to have a significantly reduced BILD. The distortion sensitivity data for these listeners suggest that this reduction is caused mainly by sub-optimal auditory coding in the domains of phase and time. The mechanisms underlying binaural unmasking and monaural speech reception in noise seem to utilize only partly the same suprathreshold cues. As the suprathreshold coding problems associated with a given sensorineural hearing loss can vary in type and extent of deficits, the effects of such a hearing loss on these mechanisms can be manifold.

** This section is based on Goverts, S.T., Houtgast, T., van Beek, J.H.M., (**2004**). "**The BILD of hearing-impaired listeners - the role of suprathreshold coding**," to be submitted to JASA*

I INTRODUCTION

The human beings amazing capacity of spatial hearing is of great importance because it extends our perceptual world beyond the visual field. The combination of selective attention, sensitivity and localization accuracy provides us with an real time acoustical image of our surroundings that is essential from a perspective of social and communicative interaction and safety.

The multiple processes, bottom up as well as top down (Blauert, 1997), that are involved are even for normal-hearing subjects far from fully understood. From the fact that in many types of hearing impairment reduced spatial hearing is a very common complaint it can be concluded that the processes involved in spatial hearing are very sensitive to any type of impairment. A disability in understanding speech in noisy environments has a relatively high impact of in a hearing-impaired subject's self assessment of auditory functions. This fact, well known in everyday clinical practice, was reported by Kramer et al. (1998), for example. Both from a clinical and an experimental point of view, investigating the reduced hearing-impaired performance on spatial hearing is of great importance.

A large amount of experimental data on both spatial and binaural hearing reveal a reduced overall performance of hearing-impaired subjects compared to normal hearing. In most of the studies however a large variation between subjects is found, ranging from close to normal to very abnormal, which cannot be explained on base of audiometric data (for example Durlach, Thompson and Colburn, 1981; Colburn 1982;Bronkhorst and Plomp,1990; Koehnke and Besing, 1997; Kramer et al. 1996; Gabriel et al., 1992). In the monaural domain, recent research (Noordhoek, 2001; van Schijndel 2001a, 2001b) has renewed the insight brought up earlier (e.g. Moore,1995; Plomp,1978 and 1986) that auditory problems in speech perception are probably determined by more than audibility related factors and should thus be characterized more dimensional.

The bottom-up processes involved in spatial hearing consist, roughly speaking, of *audibility*, *suprathreshold coding* and *monaural* and *binaural* processing. In earlier studies on the precedence effect (Goverts et al., 2000 and 2002) an effect of presentation level for normal hearing was found. Hearing-impaired subjects showed a reduced performance, even while corrected for absolute level or sensation level. Also a substantial inter-individual spread was found, even for subjects with similar audiometric configuration.

This led to the central question of the current study: can we differentiate between the contribution of reduced audibility and of suprathreshold coding deficits to the reduced binaural performance of sensorineurally hearing-impaired subjects. In this study a *distortion sensitivity* approach (Houtgast, 1995; van Schijndel et al. 2001a and 2001b) is used in an attempt to answer this question. The basic idea of this approach is illustrated in Figure 4.2.1. Suppose that the performance of a hearing-impaired subject on a psychophysical test is poorer than that of the normal hearing. We are interested to know if this reduced performance is related to a suprathreshold coding deficit, for instance in the spectral domain. To study that, we measure test performance as a function of the degree of external stimulus distortions in the spectral domain (the precise way these perturbations are induced in the stimulus will be discussed later). The leading assumption is that if the performance of a hearing-impaired subject is found to be less sensitive to this distortion than that of normal-hearing subjects, the impaired auditory system does not make full use of this type of information because of a similar type of 'internal' distortion. This would lead to "convergence" in the distortion-sensitivity model (see Figure 4.2.1). Thus, this implies that the reduced performance on this test is indeed related to a suprathreshold spectral-coding deficit. Note also that it is to be expected that the more the task performance is reduced in the undistorted condition (thus, the stronger the spectral-coding deficit), the more the distortion sensitivity will be reduced. If, on the other hand, the effect of distorting the spectral information is found to be similar to that

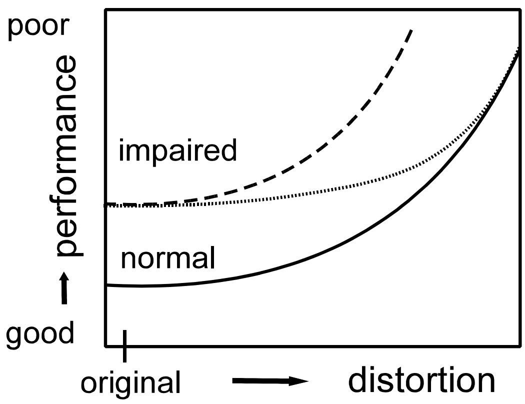


Figure 4.2.1 Illustration of the distortion-senstivity model. Performance for hearing-impaired subjects as a function of distortion is compared with that of normal-hearing subjects (solid line). The possible outcome of such an experiment is "convergence (dotted and solid lines) or "no convergence" (dashed and solid lines). Redrawn from van Schijndel et al. (2001b) with permission.

for normal hearing, it is assumed that spectral cues are coded as accurately as in normal hearing (no suprathreshold spectral-coding deficit). This would lead to “no convergence” in the distortion-sensitivity model (see Figure 4.2.1).

Van Schijndel et al.(2001a; 2001b) used this approach to study suprathreshold effects causing reduced monaural speech perception of hearing-impaired subjects. Using wavelet coding as a tool, they introduced perturbation in either time, frequency or intensity in monaural speech and noise stimuli. Finding a reduced sensitivity to distortions in the frequency and a normal sensitivity to distortions in the time domain, they concluded that the suprathreshold problems of these hearing impaired were mainly due to coding problems with respect to spectral information. The results on the contribution of intensity coding problems were not quite conclusive.

Figure 4.2.2 gives a schematic overview of the way this distortion sensitivity approach is applied to investigate suprathreshold coding in binaural hearing. Aiming to develop a test that is useful for a clinical and experimental test battery, a binaural speech-intelligibility based test

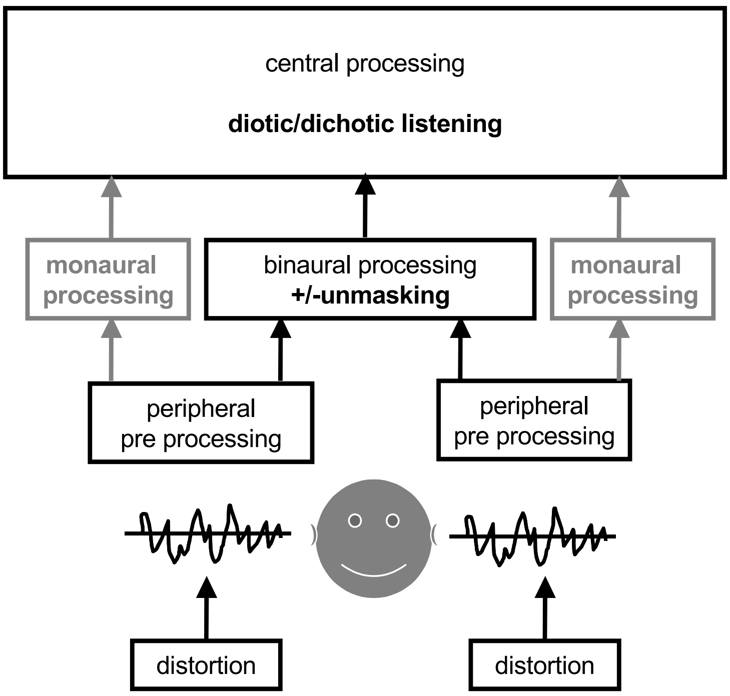


Figure 4.2.2 Schematic overview of the way the distortion-sensitivity approach is applied to investigate the role of suprathreshold coding in binaural unmasking. Speech reception thresholds are measured in diotic (N0S0) and dichotic (N0Sπ) presentation mode, while speech and noise signals are distorted in the domains of phase, frequency, time and intensity. Only the N0Sπ stimuli involve the binaural processing stage. Since the diotic N0S0 stimuli contain no binaural information, they can be considered as an estimation of monaural speech perception.

measuring the Binaural Intelligibility Level Difference (BILD) was chosen in stead of a more demanding sound lateralization test. The BILD was first described by Licklider (1948). It is a manifestation of binaural unmasking, the advantage of binaural over monaural hearing of a signal S against the background of a spatially separated noise N. In headphone experiments, often a design with a N0S0 presentation versus a N0Sπ presentation is used, in which the noise is presented homophasic and the signal either homophasic or antiphasic. The BILD is then defined as the difference in the speech reception threshold (SRT) in the N0S0 and N0Sπ presentation mode. Olsen et al. (1976) have explored the use of the BILD for diagnosis of retro-cochlear pathology. However, the majority of research on binaural unmasking is concerned with the binaural masking level difference (BMLD), which is defined as the difference in detection threshold between the diotic and dichotic presentation mode. Levitt and Rabiner (1967a;1967b) demonstrated a similarity between binaural unmasking in speech detection and speech intelligibility for normal hearing from an audibility point of view. In a research note submitted to ACTA Acustica (Goverts and Houtgast, 2004) we demonstrated for normal hearing that these processes underlying BILD and BMLD are based on a similar set of stimulus cues. Blauert (1997) provides an overview of experimental work on BMLD and BILD. In a recent publication Johansson and Arlinger (2002) drew the attention again to the clinical use of the BILD.

Outline of the present study

Firstly, for normal hearing the effect of presentation level on the BILD will be examined, in order to investigate the role of audibility. Then, the sensitivity of the BILD to different types of distortions will be determined for six normal-hearing subjects (reference data needed for the distortion sensitivity approach) and 25 mild to moderate sensorineurally hearing-impaired subjects. To rule out audibility effects all speech and noise stimuli are presented in the middle of the individual subject's dynamic range.

Estimating the BILD requires SRT measurements in the N0S0 and N0Sπ presentation modes. From Figure 4.2.2 it is clear that only the N0Sπ stimuli involve the binaural processing stage. Since the diotic N0S0 stimuli contain no binaural information, they can be considered as an estimation of monaural speech perception (Siegel and Colburn, 1983). By comparing the distortion sensitivity data for BILD and SRT_{N0S0} we can distinguish between suprathreshold coding deficits involved in the process of binaural unmasking and those involved in monaural speech reception in noise.

Summarizing, the research questions addressed in this study are

- What is the effect of presentation level on the BILD for normal-hearing subjects?
- What is the sensitivity of normal-hearing subjects to bilateral independent distortion of cues of phase, frequency, time or intensity, with respect to the BILD and the SRT_{N0S0}? (Reference data for the distortion sensitivity approach.)
- To what extent do hearing-impaired subjects have a reduced BILD while optimal audibility is assured?
- For which of the four types of distortion do we find a reduced sensitivity of hearing-impaired subjects as compared to normal hearing, with respect to the BILD and SRT_{N0S0}?
- Do the individual data reveal the relation between distortion sensitivity and task performance, as expected within the framework of the distortion sensitivity approach?
- To what extent are the processes underlying binaural unmasking and monaural speech reception in noise, as quantified by BILD and SRT_{N0S0} related for the individual subjects?

II. METHOD

A. Introduction of distortion in speech material

In accordance with the studies of van Schijndel et al.(2000, 2001a, 2001b) a wavelet decomposition scheme was used to introduce distortion in the speech and noise stimuli. The main advantage of the wavelet decomposition is its resemblance to auditory processing because of its proportional spectro-temporal analysis window based on a ‘mother wavelet’. A detailed description of procedures and considerations is given in Appendix A. In the following a brief overview is given. Stimuli are band bass filtered between 250 and 4000 Hz. Van Schijndel et al. introduced distortion in the frequency, time and intensity domain. ergens anders Realizing that degrading the frequency coding inevitably implied degrading phase information, they introduced a fixed amount of phase distortion in the frequency-distortion conditions as well. Because of the obvious relevance of interaural phase information to binaural hearing, distortion in the phase domain is considered separately in the present study, next to distortion in the domains of frequency, time and intensity. These four types of perturbations are not really completely mutually independent. Especially, the frequency-domain perturbations will also affect the momentary inter-aural phase relations, and so will the time-domain perturbations. Estimating the amount of phase distortion by the interaural de-correlation of the signals, it was tried to choose conditions for phase perturbation that matched the amount of unwantedly *induced* phase distortion by perturbing frequency and time. The effects of real distortion in the domains of frequency and time can be separated from the effect of unwantedly induced phase distortions by comparing results in conditions with equal interaural de-correlation.

Essentially the scheme of signal degradation consisted of (1) wavelet decomposition, (2) perturbation of the wavelet coefficients, (3) re-composition of the perturbed wavelets, (4) re-scaling of the signal. Perturbations were introduced by shifting each wavelet coefficient by a random factor R . This factor was chosen from a uniform distribution characterized by a perturbation value (PV), determining the ranges from minus PV/2 to plus PV/2. Details about the perturbation process are given in the appendix. In each domain two conditions, determined by two values of PV (PV I and PV II), were chosen in line with the van Schijndel et al. study. The conditions are listed in Table 4.2.1. Conditions with equal interaural de-correlation are indicated by # and † respectively.

Table 4.2.1 Overview of the amount of perturbation that is introduced in the four domains expressed by the perturbation value (PV). Each wavelet is shifted by a factor R, randomly chosen from an uniform distribution ranging from -PV/2 to PV/2. The conditions phase PV I and frequency PV I indicated by # have equal interaural de-correlation as well as conditions phase PV II, frequency PV II and time PV I indicated by †.

domain	**PV I**	**PV II**
phase	0.019 wavelet #	0.038 wavelet †
frequency	0.25 octave #	0.75 octave †
time	3 wavelets †	7 wavelets
intensity	10 dB	20 dB

B. Subjects

Six normal-hearing subjects, aged 26 to 58 years with a mean age of 34, participated in the experiment. Pure tone air conduction thresholds of the normal hearing listeners did not exceed 15 dB HL at any octave frequency from 250 to 4000 Hz. Twenty-five mild to moderate sensorineurally hearing-impaired, aged 44 to 79 years with a mean age of 61, took part in the experiment. The mean pure tone air conduction threshold ranged from about 20 dB at 250 Hz to 65 dB at 8000 Hz. The hearing losses were essentially symmetrical, with mean absolute differences ranging from 3 dB at 250 Hz to 7 dB at 4000 Hz. All subjects were native speakers of Dutch.

C. Stimuli and apparatus

The speech stimuli consisted of lists of 13 everyday Dutch sentences of eight to nine syllables read by a female or a male speaker. The masking noise had a long-term spectrum that resembled the long-term spectrum of the male and female voice respectively (Versfeld et al. 2000). Because of the large amount of sentence lists needed to perform test en retest SRT measurements in all conditions it was decided to perform the undistorted SRT measurements at various levels (Section III-A) using the female lists and the SRT measurements in the distorted conditions using the male lists.
Signals were generated by a Soundblaster compatible soundcard. Stimuli were presented bilaterally through Beyer DT48 headphones after amplification by an Shure FP 22 amplifier.. For calibration, sound pressure levels of the stimuli were measured on a Bruel & Kjaer type 4152 artificial ear with flat-plate adapter. The entire experiment was controlled by a personal computer. Subjects were tested individually in a soundproof room.

