Enhanced Procedures for Assessment of Hearing Impairment

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Abstract

In clinical audiology, the assessment and remediation of patients is mainly based on the audiometric thresholds of the patient, which characterize the sensitivity of hearing. On the basis of this information, the administration of hearing aids and the fitting parameters most suitable for the listener are decided. In this thesis it will be suggested that the diagnosis and remediation of the hearing problem would benefit from a more detailed assessment of the patient's hearing. Information reflecting the frequency selectivity, compression and sensitivity of the hearing would be collected and form the basis of a 'hearing profile' of the listener. The hearing profile can subsequently be used to gain insight in the underlying pathology of the impairment, decide on suitable fitting strategies and form the basis for future developments of a computer model of the impaired listener's hearing.

In order to make the 'hearing profile' a clinically viable tool, efficient and reliable threshold estimation procedures need to be employed when collecting data. A single-interval procedure is developed based on a modification of Green's Maximum-likelihood procedure (1993). Using this procedure, it will be shown that a clinically suitable accuracy of the threshold estimates can be obtained in as few as 10 trials.

The hearing profiles of three normal listeners are presented and will be used to evaluate the validity of the profiles. Subsequently, the hearing profiles of four impaired listeners are presented. It will be shown that the profiles can be collected in impaired listeners and provide valuable information over and above audiometric thresholds.

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List of abbreviations

2I2AFC	two-interval, two-alternative forced choice
AN	auditory nerve
BF	best frequency
BM	basilar membrane
CF	center frequency
dB SPL	decibels sound pressure level
dB HL	decibels hearing level
$\mathbf{D_m}$	masker duration
$\mathbf{D_p}$	probe duration
DP OAE	Distortion Product Otoacoustic Emission
EP	endocochlear potential
$\mathbf{f_m}$	masker frequency
$\mathbf{f_p}$	probe frequency
GUI	Graphic User Interface
Hz	Hertz
IFMC	Iso Forward Masking Curve
IHC	inner hair cell
ML	Maximum-likelihood
ms	millisecond
µPa	microPascal
OHC	outer hair cell
OME	outer middle ear
PTC	Psychophysical Tuning Curve
RMS	Root Mean Squares
s	second
SD	standard deviation
SI-50%	Single Interval 50%

SI-UD	Single-Interval-Up-Down method
SNR	Signal-to-Noise-ratio
TMC	Temporal Masking Curve

CHAPTER 1

Introduction

This thesis is concerned with developing detailed profiles of impaired listener's hearing suitable for use in a clinical setting, employing efficient and accurate measurement procedures.

1.1 Motivation

Basic audiological assessment consists of screening of the outer, middle ear functions and a screening of the sensitivity of the auditory system by means of an audiogram. On the basis of this information, the audiologist decides which hearing aid is most suitable for the listener and which fitting strategy would provide the best results in terms of improving hearing performance (Dillon, 2001).

Finding the right hearing aid prescription for a specific hearing impaired individual is not straightforward. Although many hearing impaired people experience the hearing aid as a meaningful help in their everyday life, other do not share that same experience. This project is motivated by the belief that the information provided by the audiogram is not sufficient to form a clear picture of an individual's hearing abilities and additional information might prove to enhance the fitting of the hearing aids. Also, it is generally accepted by clinicians that patients with a similar audiogram do not necessarily respond equally well to their similarly-fitted hearing aids. This could be an indication that the origin of the hearing loss is not the same and differentiation in hearing aids and remediation is advisable.

In this thesis we suggest the development of an individual ‘hearing profile’ of impaired listeners. The hearing profile will contain a wide range of psychophysical information on a person’s hearing not currently tested in clinical practise. The hearing aspects which will be tested are sensitivity, frequency selectivity and compression. A hearing profile of a patient will provide the clinician with a large set of data. This will enable him to fit a hearing aid more tailored to the patient. Furthermore, hearing aids are becoming increasingly sophisticated. More features are available and the information will also allow for the features to be used more appropriately and efficiently. Finally, the hearing profile can be the basis to hypothesize about the underlying pathology of the hearing problem.

The development of the ‘hearing profile’ is part of a larger project which aims to develop computer models of impaired listeners’ hearing. Its basis is a computer model of the normal auditory periphery (Meddis, 2006). This model will be adapted to simulate the hearing of an impaired individual. Once a computer model of a patient’s hearing is available it can be used as a platform to evaluate different kinds of hearing aids and fitting strategies to determine which is the best fitting for a given patient. The choice of tests included in the hearing profile has been guided by the need to collect sufficient information to allow modelling of the patient. Due to time constraints, no modelling work will be presented in this thesis. However, the work presented in here has to be seen in the bigger context of the future modelling work.

1.2 The hearing profile: what information does it provide?

The hearing profile is aimed at presenting clinicians with an extensive set of measurements of an individual’s hearing.

The audiometric thresholds used in the clinic offer information on the sensitivity of a listener at threshold. In particular, an audiometric threshold represents the lowest intensity at which the listener can identify the presence of a signal at least 50% of the time (Katz, 2002). This measure informs us on how a single tone/stimulus is detected in a quiet, often sound-attenuated environment. In real life however, the demands put on our auditory system are not often situated at threshold level and in a quiet environment. Most stimuli such as speech and surrounding noise apply at supra-threshold level.

The nature of supra-threshold measurements is quite different from measurements at threshold in quiet. First, supra-threshold measurements look at how a signal interferes and interacts with additional stimuli such as pure tones, speech or noise. Moreover, a hearing impairment will not only affects the sensitivity of the auditory system but will very often introduce a variety of changes in the perception of the signal (such as distortions) that are above the threshold of detection (McFadden et al., 1997; Wojtczak and Viemeister, 2003; Moore, 2007). The two main supra-threshold measurements reported in this thesis are measures of frequency selectivity and compression and will be explained in more depth below. The selection of these two measures is based on the fact that both phenomena are well described and extensively researched. Also, it is envisaged that the resulting information is compatible with the contemporary generation of hearing aids. Testing the speech recognition in noise is undoubtedly another important supra-threshold measure, since difficulties with listening to speech in a noisy environment is the main complaint of impaired listeners. Unfortunately, due to time constraints it was not possible to include this measure in the project described in this thesis.

A major complaint heard from hearing impaired listeners is the fact that listening in a noisy environment is difficult. Ambient noise interferes with the sound source the person is trying to attend to. *Frequency selectivity*, or how one can pay attention to one sound while ignoring others, is commonly measured using Psychophysical Tuning Curves (PTCs). In normal listeners, the tuning is sharp which means that interfering sounds are ignored more easily. In impaired listeners however, the tuning is often less sharp, resulting in problems when trying to focus on one sound source in the presence of interfering sources. For example, it is well established that PTCs are broader in listeners with a sensorineural hearing loss (Carney and Nelson, 1983; Nelson, 1991; Snik and Horst, 1991; Stelmachowicz et al., 1985; Florentine, 1992; Moore and Glasberg, 1986).

The value of measuring Psychophysical Tuning Curves in a clinical setting has been discussed by a number of researchers (Stelmachowicz and Jestaedt, 1984, Vanden Abeele et al., 1992, Halpin, 2002, Glasberg and Moore, 2000). Information on the status of the tuning of the listener can be valuable when fitting hearing aids. It tells us how sounds of different frequencies will interfere with each other. This can subsequently be taken into account when finding the best fitting parameters for the hearing aid of that specific listener.

In recent years, a lot of attention has been given to the concept of ‘dead regions’. The term ‘dead region’ refers to a non-responsive frequency region due to poorly functioning hair cells (Moore, 2007). The actual severity of the damage in these regions is not necessarily reflected in the audiogram and the audiometric thresholds found at the non-functioning frequencies are likely to be underestimates of the ‘true’ threshold. The mechanism behind this is called ‘off-frequency listening’ and implies that the tone is detected at an adjacent/more sensitive place on the basilar membrane (BM) due to spread of excitation. It has been suggested that dead regions need to be taken into account when fitting hearing aids (Vickers et al., 2001; Baer et al., 2002). More specifically, amplification of frequencies falling well within dead regions may not provide any benefit and might actually be unfavorable to the speech recognition performance of the person. Dead regions can be diagnosed by measuring PTCs (Thornton and Abbas, 1980; Florentine and Houtsma, 1983; Moore and Alcantara, 2001; Summers et al., 2003; Kluk and Moore, 2005). A shift in the tip of the PTC indicates a dead region. An alternative method for diagnosing dead regions in the clinic has been proposed by Moore et al. (2000, 2004). The TEN-test involves measuring thresholds in a specifically designed background noise called threshold-equalizing noise (TEN). The test predicts that significantly higher thresholds will be measured for frequencies associated with a dead region compared to 'normal' frequencies (Moore, 2007). Comparisons between the PTC-method and the TEN-test suggest that PTCs provide a more accurate method for determining the frequency limits of dead regions (Summers et al., 2003, Kluk and Moore, 2005). In this thesis we will be using a procedure similar to PTCs to acquire information on the tuning properties and possible dead regions in our impaired listeners.

A second important measure included in the hearing profile is *compression*. Auditory compression is a result of the basilar membrane responding compressively to sounds (Rhode, 1971; Rhode and Robles, 1974; Ruggero et al., 1997; Rhode and Recio, 2000). This means that an increase in the magnitude of stimulation does not produce a proportional increase in the velocity or displacement of the basilar membrane (BM) vibration. Temporal Masking Curves (TMCs) are a measure of the compressive properties of the basilar membrane (Nelson et al., 2001; Lopez-Poveda et al., 2003, 2005; Plack and Drga, 2003, Plack et al., 2004; Rosengard et al., 2005). In impaired listeners, it is believed that basilar membrane responds in a more linear manner, meaning that compression is reduced or absent (Nelson et al., 2001, Plack et al., 2004; Lopez-Poveda et al, 2005, Stainsby and Moore, 2006). Some researchers have suggested that in certain impaired listeners the compression is not necessarily reduced but that the intensity range over which the basilar membrane responds non-linear is merely reduced (Plack et al., 2004; Lopez-Poveda et al, 2005). We are interested in exploring the compressive properties in a listener's hearing. Knowledge on the amount of compression and the intensity range over which compression occurs for a range of frequencies will be useful when fitting a hearing aid. More specifically, the overall assumption that compression is absent in impaired listeners has led to the implementation of compressive mechanisms in hearing aids. However, when TMCs reveal residual compression over a limited range of intensity levels, there is no need to add extra compression by means of the hearing aid. In fact, adding more compression would mean the listener is dealing with a greater-than-normal compression and this could potentially reduce the benefit of the hearing aid.

The third aspect that will be explored is the *dependence of threshold on the duration of the stimulus*. Thresholds are lower for longer tones than for shorter tones. In normal listeners, a threshold difference of 2.4-3 dB per octave of duration is typically found. (e.g., Hughes, 1946; Plomp and Bouman, 1959; Sheeley and Bilger, 1964; Olsen and Carhart, 1966; Hall and Fernandes, 1983; Florentine et al., 1988). Hearing-impaired listeners are reported to have shallower threshold/duration functions than normal listeners. A threshold shift of around 0.6-0.9 dB per octave has been described for this group (Wright, 1968, Watson and Gengel, 1969, Gengel and Watson, 1971; Elliott, 1975; Dempsey and Maxon, 1982; Hall and Fernandes, 1983, Florentine et al., 1988, Gerken et al., 1990).