D. Procedures

Assuring audibility
To assure optimal audibility, it was chosen to present stimuli in the middle of the individual's dynamic range over the frequency range 250 to 4000 Hz. For practical reasons, clinically established thresholds and Uncomfortable Loudness Levels for the audiometric frequencies

250, 500, 1000, 2000 and 4000 Hz were used to determine shape and level of the target spectrum for both speech and masking noise.

Measuring speech intelligibility

The BILD is defined as the difference between the speech reception threshold in noise in diotic presentation mode (SRT_{N0S0}) and in dichotic presentation mode with antiphasic speech ($SRT_{N0S\pi}$). The SRT in noise is defined as the signal-to-noise ratio (SNR) at which 50% of the sentences are reproduced correctly. Measurement procedures were in accordance with Plomp and Mimpen (1979), varying the SNR adaptively in an up-down procedure using 2-dB steps. Each single SRT-measurement is based on one 13-sentence list. All measurement are repeated once (test and retest). In order not to exceed the levels of uncomfortable loudness for patients with a small dynamic range it was decided to keep the speech signal fixed in the middle of the individual's dynamic range and to vary the noise level.

III. RESULTS

A. Undistorted SRT measurements at various levels (normal hearing)

For the six normal-hearing subjects the SRT was measured with N0S0 and N0Sπ presentation as described in the Methods section. This condition is referred to as the reference condition. Keeping the spectral shape of speech and noise constant, measurements were done also at levels 15 and 30 dB below and 15 dB above that of the reference condition. Mean data for SRT_{N0S0} and $SRT_{N0S\pi}$, and calculated BILD values as well as standard deviations among subjects are given in Table 4.2.2.

Table 4.2.2 Mean SRT values and standard deviations for six normal-hearing subjects in diotic and dichotic (N0Sπ) presentation mode as well as mean calculated BILD values as a function of presentation level, expressed relative to the reference level in the middle of the subject's dynamic range.

SRT [dB]		**normal hearing (N = 6)**	
presentation	level	mean	standard deviation
N0S0	reference	-0.3	1.4
N0Sπ	reference	-5.5	1.8
BILD	*reference*	*5.2*	*0.4*
N0S0	reference + 15 dB	0.4	1.2
N0Sπ	reference + 15 dB	-2.7	1.6
BILD	*reference + 15 dB*	*3.1*	*1.4*
N0S0	reference - 15 dB	-1.7	0.7
N0Sπ	reference - 15 dB	-7.3	1.1
BILD	*reference - 15 dB*	*5.6*	*0.8*
N0S0	reference - 30 dB	-1.4	2.6
N0Sπ	reference - 30 dB	-4.9	5.2
BILD	*reference - 30 dB*	*3.6*	*2.6*

B. Results of SRT measurements in all distorted conditions

For the six normal hearing and 25 hearing-impaired subjects the SRT was measured with N0S0 and N0Sπ presentation, for several degrees of distortion in four domains. The conditions were presented in a fixed order, using the same order of sentence lists, because of the interest in differences between subjects. While the hearing impaired results cannot be assumed to be normally distributed, presentation of the data in terms of the median and inter-quartile ranges is preferred above mean and standard deviation. For the normal hearing and hearing-impaired subjects median SRT_{N0S0} and $SRT_{N0S\pi}$ values, as well as inter-quartile ranges for all conditions are given in Table 4.2.3.[1]

[1]The SRT_{N0S0} and $SRT_{N0S\pi}$ results in the undistorted condition, for the normal-hearing listeners deviate from the results shown in Table 4.2.2. We cannot fully explain this difference, it might to be due to different effect of the bandlimiting procedure for the male and female sentence material. The BILD's values are in agreement.

Table 4.2.3 Median SRT values and quartile ranges for six normal-hearing and 25 hearing-impaired subjects in diotic and dichotic (N0Sπ) presentation mode for the undistorted conditions and in the eight distorted conditions.

SRT [dB]		**normal hearing (N = 6)**			**hearing impaired (N = 25)**		
presentation	distortion	median	1st quartile	3rd quartile	median	1st quartile	3rd quartile
N0S0	no	-2.5	-2.8	-1.5	0.4	-0.6	1.4
N0Sπ	no	-7.3	-8.2	-6.3	-4.0	-5.8	-2.6
N0S0	phase I	-2.1	-2.6	-1.2	-0.2	-1.6	1.0
N0Sπ	phase I	-6.0	-6.5	-5.8	-3.6	-5.0	-2.8
N0S0	phase II	-0.6	-1.7	-0.3	0.4	-0.6	1.8
N0Sπ	phase II	-3.2	-4.0	-2.4	-1.8	-3.2	-0.6
N0S0	frequency I	-1.7	-2.4	-1.5	-1.2	-1.6	0.0
N0Sπ	frequency I	-5.0	-5.7	-4.2	-3.4	-4.2	-3.0
N0S0	frequency II	0.5	0.4	0.7	2.0	1.2	3.0
N0Sπ	frequency II	-1.1	-1.7	-0.7	0.0	-1.2	0.8
N0S0	time I	-0.8	-1.2	-0.2	1.8	0.4	2.8
N0Sπ	time I	-1.6	-1.8	-1.5	-0.2	-1.0	1.2
N0S0	time II	0.6	0.1	0.9	3.0	1.4	5.2
N0Sπ	time II	1.3	0.8	1.6	3.4	2.0	4.4
N0S0	intensity I	-2.0	-2.5	-1.7	-1.0	-2.2	0.4
N0Sπ	intensity I	-6.6	-7.1	-6.1	-3.6	-5.8	-2.8
N0S0	intensity II	-0.9	-1.7	-0.3	-0.2	-1.0	0.6
N0Sπ	intensity II	-4.1	-5.2	-3.8	-2.8	-3.8	-1.4

C. Calculated BILD values and distortion sensitivity

For all subjects and all conditions, the BILD is calculated as the SRT_{N0S0} minus the $SRT_{N0S\pi}$. For the normal hearing and hearing-impaired subjects median BILD values, as well as inter-quartile ranges for all conditions of distortion are given in Table 4.2.4. As an estimate of an individual's binaural distortion sensitivity in a certain domain, we determined the slope of the first order polynomial through the BILD-values for the three levels of distortion (i.e., the undistorted and the two distorted conditions).

Table 4.2.4 Median BILD values and quartile ranges for six normal-hearing and 25 hearing-impaired subjects for the undistorted conditions and in the eight distorted conditions.

BILD [dB]	**normal hearing (N = 6)**			**hearing impaired (N = 25)**		
distortion	median	1st quartile	3rd quartile	median	1st quartile	3rd quartile
no	5.1	4.2	6.3	4.4	3.6	5.4
phase I	4.0	3.9	4.6	3.4	2.8	4.0
phase II	2.1	1.9	2.4	2.4	1.6	2.8
frequency I	3.3	2.8	3.7	2.8	2.2	3.4
frequency II	1.9	1.7	2.2	2.0	1.4	3.0
time I	0.9	0.5	1.3	1.2	0.6	2.0
time II	-0.6	-0.8	-0.2	-0.2	-1.0	1.0
intensity I	4.6	4.1	5.2	3.6	2.2	4.2
intensity II	3.6	3.3	3.6	2.4	1.6	3.2

Similarly, the diotic distortion sensitivity is estimated by the slope of the first order polynomial through the SRT_{N0S0} values for the three levels of distortion. As an example Figure 4.2.3 gives the results of an individual subject for distortion in the phase domain. It should be noted that a positive slope for the diotic SRT corresponds to a high sensitivity, while the opposite is the case for the BILD.

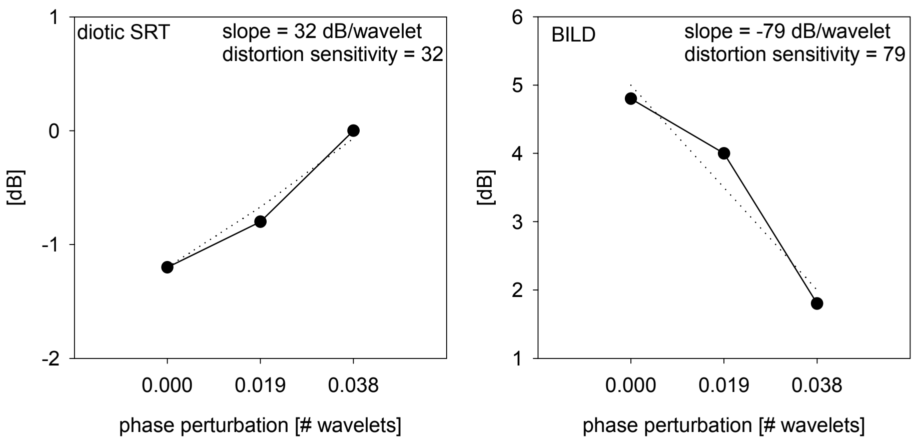


Figure 4.2.3 Illustration of the estimation of an individual's distortion sensitivity, by determining the slope of the first order polynomial through the data points at the three levels of distortion (i.e., the undistorted and the two distorted conditions). As an example, this procedure is shown for distortion in the phase-domain for an arbitrary subject. It can be noticed that a positive slope for the SRT_{N0S0} implies a high sensitivity, which is opposite for the BILD. To simplify the comparison between both types of distortion sensitivity, in the binaural case the negative slope is taken.

Table 4.2.5 Distortion sensitivity for diotic SRT and BILD, calculated as this first order slope of the SRT and BILD data respectively versus the amount of distortion in a certain domain. Median data and quartile ranges are given for six normal-hearing and 25 hearing-impaired subjects for the four domains of distortion. Since the perturbation units are quite arbitrarily, and are very different among the four perturbation domains, the slope values (dB per perturbation unit) vary widely for the four domains.

distortion sensitivity		**normal hearing (N = 6)**			**hearing impaired (N = 25)**		
	distortion	median	quartile 1	quartile 3	median	quartile 1	quartile 3
SRT_{N0S0}	phase	39	24	55	-5	-26	16
BILD	phase	79	55	107	68	37	84
SRT_{N0S0}	frequency	4.34	3.79	4.81	2.51	0.97	3.26
BILD	frequency	3.51	2.94	5.63	2.69	2.23	3.43
SRT_{N0S0}	time	0.36	0.32	0.48	0.39	0.24	0.45
BILD	time	0.76	0.70	0.86	0.70	0.45	0.81
SRT_{N0S0}	intensity	0.06	0.03	0.13	-0.05	-0.08	-0.01
BILD	intensity	0.09	0.04	0.14	0.09	0.06	0.16

To simplify the comparison between both types of distortion sensitivity, in the binaural case the inverse of the slope is taken.

For the normal hearing and hearing-impaired subjects median values for distortion sensitivity in BILD and SRT_{N0S0} in all four domains, as well as inter-quartile ranges are given in Table 4.2.5. Since the perturbation units are quite arbitrarily, and are very different among the four perturbation domains, the slope values (dB per perturbation unit) vary widely for the four domains. However, within each perturbation domain the values can be compared among normal hearing and hearing-impaired subjects, and among BILD and SRT_{N0S0}.

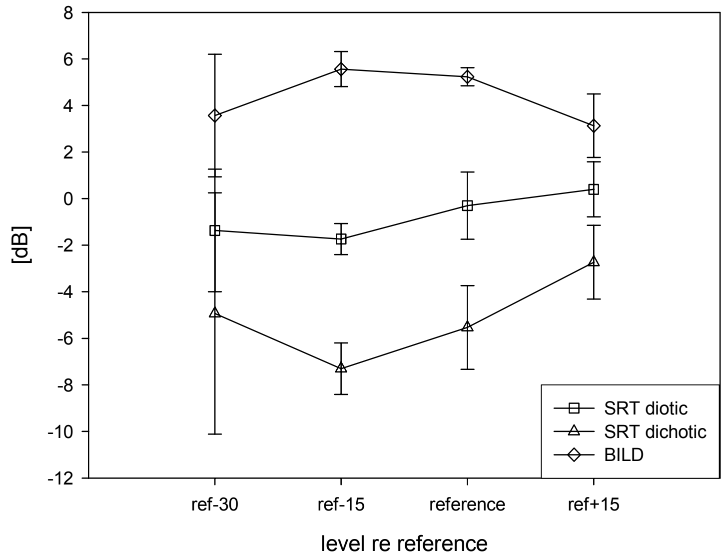


Figure 4.2.4 Mean data of six normal-hearing subjects for SRT_{N0S0} (open squares), $SRT_{N0S\pi}$ (open triangles) and mean calculated BILD values (open diamonds) as a function of presentation level. Presentation level is expressed relative to the reference level determined by the speech spectrum in the middle of a subject's dynamic range. The error-bars represent the standard deviation.

IV. DATA ANALYSIS IN RELATION TO THE RESEARCH QUESTIONS

- ***The effect of presentation level on the BILD for normal-hearing subjects***

In order to answer the question on the level dependence of the BILD for normal hearing, average results of the six subjects are plotted in Figure 4.2.4 (data in Table 4.2.2). The SRT_{N0S0} data are represented by the open squares, the $SRT_{N0S\pi}$ data by the open triangles and the calculated BILD values by the open diamonds. The bars represent the standard deviation. A broad optimum for the BILD appears to be near the middle of the dynamic range. A BILD of 5.2 dB is found with a standard deviation of 0.4 dB. Similar results are found in the literature (Blauert 1997; Johansson and Arlinger 2000; Wilson et al. 1982). An effect of presentation level is observed especially for the N0Sπ presentation mode. This is in line with results on the dependence of the BMLD on presentation level (Blauert 1997). In earlier research a similar pattern was found for the dependence of precedence effect on presentation level (Goverts et al. 2000). However, in the present data set this trend cannot be confirmed to be statistically significant, mainly due to the large spread in the data for the BILD at 30 dB below the reference level..

- ***Sensitivity pattern of normal hearing for different types of distortion***

As a set of reference data, needed in the distortion sensitivity approach, the normal hearing performance is investigated (data in Tables 4.2.3 and 4.2.4). In the undistorted condition the SRT_{N0S0} ranged from -3.0 to -0.6 dB with a median value of -2.5 dB. Taken into account the bandwidth limitation (250 - 4000 Hz) these values are in line with literature. The $SRT_{N0S\pi}$ varied from -9.6 to -6.0 dB with a median of -7.3 dB. The BILD ranged from 3.8 to 7.4 dB with a median of 5.1 dB.

Median results of the 6 subjects are plotted in Figure 4.2.5 as a function of the degree of distortion of phase information (panel a), frequency information (panel b), time information (panel c), and intensity information (panel d). The SRT_{N0S0} data are represented by the open squares, the $SRT_{N0S\pi}$ data by the open triangles and the calculated BILD values by the open diamonds. The error bars represent the inter-quartile ranges.