The difference in the threshold/duration effect between normal and impaired listeners is potentially important from a diagnostic point of view. This clinical application of the threshold/duration effect, called ‘brief-tone audiometry’, has been explored by a number of researchers prior to 1990 but has never been accepted as a routine audiological tool (Wright, 1978). One explanation is that the effect is so variable that the various measures could not be used to discriminate reliably between normal and impaired listeners. Moreover there is evidence that only some types of impairment produce a change in the threshold/duration function. Reduced threshold/duration effects have been reported for listeners with presbycusis, noise-induced hearing loss, cochlear hearing loss and Meniere's disease. Listeners with a conductive hearing loss or an eighth nerve lesion appear to have normal threshold-duration functions (Harris et al., 1958; Pedersen and Elberling, 1973; Young and Kanofsky, 1973; Olsen et al., 1974; Pedersen and Salomon, 1977; Chung and Smith, 1980; Chung, 1982). Unfortunately the variability of the threshold/duration effect among listeners for both normal and impaired listeners has prevented clinical implementation of this phenomenon. A new method to analyse threshold/duration functions will be explored and the potential value from a clinical point of view will be discussed

1.3 Simple, efficient and accurate threshold estimation methods

It is probable that clinicians will take a rather pessimistic view of this extensive set of tests. They might not feel it is feasible in a clinical setting mainly because of the time needed to collect all the data in the profile. We have paid special attention to this concern. A second issue we were faced with is that the task participants (and patients) will perform needs to be simple. It is important that a participant/patient understands the measurement task with a minimum of instructions. Furthermore, once the measuring has started, reliable estimates need to be obtained with a minimum of training. A substantial part of this thesis is concerned with the development of a threshold estimation procedure which is simple, fast and reliable and can be used both in a clinical and a laboratory setting. The procedure will subsequently be used in the psychophysical measures presented later in this thesis.

In the clinic, the standard threshold estimation procedure used is the 'modified Hughson-Westlake procedure' proposed by Carhart and Jerger (1959). Hughson and Westlake (1944) initially devised a quick method for searching auditory thresholds by starting a test run with an inaudible sound level. The signal level was increased until a positive response was elicited from the patient. Carhart and Jerger subsequently proposed some modifications by beginning the test at a high signal level aimed at demonstrating the sound to the listener. The level was subsequently dropped in large steps (10-dB steps) until the sound became inaudible (negative response of the listener) after which the level was increased in smaller steps (5-dB steps) until the listener responded positive again. The threshold is set at the lowest level that yields at least three positive responses in the ascending tracks. The information used to determine the threshold is distilled only from the ascending tracks. The descending tracks function mainly as probes for the listener.

It is likely that early adaptive algorithms like the ones proposed by Hughson and Westlake (1944) and Carhart and Jerger (1959) are at the basis of the modern adaptive procedures used in research. The main reason for the current success of adaptive methodologies is that they converge rapidly on the threshold, allowing efficient threshold estimations. A paper by Leek and colleagues (2000) drew our attention to a single interval adaptive procedure suggested by Green (1993). Green's Maximum-likelihood method (ML) uses a single interval yes-no paradigm coupled with the maximum-likelihood adaptive tracking rule. Two aspects of this procedure were important for us with regards to finding a simple and efficient estimation procedure; namely the task the participant is asked to do and the fact that an adaptive rule was used.

A single interval procedure is associated with a simple task, which requires only a yes-no judgment of the listener. A stimulus is presented and the listener is asked whether or not it is heard. It is similar to the task used in the clinic where patients need to respond when a signal has been detected.

The adaptive rule used in Green's method is a maximum-likelihood adaptive tracking rule. A maximum-likelihood procedure is characterized by estimations of the underlying psychometric function after every response. After each trial, the set of stimulus levels and their associated proportions of YES-responses are combined to form a psychometric function. A maximum-likelihood fitting algorithm is typically used to fit the psychometric function and as more responses are collected, the estimate of the best psychometric function is refined. The stimulus placement is usually based on the level associated with the 50 % point of the psychometric function. A threshold estimate is extracted from the latest 'best-fit' of the psychometric function after each response. The final threshold is determined after a fixed number of trials and is the stimulus level associated with a pre-determined point on the 'last fitted' psychometric function, often set at 50% (Leek, 2001). The main advantage of Maximum-likelihood adaptive procedures is that the stimulus level converges onto the threshold rapidly. This fast convergence is possible because the stimulus level of a presentation is based on the previous responses of the listener (Leek et al., 1992) and is set at the point of the psychometric function where the threshold is expected to be (50%-point). A second advantages is that a threshold is determined using all responses in the test, unlike the 'modified Hughson-Westlake procedure' where only the responses in the ascending tracks contribute to the threshold estimation. Different maximum-likelihood procedures (ML) have been proposed by researchers such as Pentland ('the best PEST', 1980) and Watson and Pelli (QUEST, 1983). Green's method can currently be considered as the most commonly known and used ML-procedure.

In most research settings however, single-interval procedures are not commonly used and multiple-interval forced choice procedures are the standard. In a N-interval, N-alternative forced choice procedure (NI-NAFC), two or more temporal intervals are marked and the signal is presented in one of the intervals. The listener has to indicate in which interval the signal was detected. Although forced-choice procedures are commonly used in research, they are not necessarily suitable for use in a clinical setting. The listeners are asked to select the interval where a target was perceived. Even when no target is perceived, the participant must still select an interval at random. For patients, who are used to responding when a signal is heard, it could be uncomfortable to select an interval when this is not the case. This situation will occur on each trial where the signal is below threshold.

Forced-choice procedures are often combined with a staircase adaptive procedure. Staircase procedures are simpler than the maximum-likelihood procedures since no assumption needs to be made about the form of the psychometric function. The level of the stimulus is simply reduced when the listener's response is positive and increased when the response is negative. Levitt (1971) proposed a transformed up-down procedure which is now commonly used in forced-choice procedures and allows targeting different levels of performance in the estimation of the threshold, yielding different levels of accuracy. For example, the two-down, one-up procedure targets the 70.7 % point on the psychometric function. In this procedure, two correct responses are required to generate a decrease in the stimulus level and only one incorrect response is needed for an increase in the stimulus level. A three-down, one-up procedure will target the 79% point. The point at which the direction of the level-track changes is called a reversal. A threshold run usually ends after a fixed number of reversals and the final threshold estimate is calculated by averaging the levels of the reversals in the adaptive track The disadvantage of this method is that considerably more trials are necessary to yield the required number of reversals before an estimate of the final threshold can be made. This is the main reason why multiple-interval, forced-choice procedures are significantly more time-consuming than single-interval procedures.

An advantage of the forced-choice paradigms is that they are assumed to be unbiased and have the ability to control the listener's criterion. When a listener responds, an evaluation can be made whether the response was correct or incorrect. In Green's ML, the listener indicates whether a stimulus was heard or not. An evaluation is not possible and the experimenter can not control the response criterion of the listener. This issue has been addressed by estimating the false alarm rate of the participant and taking this false alarm rate into account in the estimates of the psychometric function. Estimating the false alarm rate has proven to be complicated and Gu and Green (1994) suggested adding catch trials to the test trials. A catch trial is a trial where no stimulus is being presented. On the basis of the listener's response to these catch trials a fairly reliable false alarm rate can now be estimated. A disadvantage however is that a large number of responses are required to obtain a reliable estimate.

In this thesis a new single-interval procedure, referred to as Single-Interval-Up-Down (SI-UD) will be presented. This is the threshold estimation procedure which will be used when making the psychophysical measurements presented in the hearing profiles. The SI-UD method is based on Green's Maximum-likelihood procedure. However, a number of adjustments were implemented to cater for the specific needs of this project. The main adjustments are concerned with a) adding a cue to the stimulus to facilitate the task, b) implementing an alternative use of catch trials to improve reliability and c) employing a simple one-up, one-down staircase adaptive track to enable a further simplification of the method. A detailed discussion of these adjustments will be presented in Chapter 2. Subsequently it will be shown that the procedure is simple, fast, reliable and gives similar results to the more traditional 2I2AFC-methods.

1.4 Summary

This thesis is concerned with developing a set of procedures to evaluate the hearing of impaired listeners. The procedure for the threshold measurements needs to be user friendly, efficient, reliable and fast. This method will then be deployed over a range of tests concerning key aspects of the patient's hearing.

In Chapter 2, we present the new Single-Interval-Up-Down method and describe the adjustments made to accommodate for our specific measurement requirements.

In Chapter 3, we investigate how the accuracy of threshold estimates changes with the number of trials used in a threshold run. The aim of this exercise is to find the smallest number of trials we can use for a required level of accuracy when making the measurements for the hearing profiles of the normal and impaired listeners.

Chapter 4 is concerned with evaluating the Single-Interval-Up-Down method on a number of issues. First, we look at user friendliness and how much training is required when introducing the method to listeners. Secondly, the Single-Interval-Up-Down method will be compared with a traditional 2I2AFC, two-down, one-up procedure in terms of threshold estimates, accuracy and speed.

Chapter 5 outlines the psychoacoustic measures that are used to create a psychoacoustic profile of the participants. Measures used will absolute thresholds, Iso Forward Masking Contours (IFMC) and Temporal Masking Curves (TMC).

In Chapter 6, we explore how thresholds change as a function of the duration of the stimulus in normal and impaired listeners and propose a two-parameter reciprocal function to fit the threshold/duration functions. This fitting procedure will be evaluated on its potential to distinguish between normal and impaired listeners.

In Chapter 7 the hearing profiles of 3 normal listeners are presented. The results will be evaluated to justify using the assessment protocol with impaired listeners.

In Chapter 8, hearing profiles of 4 impaired listeners are generated and discussed.

In Chapter 9, a summary will be presented of the main achievements of the work presented in this thesis, alongside with a discussion of the limitations. Subsequently, a presentation of future research will be given.

CHAPTER 2

The Single-Interval-Up-Down procedure

2.1 Introduction

The individual hearing profile of a person is a collection of measurements obtained in a number of psychoacoustic tests. Its aim is to generate an insight into the pathology of an impaired listener and to provide useful information concerning the remediation of the pathology. The development of the hearing profile however requires a large number of measurements. When faced with the task of collecting a large data set on individual listeners, one needs to carefully consider the measurement techniques one will adopt. It was our aim to employ a measurement procedure suitable both in a laboratory and in a clinical setting. This entails the development of a procedure which consists of a straightforward task, simple procedures and fast but reliable threshold estimation.

A efficient threshold estimation procedure has been suggested by Green (1993) and Gu and Green (1994). His Maximum-likelihood method entails a simple yes-no task combined with a maximum-likelihood adaptive procedure. Participants are asked the simple question ‘Did you hear the signal, yes or no?’ The procedure is designed to find the psychometric function among an array of possible candidate functions that most probably underlies the behaviour reflected in a data set. The shape of the underlying psychometric function is assumed to be a logistic function. Each function is described with a location along the stimulus axis, a slope (estimated change in performance with a given change in stimulus level) and an assumed false-alarm rate. Throughout the threshold search, each of the candidate functions is evaluated after each trial. The probability of each function is upgraded based on the responses on all previous trials to determine which function most probably underlies the listener's behaviour. After a fixed number of trials the threshold search ends and a threshold is estimated from the best psychometric function. Green (1993) suggested that a threshold estimate with a 3-dB standard deviation could be obtained in as few as 12 trials.

Green's procedure seemed suitable for our measurement since the task is simple and the adaptive stimulus selection allows a fast convergence on the threshold. Initial measurements were conducted using this method. However, it was felt that a number of adjustments could be made to further improve the procedure and to cater for the specific needs of the project. More specifically, the procedure needs to be user-friendly, simple, efficient and reliable. Several modifications were implemented over a course of time which resulted in a threshold estimation method which we will refer to as the Single-Interval-Up-Down method.

In the first section of this chapter, the Single-Interval-Up-Down method (SI-UD) will be presented for use in a detection task in quiet and for a detection task in the presence of a forward masker. The second section in this chapter is concerned with describing the stages of development of this new procedure, discussing the initial procedures and the adjustments implemented to adhere to the criteria of user-friendliness, simplicity and reliability.

The Single-Interval-Up-Down procedure will subsequently be used in the measures presented in the hearing profiles of normal and impaired listeners (Chapters 7 and 8).