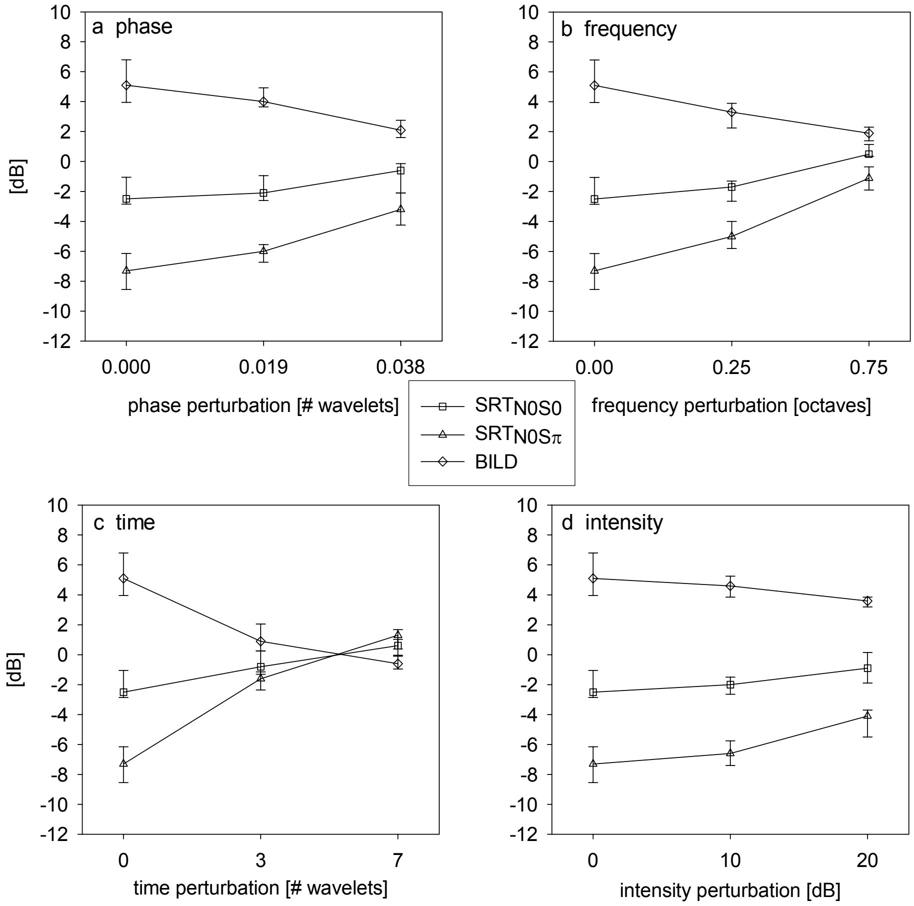


Figure 4.2.5 Median data of six normal-hearing subjects for SRT_{N0S0} (open squares), $SRT_{N0S\pi}$ (open triangles) and median calculated BILD values (open diamonds) for the four types of distortion, as indicated in each panel. Panel a: distortion of phase information; panel b: distortion of frequency information; panel c: distortion of time information; panel d: distortion of intensity information.

The distortion sensitivity approach requires that the type and degree of stimulus perturbations are chosen such that, for normal hearing listeners, the performance on the test is reduced substantially. We are mainly interested in the BILD, and it can be observed that, indeed, the BILD is sensitive to all four types of distortion. The SRT_{N0S0} seems to be sensitive only to distortion in the frequency and time domain. While we are now considering normal hearing only, we assume the calculated distortion sensitivity values, i.e. the slopes determined by the three data points for each condition, to be normally distributed. These slopes can thus be subjected to a T-test, yielding a significant difference ($p<0.05$) from zero (i.e. sensitivity) for the BILD for all four types of distortion, and for SRT_{N0S0} only for frequency and time distortion. Because the BILD is sensitive to distortions in phase domain the observed sensitivity in the domains of frequency and time might be due to induced phase distortion. To distinguish between those effects we recalculated the individual slopes for distortion of phase, frequency and time in dB per de-correlation unit. Comparing the induced phase distortion sensitivity values associated with frequency and time distortion to the actual phase distortion sensitivity values by a paired samples T-test yields no significant difference. Because the SRT N0S0 is not sensitive to phase distortions, induced phase effects are not expected. This means that the BILD is not sensitive to frequency and time distortions, whereas the SRT_{N0S0} is affected by these distortions

- ***BILD of the hearing impaired***

The first observation is that there is no large difference between the BILD of the groups of normal hearing and hearing-impaired subjects, their median values being 5.1 and 4.4 dB respectively (see Table 4.2.4). For the SRT_{N0S0} data the differences are larger, revealing median values of -2.5 and 0.4 dB respectively. Subjecting the data to a Mann-Whitney U test confirms these observations, indicating a significant difference between normal-hearing and hearing-impaired subjects at the $p<0.01$ level for SRT_{N0S0} but no significant difference for BILD.

Aiming to define a subgroup of hearing-impaired subjects with binaural problems, the hearing-impaired subjects were divided into two groups based on their difference from the mean normal hearing BILD-score in the undistorted condition which, as mentioned earlier, are assumed to be normally distributed. To enhance the estimation of mean and standard

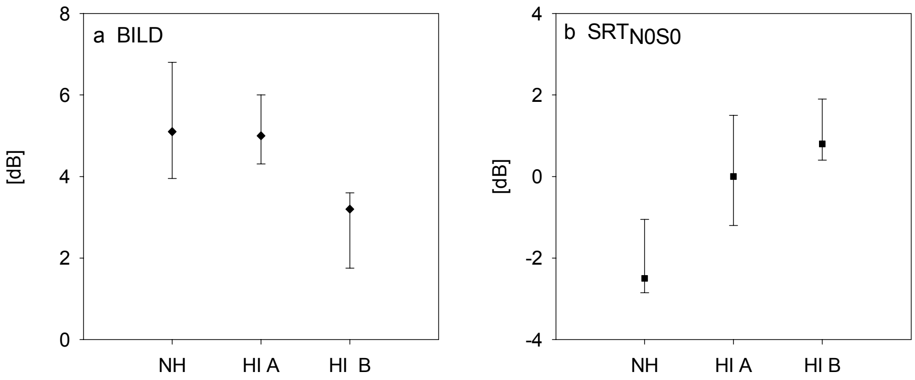


Figure 4.2.6 In panel a median values of the undistorted BILD are given for three groups: normal hearing (NH, N = 6), hearing impaired with normal BILD (HI A, N = 17) and hearing impaired with a significantly reduced BILD (HI B, N = 8). The error bars represent the inter-quartile ranges. In panel b data for the SRT_{N0S0} are presented.

deviation for normal hearing, also SRT measurements using a female voice and female voice shaped noise were added (section III.A). This led to an estimation of the mean BILD of 5.3 dB with a standard deviation of 0.8 dB. Seventeen hearing-impaired subjects with a BILD within a +/- 2 standard deviations interval were included in group A, while the remaining eight subjects were included in group B, labelled as 'significantly reduced BILD'. With regard to mean age and mean audiometric configuration there are essentially no differences between the two groups. In Figure 4.2.6 median data of the undistorted BILD and SRT_{N0S0} are given for the three groups. The error bars represent the inter-quartile ranges. It can be observed that, where group A can't be distinguished from the normal hearing in terms of BILD, both hearing impaired groups A and B have about equal (and reduced) performance in the N0S0 presentation mode.

In conclusion, while optimal audibility is assured the undistorted BILD of the total hearing-impaired group is not significantly different from the normal hearing results. However, when two subgroups of hearing-impaired subjects are defined, eight out of 25 turned out to have a significantly reduced BILD. This is in line with Bronkhorst and Plomp (1990) who designed a free field test of speech perception in noise aiming to separate the effects of several relevant monaural and binaural cues. They compared a group of 18 hearing-impaired listeners with a mild to moderate hearing loss to ten normal hearing subjects. A significantly less performance of speech perception in steady-state noise and a significantly less binaural gain resulting from

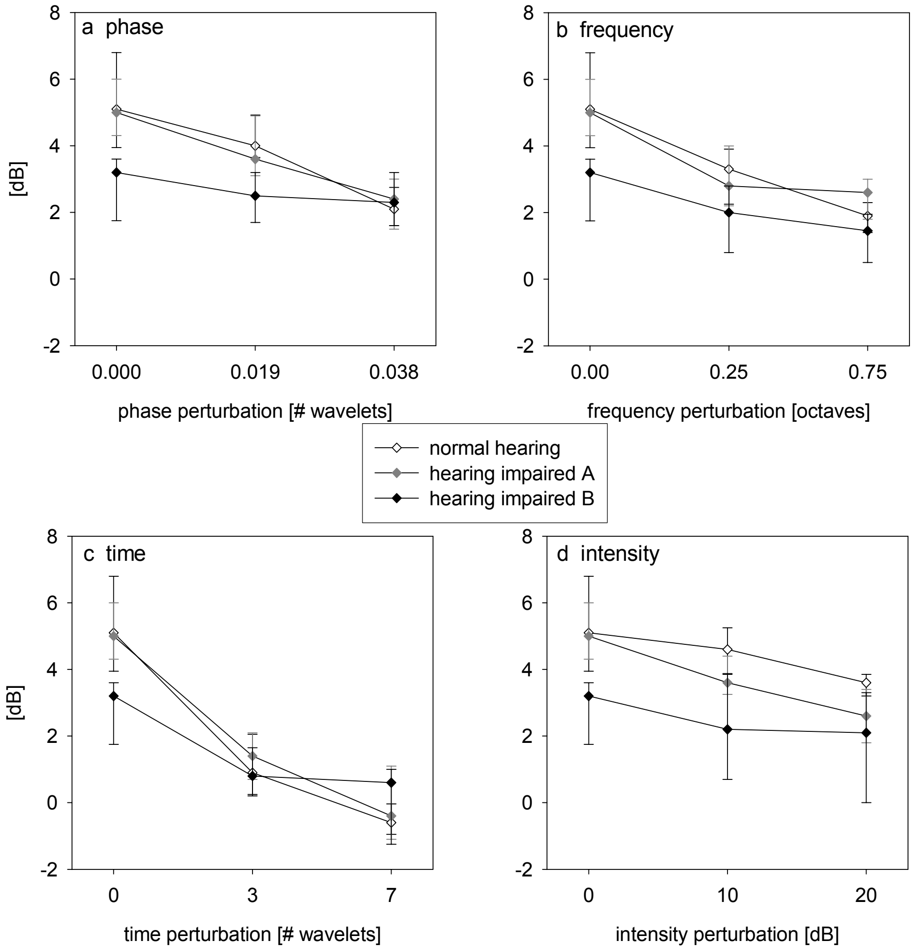


Figure 4.2.7 Median BILD values for the normal-hearing (open diamonds), the hearing-impaired subjects with normal BILD (gray diamonds), and the hearing-impaired subjects with abnormal BILD (black diamonds). The error bars represent the inter-quartile ranges. Panel a: distortion of phase information; panel b: distortion of frequency information; panel c: distortion of time information; panel d: distortion of intensity information.

head shadow was found for the group of hearing-impaired listeners, whereas only few of them had a significantly reduced binaural unmasking.

- ***Sensitivity pattern of hearing-impaired listeners for different types of distortion***The BILD scores are plotted as a function of the degree of distortion in Figure 4.2.7. Data for the two subgroups of hearing impaired are given separately. Data of the hearing impaired group A (with normal BILD) are represented by grey diamonds, while data for the hearing impaired of group B (with reduced BILD) are given by black triangles. The normal hearing data are given by open symbols. The error bars represent the inter-quartile ranges.

The data of the hearing impaired group A are very similar to the normal hearing data. This

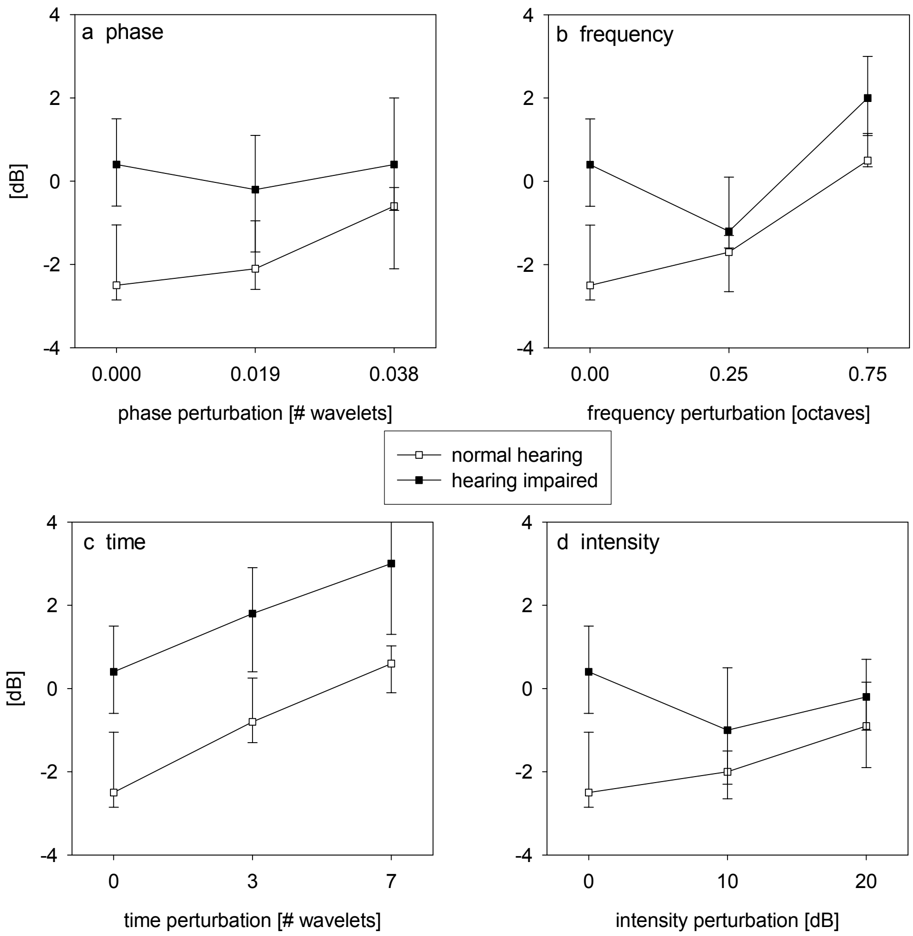


Figure 4.2.8 Median SRT_{N0S0}-data for the normal-hearing (open diamonds), and the hearing-impaired subjects (black diamonds). The error bars represent the inter-quartile ranges. Panel a: distortion of phase information; panel b: distortion of frequency information; panel c: distortion of time information; panel d: distortion of intensity information.

trend is confirmed by Mann-Whitney U Tests on the calculated distortion sensitivity values, yielding no difference between the groups for any of the four domains of phase, frequency, time and intensity. The results of group B tend to converge to the results of the normal-hearing subjects and hearing impaired group A for distortion in the domains of phase, frequency and time. For distortion in the intensity domain no convergence can be observed. These trends in the data are confirmed by Mann-Whitney U Tests on the calculated distortion sensitivity values, yielding a significant difference ($p<0.05$) between normal hearing and hearing impaired group B for the domains of phase, frequency and time, and no difference at all for the domain of intensity. To distinguish between the effects of induced phase and real sensitivity for frequency and time distortions we recalculated the individual slopes for the hearing impaired group B for distortion of phase, frequency and time in dB per de-correlation unit. Comparing the induced phase distortion sensitivity values associated with frequency distortion to the actual phase distortion sensitivity values for the HI group B by a Wilcoxon rank test yields no significant difference. Comparing the induced phase distortion sensitivity values associated with time distortion to the actual phase distortion sensitivity yields a significant difference ($p<0.05$). The effects of frequency perturbation for normal-hearing listeners and the hearing impaired group B can thus be considered as induced phase effects. These results suggest that mainly problems with coding of phase and, to a less extent, coding of time play a role in the reduced BILD.