2.2 The Single-Interval-Up-Down Procedure

In this section the Single-Interval-Up-Down method will be presented for two different tasks, namely measuring absolute thresholds (detection of a tone in quiet) and measuring forward-masked thresholds (detection of a tone in the presence of a forward masker). The procedure is similar for each task and differs only on a limited number of points. For clarity however, the two detection tasks will be described separately in the following section.

2.2.1 Detection of a tone in quiet

Task

The basic procedure for measuring thresholds is a yes-no task. In a simple yes-no task a single stimulus is presented to the participant who responds 'YES' or 'NO' according to whether or not the stimulus was heard.

Stimulus selection

The level of the stimulus is selected according to a one-up one-down staircase method. If the participant's response is 'YES' the stimulus level is reduced by a fixed step size; if the response is 'NO' the level is increased by the same step size. The threshold run starts with an initial phase (IP). In the initial phase, the stimulus level starts at supra-threshold level (generating YES-responses) and is adjusted using a large step size (called the 'Initial step size') until the first NO-response. At that point (first reversal) the Threshold phase is initiated. The step size in the Threshold phase is reduced compared to the Initial step size and will be referred to as the 'step size'. Figure 2.2-1 shows an illustration of a typical threshold run.

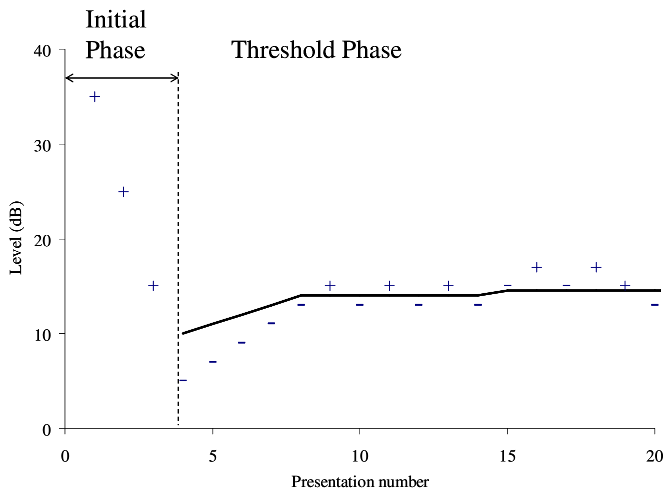


Figure 2.2-1. Illustration of a threshold run for the detection of a tone in quiet. Plus-signs represent the YES-responses, Minus-signs represent the NO-responses. The full line represents the threshold estimate after each response. The first threshold estimate is calculated at the time of the first reversal. An initial step size is used in the Initial phase, the general term 'step size' refers to the smaller step size used in the Threshold phase.

The start of the Threshold phase is also the start of the 'trial count' of the threshold run and the 'Threshold phase' (TP) ends after a previously determined number of trials. It needs to be noted that the last 'YES' of the initial phase is included in the threshold calculations although this response does not contribute to the trial count.

Threshold estimation

The threshold is estimated at the end of the threshold run by finding the logistic function that best describes the accumulated data in the Threshold phase.

The logistic function of the form

$$p = \frac{1}{1 + e^{-k\,(L-Th)}} \qquad (1)$$

is fitted to the YES/NO-responses from the Threshold phase. *P* is the proportion of 'YES' responses, *L* is the level of the presented stimulus, *k* is a slope parameter and *Th* is the estimated threshold (in dB SPL), i.e. the level of the stimulus at which the proportion of YES-responses is 0.5. The best-fit psychometric function is found using a least-squares method where k and Th are free parameters. An example of the resulting function when fit to a number of YES/NO-responses is given in Fig. 2.2-2. The Euclidian distance mentioned in this figure refers to the RMS (Root Mean Square) error between the function and the data.

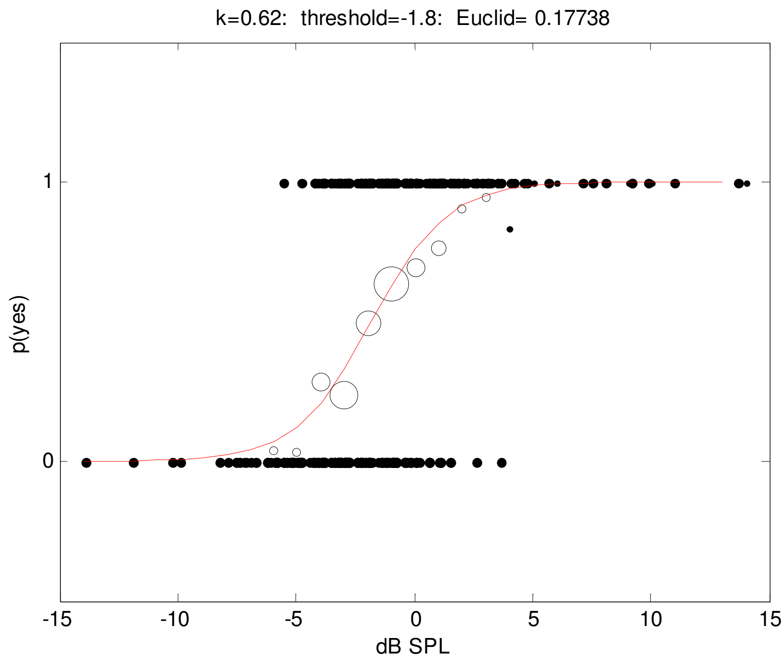


Figure 2.2-2. Illustration of psychometric function fitted to YES/NO-responses (black dots) in a detection task of a tone in quiet. The fitting is performed using a least square best fit method allowing slope (*k*) and threshold (*Th*) to vary, based on 620 responses. The size of the open circles represents the number of responses associated with a specific probability. The Euclidian distance refers to the RMS error between the function and the data. The data is taken from a threshold measurement of a 2000-Hz tone in a normal listener (WL), see chapter 3.4.