In answer to the question regarding the sensitivity pattern of monaural (or in our case, diotic) speech reception in noise of the hearing impaired, their SRT_{N0S0} scores are plotted as a function of the degree of distortion in Figure 4.2.8. Hearing impaired data are represented by filled symbols, while normal hearing data are given by open symbols. The error bars represent the inter-quartile ranges.

For distortion in the domains of phase, frequency and intensity the results of the hearing-impaired subjects tend to converge to the normal-hearing subjects group results. For distortion in the time domain no convergence can be observed. These trends in the data are confirmed by Mann-Whitney U Tests on the calculated distortion sensitivities, yielding a significant difference between the hearing impaired group and the normal hearing ($p<0.05$) for the domains of phase, frequency and intensity, and no difference at all for the domain of time. These results suggest that problems with coding of phase, frequency and intensity play a role in the reduced monaural processing of speech in noise. The results are in line with van Schijndel et al. (2001a and 2001b), who found a reduced sensitivity to distortions in the frequency and a normal sensitivity to distortions in the time domain. Their results on the contribution of intensity coding problems were not quite conclusive.

In conclusion, the BILD results show a significantly reduced sensitivity of the eight hearing-impaired subjects with a significantly reduced BILD for distortion in the domains of phase and, to a less extent, time. All groups were equally sensitive to distortion in the intensity domain. The SRT_{N0S0} results show a significantly reduced sensitivity of the hearing-impaired subjects for distortion in the domains of phase, frequency and intensity. Both groups are equally sensitive to distortion in the time domain.

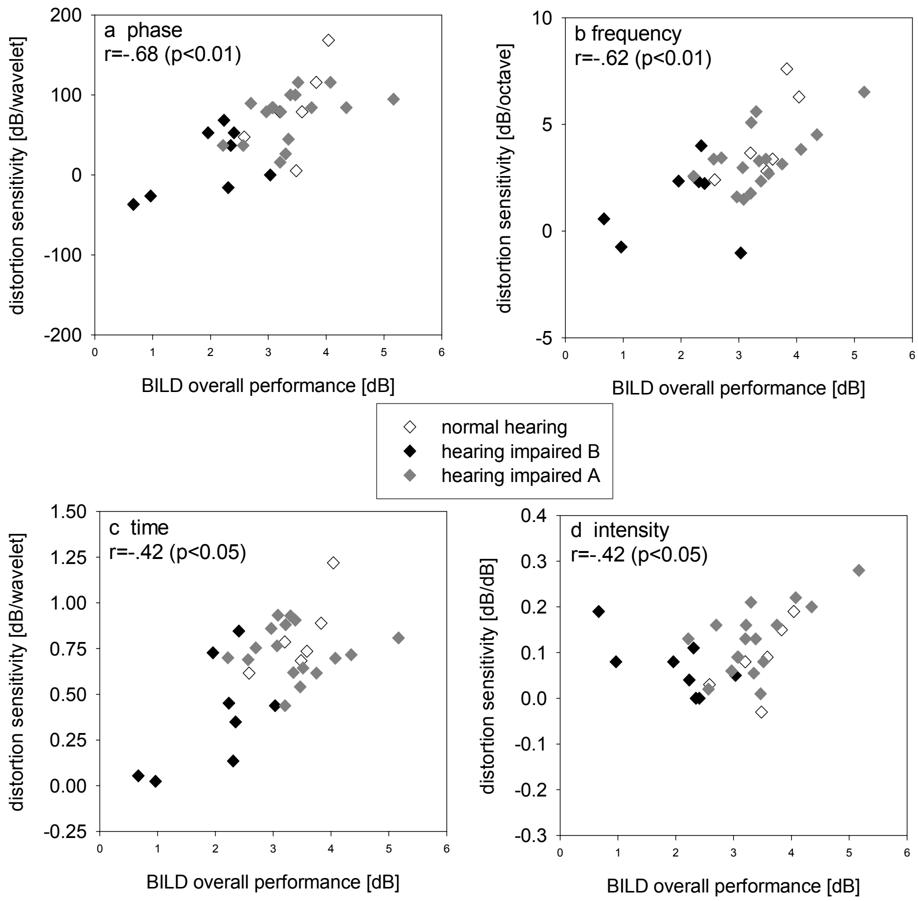


Figure 4.2.9 The individual distortion sensitivities of the BILD plotted versus overall performance on BILD (defined as the average of all data points, see text) for all subjects as well as the Spearman rank correlation coefficient between distortion sensitivity and overall BILD performance. The normal hearing data are represented by open diamonds, data for hearing-impaired subjects with normal BILD by gray diamonds and data for hearing-impaired subjects with abnormal BILD by black diamonds.
Panel a: distortion of phase information; panel b: distortion of frequency information; panel c: distortion of time information; panel d: distortion of intensity information.

- ***Relation between distortion sensitivity and performance for individual subjects***

Up to this point we have considered the differences in distortion sensitivity between *groups* of normal hearing and hearing-impaired subjects. Within the framework of the distortion sensitivity approach, as illustrated by Figure 4.2.1, it is to be expected that, *on an individual basis*, the reduction in distortion sensitivity is larger the poorer the subject's performance on the task. In order to investigate this point scatter plots including all individual subjects on distortion sensitivity versus overall performance, are given for the four domains of distortion for BILD and SRT_{N0S0} in Figures 4.2.9 and 4.2.10, respectively. Theoretically, for this purpose one could plot distortion sensitivity versus the performance in the undistorted condition. However, since that data point is also included in the estimation of distortion

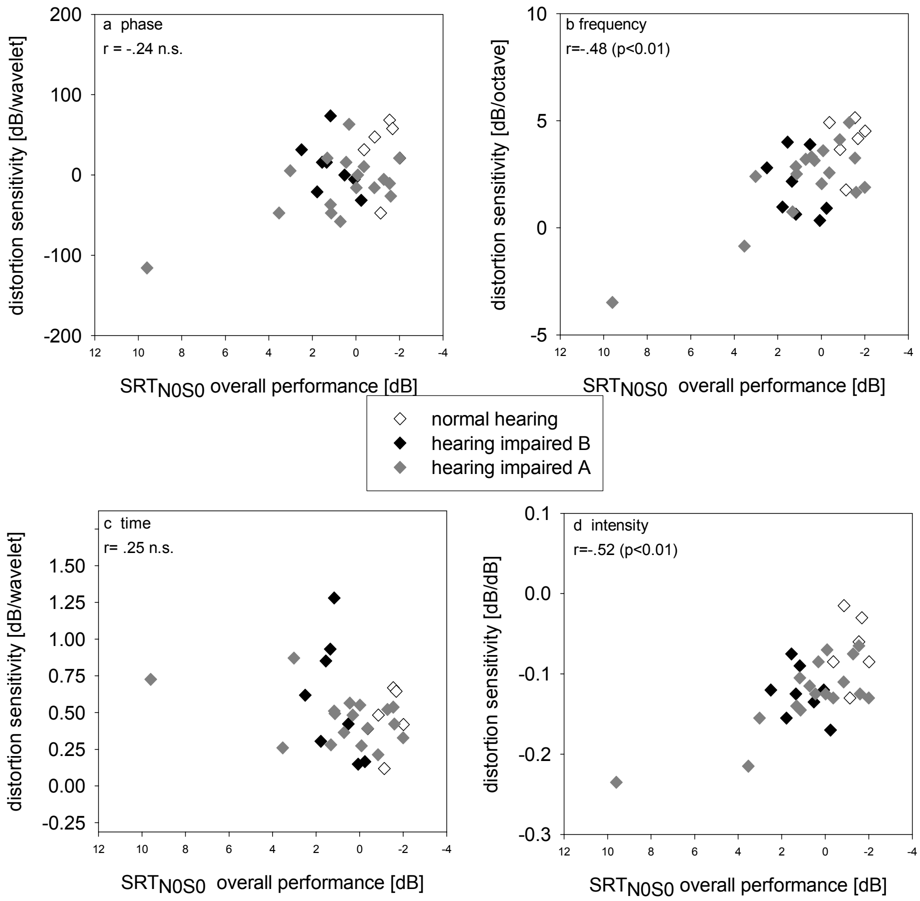


Figure 4.2.10 As Figure 4.3.9, representing data for SRT_{N0S0}

sensitivity (slope by linear fit through three data points), random measurement errors would introduce spurious correlations. Therefore, distortion sensitivity is plotted versus the subject's *overall performance* on the task (either BILD or SRT_{N0S0}). This overall performance is defined as the average of all data points, i.e. the undistorted and the eight distorted conditions (including test and retest data, this amounts to 18 SRT measurements). The normal hearing data are given by open symbols. Data of the hearing-impaired group A, with normal BILD are represented by grey diamonds, while data for the hearing impaired of group B, with reduced BILD are given by black triangles.

For all individual subjects the BILD performance relates mainly to phase and frequency distortion sensitivity. The importance of interaural phase-relations for binaural unmasking is evident. To distinguish between the effects of induced phase and real sensitivity for frequency and time distortions we recalculated the individual slopes for the total group of NH and HI in dB per de-correlation unit. Comparing the induced phase distortion sensitivity values associated with frequency distortion to the actual phase distortion sensitivity values by a

Wilcoxon rank test yields no significant difference. Comparing the induced phase distortion sensitivity values associated with time distortion to the actual phase distortion sensitivity values by a Wilcoxon rank test yields a significant difference ($p<0.01$). Thus, the role of frequency coding we found for the BILD, can be explained by induced phase effects.

For all individual subjects the SRT_{N0S0} performance relates to frequency and intensity distortion sensitivity. The relation between reduced frequency selectivity and monaural speech reception in noise has been found in several studies (see van Schijndel et al., 2001b for a discussion). The effect of inherently distorting phase by distorting frequency cues does not seem to play a role here, as no significant relation is observed between overall performance and phase distortion sensitivity.

- ***Relation between binaural unmasking and monaural speech reception***

If the processes underlying binaural unmasking (BILD) and monaural speech reception in noise (SRT_{N0S0}) would make use of the same stimulus cues, and would also require the same degree of auditory coding accuracy of these cues, one would expect a high correlation between BILD and SRT_{N0S0} among subjects. This would hold for the data in the undistorted condition, and even more so for the eight conditions of stimulus distortion. In Figure 4.2.11 BILD data are plotted versus SRT_{N0S0} results, using different symbols for normal hearing and

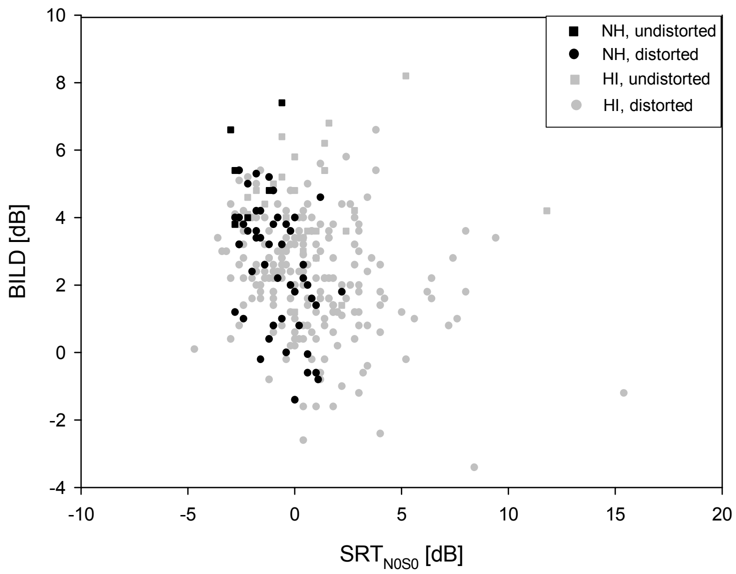


Figure 4.2.11 BILD values versus SRT_{N0S0} data for six normal-hearing subjects (black symbols) and 25 hearing-impaired subjects (gray symbols) for the undistorted condition (squares) and the eight distorted conditions (circles), amounting to a total of 31*9 = 279 data points.

hearing impaired, and also for distorted and undistorted data. Table 4.2.6 presents rank correlation coefficients for the normal hearing (6*9 = 54 data points) and for the total group (31*9 = 279 data points).[2] These correlations can be attenuated by a limited accuracy of the tests. If this inaccuracy is caused by measurement error, the true correlation (r_{true}) between two measures, m1 and m2, can be estimated by

$$r_{true}^{2} = r_{observed}^{2}/(\alpha_{m1} * \alpha_{m2}) \qquad (1)$$

where α_{m1} and α_{m2} represent the accuracy of m1 and m2 expressed, for example by the Cronbach alpha (e.g. Nunnally, 1967; Guilford, 1954). It should be noted that the thus established value for r_{true} is only an estimation with a confidence interval, which depends on the number of observations, the value of α_{m1} and α_{m2} and the actual value of $r_{observed}$.. Values for the thus established true correlation are also given in Table 4.2.6.

For the normal-hearing subjects, when corrected for attenuation, BILD and SRT_{N0S0} are substantially correlated, indicating that in monaural speech reception in noise partly the same cues are utilized as in binaural unmasking. This is in line with the result of our study on the relation between BMLD and BILD for normal hearing, where a similar relation was found between BMLD and SRT_{N0S0} (Goverts and Houtgast, 2004).

Table 4.2.6 Spearman rank correlation coefficients between BILD values and SRT_{N0S0} data in the undistorted condition and the eight distorted conditions for six normal-hearing subjects and for the total group of six normal hearing and 25 hearing-impaired subjects. Next to the observed correlations an estimate of the true correlations is given, taking into account the attenuation of correlation caused by limited accuracy of the tests. All observed correlations are significant at the p<0.01 level.

correlation	diotic (SRT_{N0S0}) and unmasking (BILD)	
	$r_{observed}$	r_{true}
normal hearing (N = 6)	-0.49	-0.82
all subjects (N = 31)	-0.28	-0.40

[2]While the BILD is mathematically derived from both SRT_{N0S0} and $SRT_{N0S\pi}$ data, simply correlating the BILD to SRT_{N0S0} measures could lead to some spurious correlation. To check for this unwanted correlation, the BILD data derived from the test data for SRT_{N0S0} and $SRT_{N0S\pi}$ are correlated to the retest results of SRT_{N0S0} and vice versa. This yielded essentially the same correlations.

Table 4.2.7 Summary of the group results and individual results of the distortion sensitivity approach for the BILD and the SRT_{N0S0} for distortion in the domains of phase, frequency, time and intensity. Convergence in the group results for distortion of a particular cue suggests that the reduced performance on this test is indeed related to a suprathreshold coding deficit of that cue (see Figure 4.2.1). A significant correlation between individual sensitivities to distortions of a particular cue and the performance indicates that the reduced performance is determined by the extent of that suprathreshold deficit. Correlations with significance at the $p<0.05$ and $p<0.01$ level are marked by * and ** respectively.

	BILD		**SRT_{N0S0}**	
	group results	**individual results**	**group results**	**individual results**
subjects	HI B versus NH	all	all HI versus NH	all
	convergence	correlation performance vs distortion sensitivity	convergence	correlation performance vs distortion sensitivity
cue				
phase	yes	-.68**	yes	-.24
frequency	yes	-.62**	yes	-.48**
time	yes	-.42*	no	-.25
intensity	no	-.42*	yes	-.52**

For the total group of subjects, even when corrected for attenuation, a low correlation is found, indicating that the accuracy in coding of the four types of suprathreshold cues is affected in various different ways for the current population of sensorineural hearing-impaired subjects. This can also be concluded from a comparison of the group results and individual results of the distortion sensitivity approach for BILD and SRT_{N0S0}. An overview of these data is given in Table 4.2.7, suggesting that the distortion sensitivity patterns for BILD and SRT_{N0S0} are not similar. The reduced BILD is mainly determined by deficits in coding of phase and frequency. The BILD performance seems to be mainly related to deficits affecting interaural phase relations, where SRT_{N0S0} performance is mainly related to deficits affecting frequency selectivity. Types and extent of deficits vary over hearing-impaired subjects.

In conclusion, the mechanisms underlying binaural unmasking and monaural speech reception in noise utilize partly the same suprathreshold cues. As the suprathreshold coding problems associated with a given sensorineural hearing loss can vary in type and extent of deficits, the effects of such a hearing loss on these mechanisms can be manifold.

V SUMMARY AND CONCLUSIONS

The central theme of this study was the role of suprathreshold coding deficits in binaural unmasking (BILD) and in monaural speech reception in noise (estimated by SRT_{N0S0}). Investigating the relation between BILD and presentation level for normal-hearing subjects, an optimum in the middle of the dynamic range was found. In order to correct properly for audibility effects, all subsequent speech intelligibility measurements were performed in the middle of the individual's dynamic range.

As a set of reference, required for the distortion sensitivity model, normal-hearing data on distortion sensitivity were collected. These data reveal significant sensitivity of the BILD for all four types of distortions considered. However the induced phase effects of frequency and time perturbations were not significantly different from the effects of phase perturbations. Thus, for normal hearing listeners, binaural unmasking is not affected by frequency and time distortions *per se*, which do affect monaural speech processing in noise. On the other hand, binaural unmasking is affected by phase distortions that do not affect speech processing in noise. For the SRT_{N0S0} only sensitivity for distortions in the domains of frequency and time was found, in line with van Schijndel (2001).

While optimal audibility is assured, the undistorted BILD of the hearing-impaired group is not significantly different from the normal hearing results. However, the hearing impaired can be divided unto two subgroups, group A (N = 17) with normal BILD and group B (N = 8) with a significantly reduced BILD (i.e. more than 2 standard deviations less than normal-hearing).This is in line with earlier findings of Bronkhorst and Plomp (1990).

The data suggest that the reduced BILD of the eight subjects of group B is mainly caused by deficits in coding of phase and, to a less extent, of time. For the individual subjects the BILD performance relates mainly to phase distortion sensitivity. The reduced SRT_{N0S0} of the total group of hearing-impaired subjects (N = 25) appears to be caused by deficits in coding of phase, frequency and intensity. For the individual subjects the SRT_{N0S0} performance relates to frequency and intensity distortion sensitivity.

For the normal-hearing subjects, BILD and SRT_{N0S0} are substantially correlated, indicating that in monaural speech reception in noise partly the same cues are utilized as in binaural unmasking. This is in line with the result of our study on the relation between BMLD and

BILD for normal hearing, where a similar relation was found between BMLD and SRT_{N0S0} (Goverts and Houtgast, 2004a). For the total group of subjects a low correlation between BILD and SRT_{N0S0} is found, indicating that the accuracy in coding of the four types of suprathreshold cues is affected in various different ways for the current population of sensorineural hearing-impaired subjects.

The mechanisms underlying binaural unmasking and monaural speech reception in noise utilize partly the same suprathreshold cues. The deficits in coding of phase and time, causing the reduced BILD seem to occur independently of the deficits in coding of frequency and intensity and less often. As the suprathreshold coding problems associated with a given sensorineural hearing loss can vary in type and extent of deficits, the effects of such a hearing loss on these mechanisms can be manifold.

ACKNOWLEDGEMENTS

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APPENDIX A INTRODUCTION OF SIGNAL DEGRADATION

A wavelet decomposition scheme was used to introduce perturbations in the domains of phase, frequency, time or intensity in the speech and noise material. The main advantage of the wavelet decomposition is its resemblance to auditory processing because of its proportional spectro-temporal analysis window based on a 'mother wavelet'. A brief overview of the approach will be given below. An extensive overview on wavelet analysis is given by Rioul and Vetterli, 1991; Vetterli and Kovacevic, 1995; Strang and Nguyen, 1996.

Details and examples on the application of this analysis tool in auditory research are given by van Schijndel et al. (2000, 2001a, 2001b), investigating the influence of distorted auditory coding on monaural speech perception.

In accordance with van Schijndel et al. a Gaussian-windowed sinusoid was chosen as mother wavelet:

$$s(t) = \sqrt{\alpha f_0} \exp(i2\pi f_0 t) \exp(-\pi(\alpha f_0 t)^2)$$

in which f_0 is the carrier frequency, α the so called *shape factor* and $\sqrt{\alpha f_0}$ normalizes the energy of the analysis function. The effective bandwidth of this time-frequency window is given by αf_0 and its effective duration by $1/(\alpha f_0)$. Van Schijndel et al. chose a shape factor of 0.1735, resulting in an effective bandwidth of 1/4 octave, which roughly corresponds to the auditory critical bandwidth (e.g. Scharf, 1970; Moore, 1995).

A basis for signal decomposition is constructed with scaled versions of this mother wavelet. As a basis for wavelet analysis should be orthonormal, wavelets ought to have a compact support. Because this is not the case for such a Gaussian wavelet the range between the points that are 25 dB down the peak is taken as the significant range of the wavelet. Van Schijndel has demonstrated that for adequate sampling in frequency and time a basis consisting of eight wavelets per octave along the spectral axis and one wavelet per three periods of the wavelet carrier frequency is needed. All the signals were bandwidth limited between 250 and 4000 Hz. This results in 33 wavelets along the spectral axis and, depending on the carrier frequency, between 83 and 1333 wavelets per second along the temporal axis. Thus, a speech signal of one second can after decomposition on this base be described by about $16 \cdot 10^{3}$ wavelet coefficients.

Degradation of stimuli

The process of degradation of signal and noise is essentially as follows:

1 Selection of sentence file and speech shaped noise file.

2 Decomposition of both sentence and noise on the above described base, resulting in a *coefficient file.*

3 Statistical perturbation of all wavelets constituting the base in one of the four domains (phase, frequency, time or intensity) by a random factor *R* . This factor is chosen from a

uniform distribution characterized by a perturbation value (PV), determining the ranges from plus PV/2 to minus PV/2. The perturbation process results in a *perturbed base for the signal and noise to be presented to the left ear*. Details about the perturbation process are given below.

4 Recomposition of sentence and noise file with coefficients from the *coefficient file* based on the *perturbed base* for the left ear stimuli and normalization of energy of the stimuli.

5 A similar independent processing as described in steps 4 and 5 for the right ear stimuli.

Perturbation in four domains

Perturbations in the phase domain were introduced by a random temporal displacement of each wavelet as a whole, both envelope and fine structure. The perturbation values (PV) were 0.019 and 0.038 wavelet, so the maximal displacement was plus or minus 0.0095 and 0.019 wavelets, corresponding to 0.055π and 0.11π , respectively.

Perturbations in the frequency domain were introduced by shifting the carrier frequency of the wavelet, thus shifting it up or down along the spectral axis. The perturbation values (PV) were 0.25 and 0.75 octave, so the maximal displacement was plus or minus 0.13 and 0.38 octave, respectively.

Perturbations in the time domain were introduced by shifting the temporal position of the envelope, while keeping the fine structure constant by extrapolation. The perturbation values (PV) were 3 and 7 wavelets, so the maximal displacement was plus or minus 1.5 and 3.5 times the effective wavelet duration, equalling to about 10 and 20 local periods, respectively.

Perturbations in the intensity domain were introduced by multiplying each wavelet coefficient by a random factor. The perturbation values (PV) were 10 and 20 dB, so the maximal perturbation yielded plus or minus 5 and 10 dB, respectively. An overview of perturbation values and corresponding measures is given in Table 4.2.A1.

In interpreting the results, it should be noted that these four types of perturbations are not really completely mutually independent. For instance, the frequency-domain perturbations will also affect the momentary inter-aural phase relations, and so will the time-domain perturbations. We could not find a way to completely avoid such interactions among the various types of perturbations.

Stimulus presentation during SRT measurement

During the SRT measurements the speech and noise signals that were degraded in accordance with the method described above were added in the appropriate signal-to-noise ratio. The main reason for this procedure, in stead of preparing al stimuli in all possible signal-to-noise ratios, was practically driven by considerations of calculation time and storage capacity. In N0Sπ conditions a phase shift was introduced.

Table 4.2.A1 Overview of the amount of perturbation that is introduced in the four domains expressed by the value of PV. Each wavelet is shifted by a factor R, randomly chosen from an uniform distribution ranging from -PV/2 to PV/2.

domain	PV I	maximal perturbation	PV II	maximal perturbation
phase	0.019 wavelet	0.055π	0.038 wavelet	0.11π
frequency	0.25 octave	0.13 octave	0.75 octave	0.38 octave
time	3 wavelets	10 local periods	7 wavelets	20 local periods
intensity	10 dB	5 dB	20 dB	10 dB

5 General discussion and conclusion

I OVERALL ASSESSMENT OF SPATIAL HEARING

The first part of this thesis deals with the relation between psychophysical and subjective overall assessment of spatial hearing. In chapter 2 an optimal relation between psychophysical and subjective assessment of sound localization was sought, taking self-assessment by means of the Amsterdam Inventory of Auditory Disability and Handicap as a gold standard. For a homogenous subgroup a reasonable correlation of 0.76 was found, choosing a design with everyday sound signal and masker in a 0 dB signal-to-noise ratio and using an optimized score-measure based on false-response patterns (dispersions and front-back confusions). This result indicates that a fair correspondence of psychophysical data on *overall assessment* of activity limitation (or disability) in spatial hearing to results of self-assessment can be achieved.

II THE ROLE OF AUDIBILITY IN SOUND LOCALIZATION AND PRECEDENCE EFFECT

In chapter 2, the correlation approach was used to investigate the role of audibility in *sound localization*. Subjective localization data are significantly correlated ($r = 0.56$) with the mean hearing thresholds over 500, 1000 and 2000 Hz for both ears. Taking the mean hearing threshold for only the poorer ear yielded a slight improvement of the correlation with subjective score, while taking the better ear weakened the correlation to 0.35. The psychophysical localization data, based on percentage correct responses, are significantly correlated ($r = -0.59$) with the mean hearing thresholds for both ears. Taking the mean hearing threshold for the poorer ear raised the correlation with the psychophysical score to -0.74. The mean threshold of the better ear appears to be uncorrelated to the psychophysical score. Despite these correlations between audiometric measures and both psychophysical and self-assessed data on sound localization, still a *broad range of scores* is observed for a given degree of hearing loss. A reason for the limited relation between audiometric data and psychophysical test-results might be suprathreshold or central problems, which are not reflected in pure tone audiometry.

In chapter 3 the role of audibility in the *precedence effect* was investigated, using the level dependency approach. A relatively new headphone-based paradigm was used, ensuring that the measurements are not biased by motivation, experimental skills, and subjective aspects of judgement. This design also enables differentiation between lateralization accuracy and precedence effect. For normal hearing, a decrease in precedence effect was found for sensation levels below about 30-40 dB. In daily-life conditions with a sensation level in noise of typically 10 dB the precedence effect is still manifest, albeit degraded to some extent. For six sensorineurally hearing-impaired subjects a reduced precedence effect for lateralization was found, as well as a reduced lateralization accuracy. Typically, the mean performance of the hearing-impaired subjects in quiet is similar to normal hearing performance in masking noise at a signal-to-noise ratio of 0 dB. There is a broad range of individual result patterns in terms of strength of the precedence effect and lateralization accuracy. These results cannot be explained on basis of reduced audibility alone. Thus, the reduced binaural performance of hearing-impaired subjects, as reflected by precedence effect data, is only partly caused by reduced audibility.

The *general conclusion* is that, although reduced audibility causes reduced binaural and spatial hearing, reduced performance of the hearing impaired cannot fully be explained by the audibility hypothesis.

III THE ROLE OF SUPRATHRESHOLD CODING IN BINAURAL UNMASKING AND MONAURAL SPEECH RECEPTION

For the investigation of the role of suprathreshold deficits in binaural hearing we chose the BILD because it can be measured in a clinically friendly way. In section 4.1, it was found that binaural unmasking in speech intelligibility (the BILD) is closely related to binaural unmasking in speech detection (the BMLD) for normal hearing. Levitt and Rabiner (1967b) explained the absolute difference between BMLD and BILD as a consequence of the frequency dependence of binaural unmasking and the frequency importance fubnction in relaion to speech intelligibility, which can be considered the domain of audibility. In this thesis we found that BMLD and BILD are similarly affected by suprathreshold perturbations using the perturbation effect approach. This means that BILD performance can be essentially considered as binaural unmasking, measured by means of speech intelligibility.
In section 4.2 the role of suprathreshold coding in the BILD was investigated using the distortion sensitivity approach for a group of mild to moderate sensorineurally hearing impaired. The effect of audibility was ruled out by presenting stimuli in the middle of the subjects dynamic range, so reduced BILD performance should be caused by suprathreshold coding deficits affecting the binaural processing, neglecting the central processing component (see Figure 1.3). The role of suprathreshold coding in binaural unmasking was compared to its role in monaural speech reception in noise as operationalized by the SRT_{N0S0} stimuli used to establish the BILD. As these stimuli contain no binaural information, we consider them as essentially monaural (Siegel and Colburn, 1983; Durlach 2003, personal communication). Normal hearing data were gathered as a set of reference for the distortion sensitivity approach. This data set shows how the cues of phase, frequency, time and intensity are used in binaural unmasking and monaural speech intelligibility. For normal-hearing listeners, binaural unmasking is affected by phase distortions, which do not affect monaural speech processing in noise. On the other hand, binaural unmasking is not affected by frequency and time distortions, which do affect monaural speech processing in noise. For the SRT_{N0S0} only sensitivity for distortions in the domains of frequency and time was found, in line with van Schijndel (2001).