Procedure

Stimulus. The stimulus consists of a test tone preceded by a cue (see Figure 2.2-3). In this cued task, the listener hears a cue presented before the test sound. The cue is similar in all respects to the test sound but is presented well above threshold so that it is easier to hear. The listener is asked to count the number of sounds he hears including the cue stimulus. When the cue-tone and the test-tone are heard, the participant answers “2”, which is processed as a YES-response. When the test-stimulus is below threshold, the participant will answer “1” or “0” (0 in the case where the cue is not heard), and a NO-response is recorded. The purpose of the cue in our yes-no task is to remind people what they are listening for. The cue tone is generally always above threshold and therefore detectable. It implies that listeners have some kind of anchor point which is the same distance from the test target in all participants. The intensity difference between cue and test tone is referred to as the CueTestDifference and is usually set at 10 dB SPL.

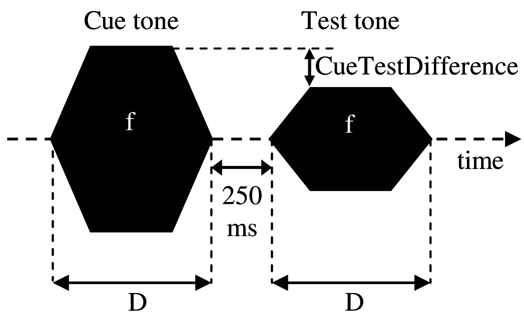


Figure 2.2-3. Schematic representation of the stimulus in a detection task in quiet. The cue is identical to the test tone in terms of duration (D) and frequency (f) but is presented at a stimulus level more intense than the test tone level. This difference in intensity level is determined by the CueTestDifference.

The listener responds by selecting the button which corresponds with the number of tones heard (0, 1 or 2) using a response box. A visual display in front of the listener shows a GUI (Graphical User Interface) depicting the response box (Figure 2.2-4). While the stimulus is being presented, the selection buttons disappear from the GUI. The buttons re-appear at the end of the presentation interval to inform the participant a response is required. The listener subsequently responds by pressing the appropriate button on the response box.

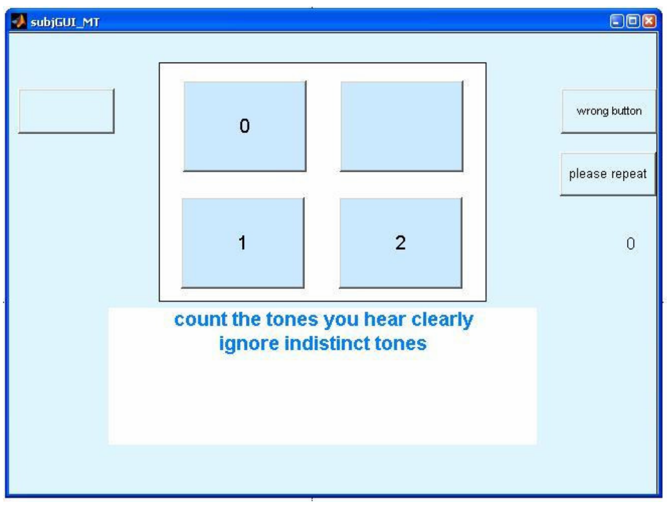


Figure 2.2-4. Graphical User Interface for participants.

Catch trials. Catch trials, where no stimulus is presented, are presented at random interspersed among the regular trials. If a 'YES' response occurs following a catch trial, the participant is informed, the run is aborted and is restarted. Otherwise no feedback is given. The participant is strongly encouraged not to indicate the presence of a sound if he is uncertain and this is enough to produce very few false positives. Initially, 20% of trials in each run are catch trials. However, the catch trial rate is gradually reduced between runs to no less than 10% if there are no instances of false positives. The absence of false positives indicates that the participant is reliable in his/her responses reducing the need for catch trials. When false positives are more common, the catch trial rate is increased after each false positive up to a maximum of 50% of the trials. Catch trials are not included in the total trial count.

2.2.2 Detection of a tone in the presence of a forward masker

Detection of a tone in quiet as described in the previous section is a simple and straightforward task. A slightly more complex task is involved when thresholds are estimated for tones in the presence of a forward masker (forward-masked thresholds). The estimation procedure is similar to the procedure for detecting a tone in quiet. The main difference lies in the task and the stimulus selection rule.

Task

Similar to the detection task for a tone in quiet, the procedure for measuring forward-masked thresholds consists of a yes-no task. A probe tone preceded by a masker is presented to the listener (see Figure 2.2-5). The listener is asked to respond 'YES' or 'NO' according to whether or not the probe tone was heard. The basic idea is that the listener ignores the masker and listens for the probe only.

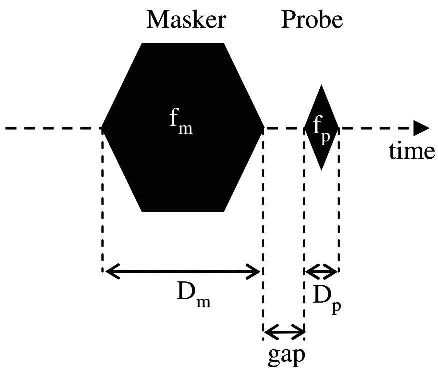


Figure 2.2-5. Schematic representation of a tone (probe) in the presence of a forward masker. (D_m = masker duration, D_p = probe duration, f_m = masker frequency, f_p = probe frequency).

Stimulus selection

In the forward-masking task, we are estimating the threshold of the masker. In other words, the masker level is varied in order to establish how intense the *masker* needs to be to just mask a probe. The probe is fixed in level. The listeners are asked to indicate whether they heard a probe 'YES' or 'NO'. When the probe is detected the masker level has to be increased in order to make it more difficult to detect the probe. When no probe is detected, the masker level will be decreased in order to make the probe audible again. The adjustments of the masker level are in the opposite direction of the level adjustments in the detection-in-quiet-task. Similar to the previous task, the threshold run starts with an Initial phase (IP). The masker level is set at a low intensity so that the detection of the probe is still relatively easy, and generates YES-responses. At the first reversal, the Threshold phase is initiated and the trial count starts. Again, a large step size is used in the Initial Phase and a smaller step size is used in the Threshold Phase. An example of the level track is shown in Figure 2.2-6.

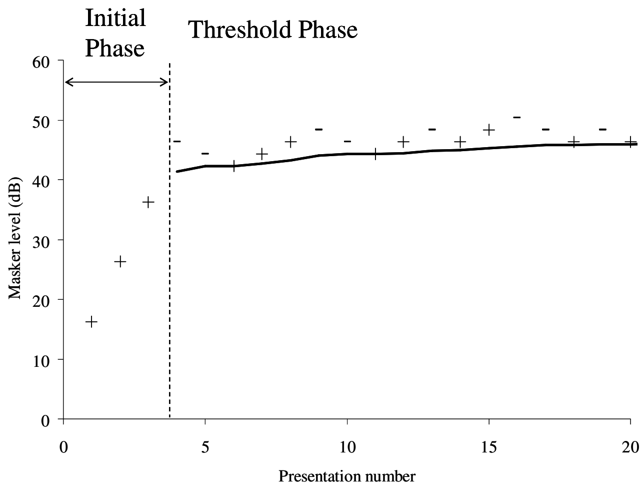


Figure 2.2-6. Illustration of a threshold run for detection of a tone in the presence of a forward masker when estimating the masker threshold. The masker level is variable. Plus-signs represent the YES-responses, Minus-signs represent the NO-responses. The full line represents the threshold estimate after each response. The first threshold estimate is calculated at the time of the first reversal. Large Initial step sizes are used in the Initial phase, smaller step sizes are used in the Threshold phase.

Threshold estimation

The threshold estimation method is similar to the method described for the detection-in-quiet-task. The threshold is estimated at the end of the threshold run by finding the logistic function that best describes the accumulated data in the Threshold phase.

The logistic function of the form

$$p = \frac{1}{1 + e^{-k\,(L-Th)}} \qquad (1)$$

is fitted to the YES/NO-responses from the Threshold phase. *P* is the proportion of YES-responses, *L* is the level of the presented stimulus, *k* is a slope parameter and *Th* is the estimated threshold (in dB SPL), i.e. the level of the stimulus where the proportion of YES-responses is 0.5.

The best-fit psychometric function is found using a least-squares method where *k* and *Th* are free parameters. The slope of the psychometric function is negative because lower masker levels are associated with a higher probability of a YES-response. An example of the resulting function when fit to data is given in Fig. 2.2-7. The Euclidian distance mentioned in this figure refers to the RMS error between the function and the data.

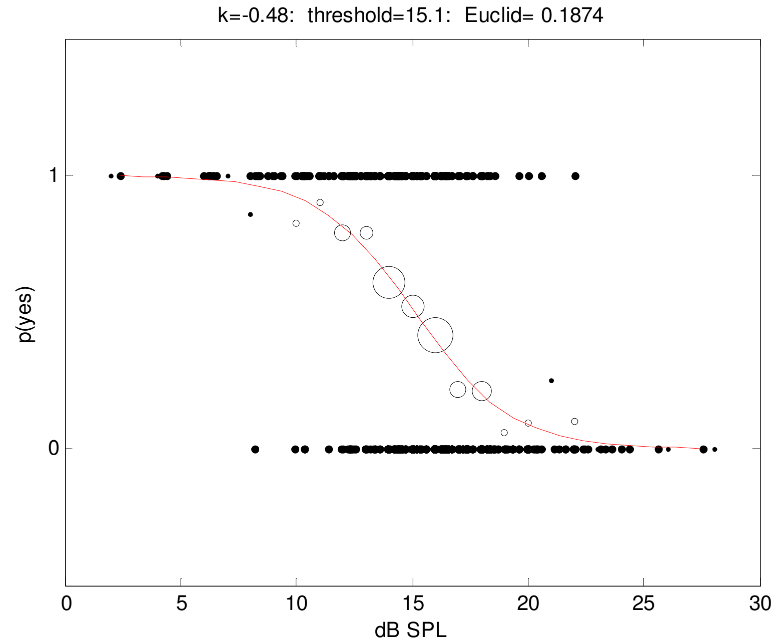


Figure 2.2-7. Illustration of psychometric function fitted to YES/NO-responses (black dots) in a detection task in the presence of a forward masker, when estimating masker threshold. The fitting was done using a least square best fit method allowing slope (k) and threshold (Th) to vary. The size of the open circles represents the number of responses associated with a specific probability. The 620 responses were generated using a computer simulation, assuming a threshold Th of 15 dB SPL and a slope parameter k of -0.5. The Euclidian distance refers to the RMS error between the function and the data.

Procedure

Stimulus. Similarly to the procedure used for the detection-in-quiet-task, a cue precedes the actual test stimulus. A schematic representation of this cued forward-masking stimulus is shown in Figure 2.2-8. The cue is similar to the test stimulus but the cue-masker is 10 dB (CueTestDifference) less intense than the test-masker. A lower cue-masker results in a cue-probe which is easier to detect and will therefore serve as a cue for the test-probe.

The listener is asked to count the number of probes heard, including the cue-probe. Similar to the detection-in-quiet-task the cue insures that the participant will usually have at least one probe to count. When the cue-probe and the test-probe are heard, the participant answers 2, which corresponds to a YES-response. When the test-probe is not heard, the participant will answer 1 or 0 (0 in the case that the cue-probe is not heard), and a NO-response is recorded. The purpose of the cue is to remind people what they are listening for. The cue-masker is generally below masker threshold, which means the cue-probe is still detectable.