Remarkably, while optimal audibility is assured the BILD in the undistorted conditions of the hearing-impaired group is not significantly different from the normal hearing results. However, the hearing impaired can be divided unto two subgroups, group A (N = 17) with normal BILD and group B (N = 8) with a significantly reduced BILD (i.e. more than 2 standard deviations less than normal-hearing). Thus, for the present sub population of hearing impaired we conclude that, when restoring reduced audibility, only about 30% have reduced binaural performance. Of course these results cannot simply be generalized, mainly because of the limited number of patients and relatively mild hearing losses.

The distortion sensitivity data suggest that the reduced BILD of these eight subjects of group B is mainly caused by deficits in coding of phase and, to a less extent of time. For the individual subjects the BILD performance relates mainly to phase distortion sensitivity. The reduced SRT_{N0S0} of the total group of hearing-impaired subjects (N = 25), appears to be related to deficits in coding of phase, frequency and intensity. For the individual subjects the SRT_{N0S0} performance relates to frequency and intensity distortion sensitivity. The deficits in coding of phase and time, causing the reduced BILD seem to occur independently of the deficits in coding of frequency and intensity and less often.

The mechanisms underlying monaural speech reception in noise and binaural unmasking utilize partly the same suprathreshold cues. As the suprathreshold coding problems associated with a given sensorineural hearing loss can vary in type and extent of deficits, the effect of such a hearing loss on these mechanisms can be manifold.

The distortion sensitivity approach turns out to be a useful research tool for investigating the suprathreshold deficits that might account for the problems in binaural unmasking and in monaural speech reception in noise as well. For this approach a we made a choice of a set of suprathreshold cues to be perturbed, i.e phase, frequency, time and intensity. These four types of perturbations are not really completely mutually independent. Especially, the frequency-domain perturbations will also affect the momentary inter-aural phase relations, and so will the time-domain perturbations. The effects of real distortion in the domains of frequency and time can be separated from the effect of unwantedly induced phase distortions by comparing results in conditions with equal interaural de-correlation.

This set of cues turns out to be sufficiently broad to cover the range of suprathreshold problems that cause reduced monaural and binaural processing. If this were not the case, no convergence would have been found in any of the four domains for binaural unmasking and monaural speech intelligibility. While this study seems to indicate different *patterns* of suprathreshold deficits for the subgroups, rather than groups with only a reduced sensitivity to *one specific cue*, one might question whether the set of cues is maybe not appropriate to identify specific cochlear pathology unambiguously. Inspection of the individual distortion sensitivity data for the eight subjects with reduced BILD yields patterns of deficits for each of them, suggesting no direct relationship between the four cues and suprathreshold (e.g. cochlear) pathology in this sub population.

A second, more fundamental question is whether the suprathreshold problems that were identified using the distortion sensitivity approach are localized in the cochlea, the binaural processing system (or in the central system). The present results can be brought in line with cochlear pathology having a diverse effect on monaural and binaural processing, but also with a combined cochlear and binaural-processing pathology. This problem might be solved by combining the distortion sensitivity approach with both monaural and binaural psychophysical tests. Another possibility is to include patients for which the physiological localization of pathology is known, for example with certain types of congenital hearing loss, or known binaural-processing pathology. This would require a substantial amount of time of the subjects, as well as efficient planning of research.

So far, we have considered the data in an attempt to enhance our insight in the role of reduced audibility and suprathreshold coding in binaural processing of the hearing impaired. One can also look at the data from the point of view of *rehabilitation*. Techniques to enhance the representation of cues in the domains of phase, frequency, time and intensity monaurally (e.g. Lyzenga et al., 2002) or even binaurally (e.g. Kollmeier, 1997) are coming available in hearing aids. Despite the fundamental questions addressed above, the distortion sensitivity approach would be a very useful *pragmatic tool* to assess candidacy for an individual patient. For that purpose looking at distortion sensitivity of the dichotic $SRT_{N0S\pi}$ might be preferred, as the performance on that task is most typical for the daily process of combining binaural unmasking and speech intelligibility in noise.

IV MAIN CONCLUSIONS

Resuming the preceding sections of this chapter, the general conclusions of this thesis are listed below. In chapter 6, the implications for clinical practice will be discussed.

- It is possible to define a listening test to assess overall sound localization performance, in reasonable agreement with self-assessment scores.
- Reduced audibility causes reduced spatial and binaural hearing, but the audibility hypothesis cannot fully account for the total amount of performance reduction observed for the hearing impaired.
- In sound localization the hearing loss of the poorer ear is of major importance.
- The non-classic headphone-based paradigm used in this thesis, is an appropriate research tool for investigating the precedence effect for normal hearing and hearing-impaired subjects.
- Binaural unmasking in speech intelligibility (BILD) reflects essentially the same auditory processing as in speech detection (BMLD).
- For normal-hearing listeners, binaural unmasking is affected by phase distortions, which do not affect monaural speech processing in noise. On the other hand, binaural unmasking is not affected by frequency and time distortions, which do affect monaural speech processing in noise.
- In a group of 25 mild to moderate hearing-impaired subjects, despite restoring audibility, eight subjects show reduced binaural unmasking. The distortion sensitivity data suggest that this is mainly caused by sub-optimal auditory coding in the domain of phase and, to a less extent, of time. These deficits seem to occur independently of the deficits in coding of frequency and intensity and less often.
- The effect of hearing impairment on binaural unmasking and monaural speech intelligibility in noise and can be diverse.
- The distortion sensitivity approach is a valuable research tool in specific assessment of spatial and binaural hearing both on a group level and for the individual hearing impaired. Some remaining questions on the precise interpretation of the data would require extensive additional experiments.

6 Clinical implications

In this chapter implications and relevance of the results of this thesis for clinical practice are discussed. Not all of the preceding can be directly applied in everyday audiological care. Not in every clinic, for example, facilities for psychophysical assessment of spatial and binaural hearing are available. However, in my opinion the results as summarized and discussed in chapter 5 have relevance for all professionals who are involved in diagnosis and rehabilitation of hearing impaired, e.g. otolaryngologists, audiologists, speech and hearing therapists, hearing aid dispensers, general practitioners, company doctors. This will be illustrated in the next sections. Finally, some consequences for the political and social debate on auditory rehabilitation will be indicated.

I IMPLICATIONS FOR DIAGNOSIS OF THE AUDITORY FUNCTION

The main implication for the diagnosis of the auditory function is the overwhelming diversity of problems that can be associated with one clinical pure-tone audiogram. To illustrate this, the raw data of the patients with a symmetrical mild to moderate sensorineural hearing loss (mean threshold at 500, 1000 and 2000 Hz not exceeding 60 dB) participating in the studies in chapters 2, 3 and 4 are revisited. The range of results on several tests for those patients are summarized below.

- The self-assessed localization performance score can range from 0 to 2.8 (on a scale from 0 to 3), which means form normal values to very deviant. The percentage correct localization score can range from about 90 to 10%, which means from normal to very deviant values (see Figure 2.6)
- The precedence effect can vary from normal ranges to total absence. The lateralization accuracy can vary from normal range to almost chance.
- The SRT_{N0S0} with optimized audibility can vary from -2.2 to +11.8 dB, which means from normal values to very deviant (far more than two standard deviations).
- The BILD with optimized audibility can vary from +8.2 to +1.2 dB which means from normal to deviant.
- The sensitivity of diotic (monaural) speech intelligibility to distortions in the domains of phase, frequency, time and intensity can vary from normal to deviant (more than 1 standard deviation for time and more than 2 standard deviations for the other domains).
- The sensitivity of binaural unmasking to distortions in the domains of phase, frequency, time and intensity can vary form normal to deviant (more than 1 standard deviation for intensity and more than 2 standard deviations for the other domains).

This survey implies that there may be a large variety in deficits associated with the pure tone audiogram, resulting in a range of activity limitation. Even for professionals working in a setting without facilities for assessment other than pure-tone audiometry, it is important to be aware of this diversity.

This thesis has underlined that a discrepance between the results of audiometry and any form of subjective assessment, be it by a validated instrument or by 'simple' anamnesis, can be caused at several stages in the auditory system. Reduced audibility causes reduced spatial and binaural hearing, but the audibility hypothesis cannot account for all of the reduced performance of the hearing impaired. The reduced performance of hearing impaired that is not accounted for by audibility, can be identified by the reduced sensitivity to distortions in the domains of phase, frequency, time and intensity.

For professionals, aiming to include some psychophysical assessment of binaural processing, measurements of binaural unmasking can be a good choice, while this can be evaluated by the clinically friendly BILD as shown in chapter 4.

Whatever binaural measurement is chosen, the use of audibility optimization is recommended, in order to avoid to identify 'binaural' problems that are in fact caused by reduced audibility. In a group of 25 mild to moderate hearing-impaired subjects, despite restoring audibility, all show a reduced monaural speech intelligibility while only eight show reduced binaural unmasking.

For an individual patient the distortion sensitivity approach can contribute to the diagnosis of suprathreshold deficits underlying the reduced performance. The sensitivity of the $SRT_{N0S\pi}$ to distortions in the different domains indicates how the integral binaural processing of dichotic speech stimuli might be improved by restoring specific auditory cues, if the latter becomes possible.

II MODULAR PROTOCOL FOR THE ASSESSMENT OF SPATIAL AND BINAURAL HEARING

The results of this thesis can be applied to clinical practice by composing a protocol for the assessment of spatial and binaural hearing consisting of several modules (see Figure 6.1). The idea behind the identification of different modules is that the elaborateness of assessment should be chosen depending on context and earlier findings. Distinction of modules also provides insight in the way how professionals and institutions with a different extent of facilities can adequately cooperate in audiological care. In many clinical settings the assessment of hearing impairment will be restricted to module A.

Extension of the anamnesis in this module with self-assessment requires relatively minor effort. In this thesis the AIADH is used (Kramer et al. 1995). A survey of other instruments is given by Bentler and Kramer (2000). Recently Gatehouse and Noble released their Speech Spatial and Quality of hearing scale (Gatehouse and Noble, 2004). By using self-assessment in combination with module A, subjects with (unexpectedly) high activity limitations can be identified. This should serve as an indication for additional care.

A Assessment of impairment

Anamnesis
ENT examination
Basic audiometry

B Overall assessment of activity limitation

Subjective assessment
•AIADH

Psychophysical testing
•Sound localization
•*Spatial speech perception (e.g. Boymans et al. 2003, Bronkhorst and Plomp, 1990)*

C Specific assessment: audibility, monaural speech reception and binaural unmasking

•SRT and SRT with optimized audibility
•BILD and BILD with optimized audibility

D Specific assessment: suprathreshold coding in monaural speech reception, and binaural unmasking

•Distortion sensitivity SRT with optimized audibility
•Distortion sensitivity BILD with optimized audibility
•*Other monaural and binaural psychophysical tests e.g. on spectral and temporal resolution*

E Specific assessment central processing

•*Central auditory processing (e.g. testbattery, Neijenhuis, 2003)*
•*Mental effort (e.g. pupil dilatation, Kramer et al.,1997)*
•*Cognitive function*

Figure 6.1 modular protocol for clinical assessment of patients with complaints of reduced spatial hearing

At least in medico-legal cases, but also in cases regarding the auditory function and employment, subjective overall assessment should be accompanied by psychophysical assessment. This requires more extended hardware facilities, i.e. sound card, speakers and measurement room. In chapter 2 a sufficiently valid psychophysical test for sound localization is described. An example of a test-design for the assesment of sptial speech perception requiring only a simple hardware setup is given in Boymans et al. (2003). In this design the speech reception threshold is measured for sentences played through two loudspeakers at plus and minus 45 degrees. Time reversed speech presented from the opposite direction is used as a masker. Bronkhorst and Plomp (1990) describe a test using four loudspeakers, which is designed to separate the effects of serveral relevant monaural and binaural cues for speech perception in noise.

A reason to extend the assessment protocol to module C, is to identify the quality of monaural speech perception in noise and binaural unmasking as well as the role of reduced audibility. In this thesis the assessment of SRT_{N0S0} and $SRT_{N0S\pi}$ was restricted to measurements with optimized audibility. Thus, only the BILD with optimal audibility is available (obtained by subtracting SRT_{N0S0} and $SRT_{N0S\pi}$).The SRT_{N0S0} data are used as an estimation of monaural speech intelligibility. While audibility was optimized, I assume that reduced performance is caused by suprathreshold or binaural-processing deficits. If data without audibility optimization for SRT_{N0S0}, $SRT_{N0S\pi}$ and BILD were available, comparing them to data with audibility optimization would yield the amount of performance loss caused by audibility problems. BILD measurements require a rather basic psychophysical hardware setup, however for audibility optimization a more elaborate procedure is needed.

If monaural speech reception in noise and/or binaural unmasking are reduced even while audibility is optimized, using module D one can try to differentiate between possible types of suprathreshold deficits. This module is much more time consuming and requires more facilities.

Finally, in module E not all elements are yet fully specified. The central processing was not considered in this thesis. Recently a test-battery for central auditory processing disorders has been released (see Neijenhuis, 2003). Kramer et al.(1997) found that the process of sound perception is very energy-consuming, especially for hearing impaired. They suggested the use of a pupil dilation measure. The interaction of hearing and cognitive function can be very complicated. Here, on the basis of this thesis, no specified suggestions can be given.

Summarizing, in module B the extent of spatial hearing problems is examined by overall assessment. Module C1 quantifies the role of reduced monaural speech reception in noise and reduced binaural unmasking in these spatial hearing problems. Furthermore, this module indicates to what extent reduced audibility is the cause of the reduced monaural speech reception in noise and reduced binaural unmasking. This essentially determines a quantitative *target* of hearing-aid rehabilitation. The specific role of suprathreshold coding deficits in the reduced monaural speech reception in noise and reduced binaural unmasking can be investigated by module D.

III ILLUSTRATION OF THE USE OF THE MODULAR PROTOCOL

The data of an arbitrary subject, participating in the study described in section 4.2, will serve as an illustration of the way protocol presented in the preceding section can be used in clinical practice although not all extra data are available for this patient.
As a part of module A, the pure-tone audiogram of the subject yields a symmetrical mild sensorineural hearing impairment (Figure 6.2). Extending this by subjective overall assessment, shows activity limitations for all factors except speech intelligibility in quiet (see Figure 6.3).

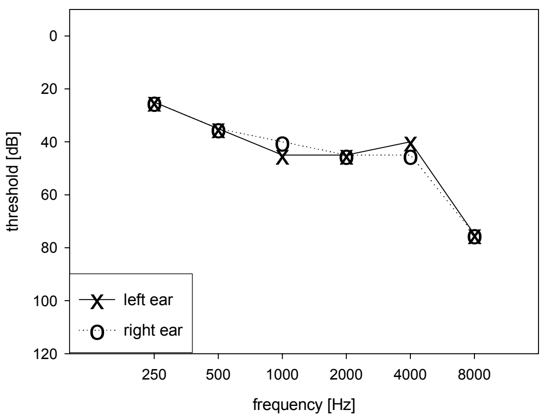

Figure 6.2 Pure-tone audiogram of the arbitrary hearing-impaired patient, whose data ware used to illustrate the use of the modular protocol. No air-bone gap is observed.

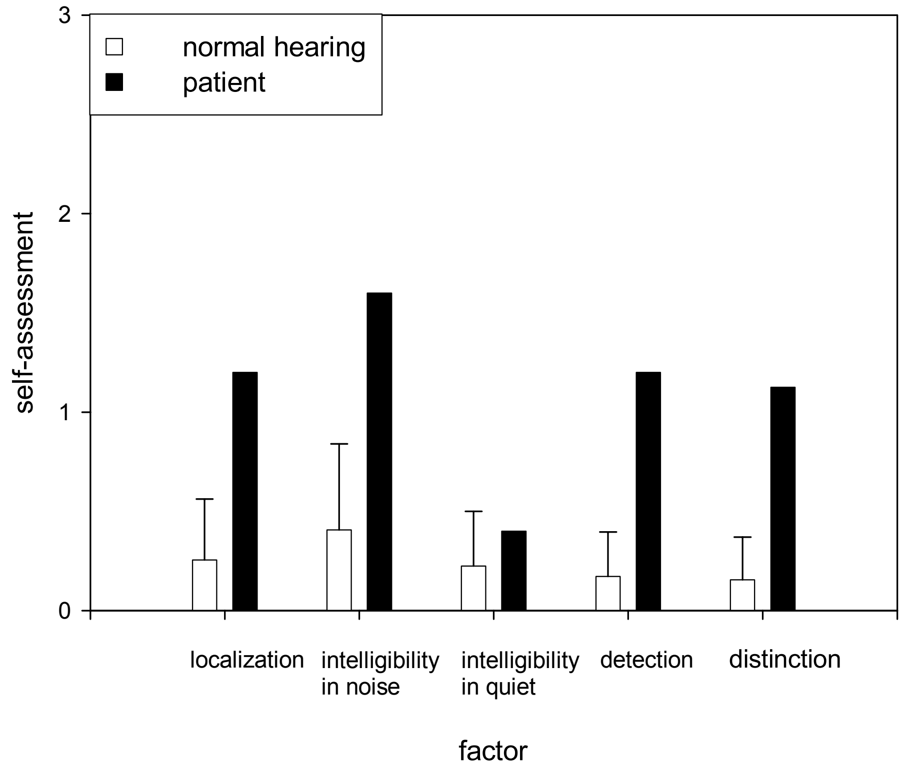

Figure 6.3 Subjective assessment of the auditory functions, by means of the AIADH, differentiating five factors. Norm data from 51 normal hearing are given as well as standard deviations.

For sound localization and spatial speech perception the results are accompanied by psychophysical assessment. The result of psychophysical assessment of sound localization is plotted in Figure 6.4. The score of the patient is 45%. Assessment of spatial speech perception at a presentation level of 65 dB, without correction for audibility (design as described by Boymans et al., 2003) yields a SRT-deviation of 10 dB. The conclusion of module B is that the patient has severe activity limitations, with respect to spatial hearing. The results of specific assessment (module C) are listed in Table 6.1.

Table 6.1 Data for SRT_{N0S0} and $SRT_{N0S\pi}$ as well as the calculated BILD value for the hearing-impaired patient. Median normal hearing data are given as a reference.

Specific assessment with optimized audibility		
	normal hearing	hearing-impaired patient
SRT_{N0S0} [dB]	-2.1	1.0
$SRT_{N0S\pi}$ [dB]	-7.4	-1.8
BILD [dB]	5.3	2.8

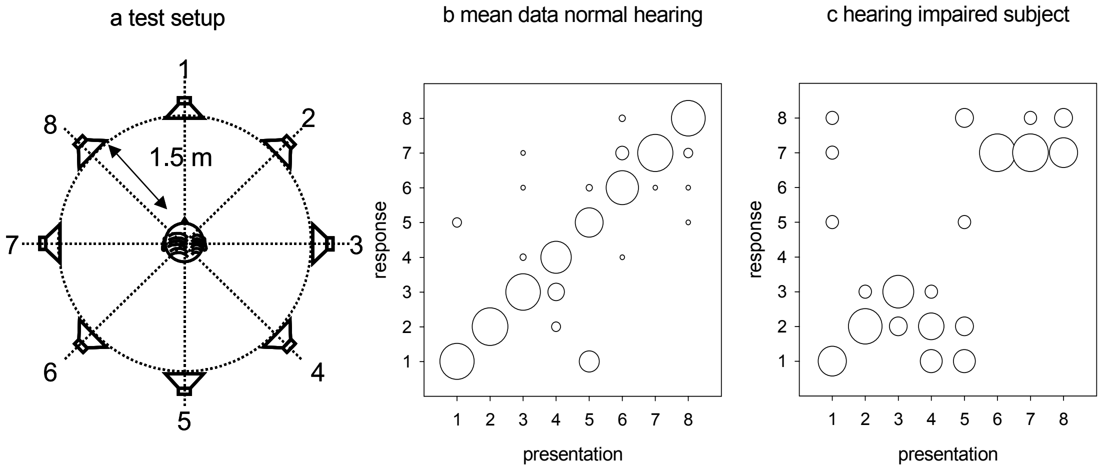

Figure 6.4 Psychophysical assessment of activity limitation in the domain of sounds localization using the test as described in chapter 2 of this thesis. Panel a gives a schematic overview of the test-setup, panel b gives mean normal hearing data, panel c presents data of the this patient. The surface of the bullets represents the percentage of a certain stimulus-response combination.

Only the results with audibility optimization are available for this patient. From these data one can tentatively conclude that this patient has a reduction in monaural speech intelligibility of about 3 dB, and a reduction in binaural unmasking of about 2.5 dB. Thus, these problems together account for about 5.5 dB of the 10 dB found by overall assessment. If data without audibility optimization were available one could have estimated the contribution of audibility. If, hypothetically, the amount of this contribution turned out to be less than 4.5 dB, this would indicate that central aspects might play a role in the reduced overall spatial speech perception in addition to the monaural and binaural performance (module E).

The specific assessment of suprathreshold coding using the distortion sensitivity approach (module D) shows for diotic speech intelligibility reduced sensitivity for distortions in the domains of frequency and intensity. For binaural unmasking a reduced sensitivity in all domains except intensity is found. No data from the tests indicated in module E are available for this patient.

Consequences for auditory rehabilitation

Completing all modules of the protocol would contribute to our insight in the deficits associated with the particular pure-tone audiogram of this patient, and the activity limitations that are caused by these deficits. Secondly, it might also enable us to establish the *target* for rehabilitation by hearing aids, assuming that these should restore audibility optimally. In this

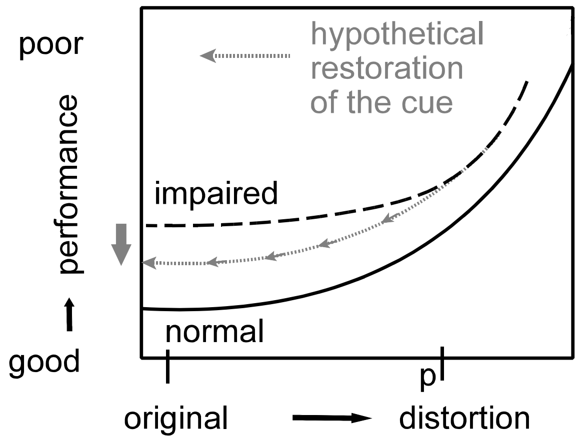

Figure 6.5 Modified replot of Figure 4.2.1 to illustrate a method to estimated the role of a distortion of a specific cue, and thus the hypothetical effect of restoring that cue. For an individual hearing-impaired subject, performance is plotted as a function of externally applied distortion (dashed line). Normal hearing performance is given as a reference (solid line). In the undistorted condition the hearing-impared performance is reduced. Up to a certain degree of distortion, indicated by 'p', the hearing impaired is less sensitive than normal hearing, because of similar internal distortion. The hearing impaired is equally sensitive to distortions of a degree, higher than p; dashed and solid line are parallel from this value of distortion. The grey dotted line represents the hypothetical process of externally restoring the cue. The grey arrow indicates the amount of improvement that can be established.

example the patient had a reduction in spatial speech intelligibility of 10 dB. The results presented in Table 6.1 suggest that about 5.5 dB of this reduction will not be rehabilitated by appropriate amplification. Thus rehabilitation by hearing aids (without directional microphones etc.) is successful if an improvement of about 4.5 dB is reached.

Finally, the distortion sensitivity approach can give an indication what cues in the auditory signal should be enhanced monaurally or binaurally for this particular patient, if possible. This is most easily seen in the results for the $SRT_{N0S\pi}$, in which the effects of distortion on monaural (diotic) speech reception in noise and binaural unmasking are combined. Figure 6.5 illustrates how, in general, the results of the distortion sensitivity approach can lead to a quantitative estimation of the role of a suprathreshold coding deficit. Performance of a hearing-impaired subject is compared to normal hearing data. Up to external distortions of a certain degree ('p', see Figure 6.5), the hearing-impaired subject is less sensitive than normal hearing to the externally applied distortion. For distortions of a higher degree this subject is equally sensitive. The hypothetical process of externally restoring the cue that is distorted internally for this subject, results in a partially improved performance for the hearing-impaired subject. Applying this method to the $SRT_{N0S\pi}$ results of the patient, as presented in

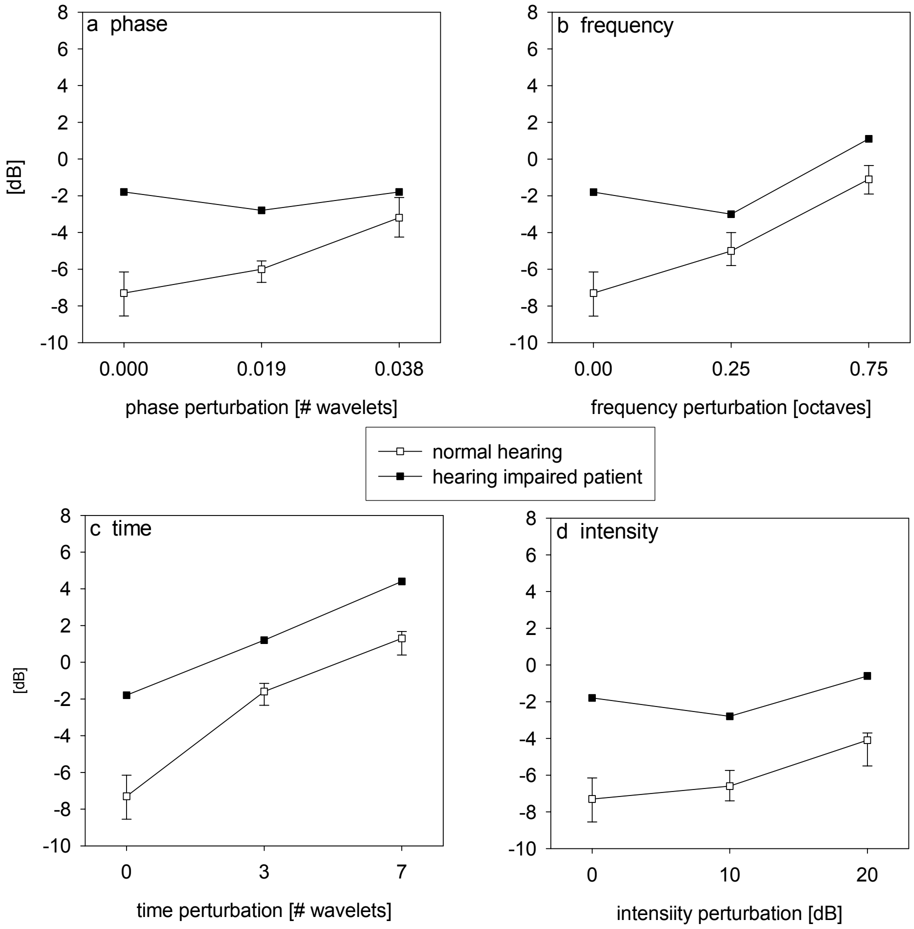

Figure 6.6 Distortion sensitivity of the $SRT_{N0S\pi}$ for the arbitrary patient, compared to the median norm data of six normal hearing. From these data the maximal improvement in spatial hearing that can theoretically be expected if it would be possible to restore the coding deficit by signal processing can be estimated (see text and Figure 6.5).

Figure 6.6 gives the following results. This leads to estimates of about 4, 3 and 3 dB for the cues of phase, frequency and time respectively. Restoring coding of intensity would not yield much improvement. In this rough illustration induced phase effects (see chapter 4.2) are not considered.

These final considerations are still theoretical and highly speculative. For one thing, the perturbations applied in this study are not strictly independent and the expected effects of 4, 3 and 3 dB indicated above cannot be simply added. Also, it is far from obvious by what type of signal processing a specific suprathreshold coding deficit can be compensated or restored.

IV POLITICAL AND SOCIAL IMPLICATIONS

The results of this thesis underline that there is a large variety in deficits associated with the pure-tone audiogram, resulting in a range of activity limitation. A first consequence of this finding is that persons with a sensorineural hearing loss should be referred to as *patients*, with a rather complex and not yet fully understood pathology, rather than clients, customers or hearing aid consumers. A second major consequence of the diversity is that for patients with a sensorineural hearing loss, a relevant and realistic quantitative target of hearing rehabilitation by means of hearing aids, i.e. in terms of speech intelligibility in noise, cannot easily be established. Actually, extension of assessment to module C is needed to establish such a target, otherwise neither the patient, nor the professional can decide whether the results of amplification are optimal (unless of course complete normal hearing function is established). In the current discussion in The Netherlands about the organization of hearing rehabilitation and the role of the different partners, the complexity of hearing impairment, as illustrated in this thesis, is underestimated. There is, of course, a sub population of sensorineural hearing-impaired patients that can be adequately rehabilitated in the setting of the hearing aid dispenser, being easier accessible and less extensively facilitated than the audiology clinic. However, I made clear that identification of this sub population cannot be based on the pure-tone audiogram. Government and health services should therefore keep guaranteeing a structure in which every hearing-impaired patient can, without any barrier, visit all partners in audiological rehabilitation, i.e. hearing aid dispensers, ENT and audiology clinics. Meanwhile the professionals should cooperate in research and finding appropriate criteria that can ensure an adequate routing of the patient, using the specific power and expertise of all partners. Finally, we should provide the hearing impaired with knowledge and insight so that the patient can be really the principal person.

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Summary

Though complex models are known to describe the underlying processes of spatial hearing, in this thesis a simplified model is used as a research framework to assess the effect of hearing impairment on spatial hearing. This model distinguishes audibility, suprathreshold coding, binaural processing and central processing. As a further simplification, the central processing component is not investigated.

The main goal is the *specific psychophysical assessment* of the effect of hearing impairment on binaural processing. More precisely, the role of reduced audibility and suprathreshold coding in binaural processing of the hearing impaired is investigated.

In the first place, in *chapter 2* the relation between psychophysical tests and self-assessment for sound localization, as an example of spatial hearing, is examined. It is shown to be possible to define a listening test to assess overall sound localization performance, the results of which sufficiently agree with self-assessment. The relation between both the psychophysically and subjectively evaluated localization performance and the pure tone audiogram is examined. Significant relations are found, however given a certain degree of hearing loss a large range of localization scores can be found. This implies a scattered relation between localization performance and pure-tone audiometric data, i.e. between overall auditory performance and audibility. In sound localization the hearing loss of the poorer ear is of major importance. Thus, this chapter is the point of departure for chapters 3 and 4 in which the role of audibility and suprathreshold coding in binaural processing is investigated in a more detailed way.

In *chapter 3* the role of audibility problems in binaural processing is further investigated, taking the phenomenon of the precedence effect as an example. A relatively new headphone-based design to quantify the precedence effect is used, aiming to minimize the effects of subjects skills and motivation that are usually inherent to this kind of measurements. The dependence of precedence effect on sensation level is measured for normal hearing in *section 3.1*. These data serve as a set of reference for *section 3.2* where they are compared to data of six mild to moderate sensory neurally hearing-impaired subjects at either equal sound pressure level or equal sensation level. In *section 3.3* the correspondence between the results

obtained using the new head-phone based paradigm and those obtained using a more classic paradigm with loudspeakers is investigated. The latter is used to evaluate the precedence effect at low sensation levels for six normal-hearing subjects and the results are found to be be very similar to those found in section 3.1. Therefore, the non-classic headphone-based paradigm used in this thesis, is an appropriate research tool for investigating the precedence effect for normal hearing and hearing-impaired subjects. It is concluded that reduced audibility causes reduced binaural and spatial hearing. However, the audibility hypothesis cannot fully account for all of the reduced performance of the hearing impaired.

In *chapter 4* the role of suprathreshold coding problems in binaural processing is investigated. While measurements of the precedence effect are time consuming and require much effort of the subject, even using the new design, a more 'friendly' binaural phenomenon is chosen, namely the Binaural Intelligibility Level Difference (BILD), based on speech intelligibility measurements. *Section 4.1* considers the question whether the underlying mechanism in binaural unmasking in speech intelligibility (BILD) is the same as in binaural unmasking in speech detection, the Binaural Masking Level Difference (BMLD). The perturbation effect approach is used. The relation between BMLD and BILD is examined for six normal-hearing subjects in several conditions of signal degradation. The data suggest that binaural unmasking in speech intelligibility reflects essentially the same auditory processing as in speech detection. In *section 4.2* the role of suprathreshold coding in the BILD of 25 mild to moderate hearing-impaired subjects is examined while optimal audibility is assured. In this study the distortion sensitivity approach is used. In a group of 25 mild to moderate hearing-impaired subjects, despite restoring audibility, eight subjects show reduced binaural unmasking. The distortion sensitivity data suggest that this is mainly caused by sub-optimal auditory coding in the domain of phase and, to a less extent, of time. These deficits seem to occur independently of the deficits in coding of frequency and intensity and less often. The effect of hearing impairment on monaural speech intelligibility in noise and binaural unmasking can be diverse. The distortion sensitivity approach is a valuable research tool in specific assessment of spatial and binaural hearing both on a group level and for the individual hearing impaired. Some remaining questions on the precise interpretation of the data would require extensive additional experiments.

In *chapter 5* a general conclusion is given, discussing the results of the successive chapters in relation to the central issue of this thesis, i.e. to enhance our understanding of the role of reduced audibility and suprathreshold coding deficits in the reduced performance of hearing-impaired subjects on binaural and spatial hearing.

In *chapter 6* the implications of the results of this research for clinical practice are discussed and a modular approach for systematic evaluation of binaural and spatial hearing will be presented. The main implication for the diagnosis of the auditory function is the overwhelming diversity of problems that can be associated with one clinical pure-tone audiogram. Therefore access to specialized audiological care should remain guaranteed for the hearing impaired.

Samenvatting

Summary in Dutch

DE BEPALING VAN RUIMTELIJK EN BINAURAAL HOREN BIJ SLECHTHORENDEN

Het horen, onderscheiden en lokaliseren van verschillende geluidbronnen om ons heen wordt ruimtelijk horen genoemd. Ruimtelijk horen is een complex fenomeen. De geluiden om ons heen veroorzaken trillingspatronen aan beide trommelvliezen. Op basis van deze twee trillingspatronen is het gehoor in staat de positie en beweging van de verschillende geluidbronnen te bepalen. Naast waarnemen en onderscheiden komt dan voor de mens nog het verstaan van spraak. We zijn zelfs in staat om spraak te verstaan, te midden van verschillende stoorbronnen, juist als deze ruimtelijk gescheiden zijn van de spreker. Slechthorenden hebben vaak klachten over een vermindering van dit ruimtelijk horen. In wetenschappelijk onderzoek wordt voor diverse tests gevonden dat slechthorenden gemiddeld slechter scoren dan normaalhorenden. Binnen groepen slechthorenden wordt echter een grote spreiding gevonden. Zelfs mensen, die bij standaard toon-audiometrisch onderzoek hetzelfde gehoorverlies lijken te hebben, behalen soms zeer uiteenlopende resultaten.
Om het effect van slechthorendheid op ruimtelijk horen meer in detail te onderzoeken worden in dit proefschrift vier aspecten onderscheiden. Ten eerste kan verminderde hoorbaarheid leiden tot een verminderd ruimtelijk horen. Een tweede aspect is dat geluiden die boven de hoordrempel zijn, zodat ze wel worden waargenomen, als gevolg van de slechthorendheid minder nauwkeurig worden gecodeerd tot neuraal signaal. Ten derde kan de binaurale verwerking van de geluidsignalen verstoord zijn. Tenslotte kan er een storing zijn in de centrale ‘top-down’ verwerking van de auditieve informatie. Een eenvoudig model op basis van deze vier aspecten wordt gebruikt als raamwerk om het effect van slechthorendheid op ruimtelijk horen te onderzoeken. Voor het onderzoek laten we, als een extra vereenvoudiging, de centrale aspecten buiten beschouwing. Uitgaande van dit model van ruimtelijk horen is het hoofddoel van het onderzoek om langs psychofysische weg het effect van slechthorendheid op de binaurale verwerking te bepalen. In het bijzonder richten we ons op de rol van verminderde hoorbaarheid en verstoorde bovendrempelige codering.

Om deze psychofysische benadering een plaats te geven, start het proefschrift in *hoofdstuk 2* met onderzoek naar de relatie tussen psychofysische en subjectieve bepaling van het algehele ruimtelijk horen. Bij 39 slechthorenden wordt de relatie tussen psychofysische testresultaten op geluidlokalisatie en de beoordeling van de geluidslokalisatie door de slechthorende zèlf onderzocht. Het blijkt mogelijk een psychofysische test te ontwerpen, waarvan de resultaten goed overeenstemmen met de subjectieve beoordeling verkregen met een gevalideerde vragenlijst. Tevens wordt de relatie tussen audiometrische data en de resultaten van psychofysische en subjectieve evaluatie onderzocht. Er worden significante relaties gevonden, met name met de gehoordrempel van het slechte oor, maar desondanks blijft er een grote variatie in scores bij een gegeven gehoorverlies. Dit impliceert een grote variëteit in de relatie tussen geluidlokalisatie en toonaudiometrisch vastgesteld gehoorverlies, met andere woorden tussen auditief functioneren en hoorbaarheid.
De *conclusie* van dit deel van het onderzoek is, dat het mogelijk is een overeenstemming te bereiken tussen de resultaten van psychofysische en subjectieve evaluatie van geluidlokalisatie. De correlatie tussen geluidlokalisatie zoals bepaald met deze tests en de audiometrische data is beperkt. Dit gegeven is het vertrekpunt voor de hoofdstukken 3 en 4, waarin de rol van hoorbaarheid en bovendrempelige codering in binaurale processing meer in detail wordt onderzocht.

In *hoofdstuk 3* wordt de rol van verminderde hoorbaarheid bij binaurale processing nader onderzocht bij het “precedence effect” (het verschijnsel dat de van de eerst arriverende informatie de geluidlokalisatie domineert). Hierbij maken we gebruik van een niveau-afhankelijke benadering. De scores van normaalhorenden en slechthorenden worden vergeleken bij gelijk absoluut niveau van presentatie en bij gelijk sensatieniveau boven de hoordrempel. Om het precedence effect te kwantificeren is een relatief nieuw paradigma gebruikt, dat ontwikkeld is met als doel de effecten van motivatie en test-ervaring op de resultaten te minimaliseren. Met deze hoofdtelefoontest is het verband tussen precedence effect en presentatieniveau gemeten voor normaalhorenden. De resultaten dienen als referentieset voor een tweede onderzoek, waarin de resultaten vergeleken worden met data van zes proefpersonen met een symmetrisch, licht tot matig, perceptief gehoorverlies, zowel bij gelijk absoluut niveau als bij gelijk sensatieniveau. Tenslotte is bij zes normaalhorenden het precedence effect bij lage sensatieniveau’s onderzocht met een klassiek paradigma. Daar de resultaten, verkregen met beide paradigma's met elkaar in overeenstemming bleken, is geconcludeerd dat het nieuwe paradigma kan worden toegepast in psychofysisch onderzoek naar het precedence effect.

De *conclusie* van dit deel van het onderzoek is dat een lager sensatieniveau (verminderde hoorbaarheid) bij normaal- en slechthorenden leidt tot een verminderde capaciteit van het binauraal horen. Echter, zowel bij gelijk absoluut niveau van presentatie als bij gelijk sensatieniveau is het precedence effect van slechthorenden minder dan bij normaalhorenden. Tevens hebben slechthorenden met gelijk toonaudiogram soms zeer uiteenlopende resultaten bij onderzoek van het precedence effect. De hoorbaarheidshypothese biedt daarom geen volledige verklaring voor de reductie in prestatie van slechthorenden. In het gekozen eenvoudige model kunnen de overige effecten veroorzaakt worden veroorzaakt door problemen in de verwerking van bovendrempelige geluiden of door specifieke problemen in de binaurale verwerking.

In *hoofdstuk 4* wordt de rol van bovendrempelige problemen bij binaurale verwerking onderzocht. Omdat metingen van het precedence effect veel tijd en inspanning van de proefpersonen vragen, hebben we hiervoor een ander binauraal fenomeen gekozen, namenlijk de binaurale winst in spraakverstaan. Dit wordt gekwantificeerd in de BILD (Binaural Intelligibility Level Difference): het verschil in de verstaanbaarheid van spraak tussen een conditie waarin spraak en stoorruis beide in fase worden aangeboden (SRT_{N0S0}) en een conditie waarin spraak met een faseverschil tussen beide oren wordt aangeboden en de stoorruis in fase worden aangeboden ($SRT_{N0S\pi}$). Deze metingen van het spraakverstaan zijn meer proefpersoonvriendelijk dan metingen van het precedence effect. In *sectie 4.1* wordt onderzocht of het onderliggende mechanisme bij de BILD gelijk is aan het mechanisme bij de binaurale winst in spraak detectie (BMLD, Binaural Masking Level Difference). Hierbij is voor beide fenomenen het effect van verstoringen van signaalcues onderzocht voor zes normaalhorende proefpersonen. Via een wavelet decompositie wordt voor beide oren onafhankelijk aan de spraak en ruis stimuli een verstoring aangebracht in het fase-, frequentie-, tijd- of intensiteitsdomein. De relatie tussen BMLD en BILD is onderzocht voor de verschillende type signaalverstoring. Op basis van de data lijkt het onderliggend mechanisme bij BMLD en BILD inderdaad gelijk te zijn. In *sectie 4.2* wordt de rol van bovendrempelige problemen onderzocht bij 25 proefpersonen met een licht tot matige perceptief gehoorverlies. Daarbij maken we gebruik van een paradigma dat bekend staat als de distortion sensitivity benadering. Via een wavelet decompositie wordt voor beide oren onafhankelijk aan de spraak en ruis stimuli een verstoring aangebracht in het fase-, frequentie-, tijd- of intensiteitsdomein. De gevoeligheid van de BILD voor deze verstoring is gemeten en vergeleken met die van zes normaalhorenden. De onderliggende hypothese is dat wanneer een slechthorende minder gevoelig is voor een type verstoring dan de normaalhorenden, de codering in het betreffende domein is aangetast. Bij deze metingen zijn de stimuli midden in het dynamisch bereik van het gehoor aangeboden om een effect van verminderde hoorbaarheid uit te sluiten. Acht van de 25 slechthorenden blijken een gereduceerde BILD te hebben. De overige hebben dus bij optimale hoorbaarheid geen probleem in de binaurale verwerking. Bij groep van acht proefpersonen blijkt de oorzaak van het de verminderde BILD gelegen te zijn in problemen met fase- en tijdcodering. De resultaten worden vergeleken met die voor het monauraal spraakverstaan in ruis, in het huidig onderzoek benaderd met de diotische conditie (SRT_{N0S0}). In de diotische conditie behaalt de groep van 25 slechthorenden een significant lagere score dan de normaalhorenden. Deze vermindering blijkt met name gerelateerd te zijn aan problemen met de frequentie- en intensiteitscodering. Bij normaalhorenden is de BILD gevoelig voor fase-verstoringen, die het monauraal spraakverstaan in ruis niet aantasten. De BILD is niet gevoelig voor verstoring van frequentie en tijd, waar het monauraal spraakverstaan in ruis wel gevoelig voor is.

In *hoofdstuk 5* worden de resultaten van de voorgaande hoofdstukken bediscussieerd in relatie tot het centrale thema van dit proefschrift: de vergroting van het inzicht in de rol van verminderde hoorbaarheid en verstoorde bovendrempelige codering bij het verminderd ruimtelijk en binauraal horen van slechthorenden.

In *hoofdstuk 6* worden de implicaties van de resultaten bediscussieerd voor de klinische praktijk van huisarts, KNO arts, audiologisch centrum en audicien. De belangrijkste implicatie is dat er een grote diversiteit aan problemen geassocieerd kan zijn met een zelfde toonaudiometrisch gehoorverlies. Voor de klinische praktijk wordt een modulaire benadering voor een systematische evaluatie van ruimtelijk en binauraal horen gepresenteerd en geïllustreerd. Dit proefschrift onderstreept dat de gevolgen van gehoorverlies divers kunnen zijn en vaak slechts ten dele begrepen worden, zeker op het niveau van de individuele patiënt.

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Curriculum vitae

S. Theo Goverts was born in Leiden on May 28th, 1970. He attended high school at *Chr. Lyceum dr W.A. Visser 't Hooft* in Leiden (1982-1988). From 1988 to 1994, he studied experimental physics at the *Vrije Universiteit* in Amsterdam. He conducted his Master's thesis work on auditory discrimination and speech intelligibility under supervision of dr.ir. J.M. Festen and prof.dr.ir. T. Houtgast. In 1994 and 1995 he worked on a project on optimal hearing aid fitting rules in the audiology section of the University Hospital *Vrije Universiteit*. In 1995 and 1996 he was employed as a project manager at the TNO human factors institute in Soesterberg where he was concerned with active noise reduction and speech transmission index measurements. In 1996 he started the training to be a clinical physicist audiologist in the audiology centre of the Otorhinolaryngology / Head & Neck Surgery of the University Hospital *Vrije Universiteit* (supervisor dr. T.S. Kapteyn). After the training was finished in 1999, he joined the ENT-audiology staff and since 2001 he is head of the audiology centre. During the training he initiated the current research under supervision of prof.dr.ir. T. Houtgast which resulted in this thesis.

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