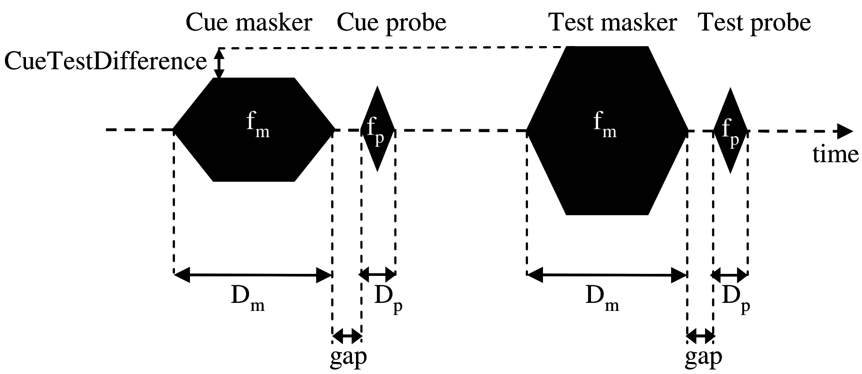


Figure 2.2-8. Schematic representation of a cued forward masking stimulus. The cue is identical to the test stimulus with respect to the durations (D_m and D_p) and frequencies (f_m and f_p) except for the cue masker which is less intense than the test masker (determined by the CueTestDifference).

As described in the detection-in-quiet-task, the listeners respond using a response box in front of them. A visual display shows a GUI (Graphical User Interface) depicting the response box. This GUI (Figure 2.2-9) is similar to the GUI used in the first task. While the stimulus is being presented, the selection buttons disappear from the GUI. The buttons re-appear at the end of the presentation interval to inform the participant a response is required. The listener subsequently responds by pressing the appropriate button on the response box.

Catch trials. In this forward-masking-task, catch trials were presented using the same procedure as in the detection-in-quiet-task.

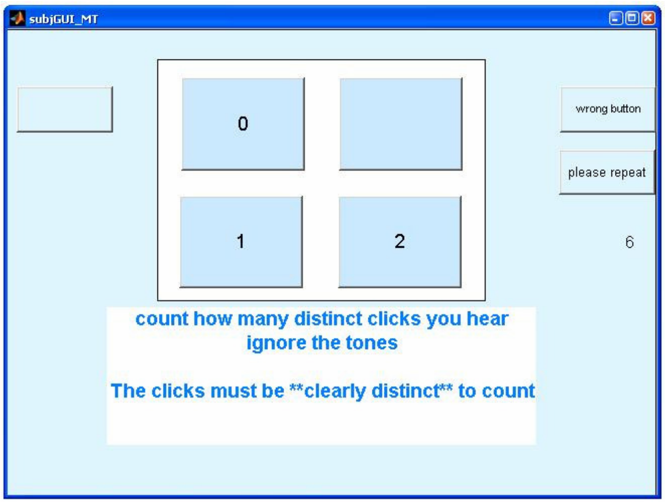


Figure 2.2-9. Graphic User Interface for participants in the forward-masking task.

2.3 Development of the Single-Interval-Up-Down method

The choice of an appropriate measurement procedure was guided by a number of considerations specific to the context of the project and data-collection. We wanted a procedure which would be suitable both in a laboratory and in a clinical setting. The main criteria that we aimed for throughout this development process were: user-friendliness, simplicity and robustness.

User-friendliness involves a number of issues. The first important issue is that the task which will be presented to the participants needs to be easy and allow for a simple explanation. Second, it is important that the actual measurements are fast and a threshold is obtained quickly. This is partly necessary because time is limited in a clinical setting but also because the task should not put too much demand on the listener's concentration. A second criterion is that of simplicity. The threshold estimation procedure must be simple, straightforward and transparent. Since the aim is to introduce the procedure in a clinical setting, it is important that clinicians can be easily introduced to the procedure and that they are able to familiarize with it quickly. Finally, an important consideration is the fact that the threshold estimates need to be reliable. The procedure needs to be robust and yield repeatable threshold estimates.

The Single-Interval-Up-Down method described in the previous section is the result of carefully considered amendments to other procedures. The method is based on Green's Maximum-likelihood method (1993) but was adjusted to simplify the procedure. Additional adjustments were implemented to improve the accuracy of the method. The adjustments were implemented over a 2-year period, as data-collection was on-going. The result is that data has been collected in the 3 stages of the modifications, thus using 3 slightly different threshold estimation procedures. The differences are not assumed to have a major impact on the threshold estimates.

In the next section, the decision process which preceded the development of the Single-Interval-Up-Down method will be described in the context of the criteria of user-friendliness, simplicity and robustness. Finally, it will be shown that threshold estimates remained stable despite the adjustments made.

2.3.1 User-friendliness

A first concern we dealt with was to make a procedure accessible for a broad public. Data collection in a laboratory setting will have different issues and complications than audiometric tests administered in a clinical setting. The main problem in a clinical setting is that you are dealing with a public which is new to the testing procedure. A simple and straightforward task is therefore important. A second issue which needs to be taken into consideration is the fact that only a limited time slot is allocated to the data collection, hence the need for a fast procedure.

Initially, it was decided that a single interval procedure, more specifically a Maximum-likelihood method (ML) proposed by Green (1993) might be more suitable for our specific aims. The motivation for this decision will be explained below.

The question about which task to use is directly related to how many intervals will be presented to the participant. Providing one stimulus interval simplifies the question you need to ask down to ‘Did you hear the target, yes or no?’ while presenting 2 or more intervals generates the question ‘In which interval did you hear the target?’. We considered a Single Interval yes-no paradigm, closely related to the task used in a clinical setting, against a two-interval two-alternative forced choice paradigm (2I2AFC), routinely used in a laboratory context.

A two-interval two-alternative forced choice paradigm (2I2AFC) is undoubtedly the most commonly used psychoacoustic paradigm for measuring thresholds in laboratory settings. It is often combined with Levitt’s transformed up-down method (1971) and widely accepted as a reference procedure when comparing different measurement procedures. In a 2I2AFC-paradigm, two or more temporal intervals are marked for the listener and the signal is presented during one of these 2 visual intervals. The listener has to indicate in which interval the signal is detected. The advantage of a forced choice up-down paradigm is its insensitivity to the listener’s response criterion. The disadvantages however are not to be taken lightly. In a forced choice paradigm the listeners are asked to select the interval where a target was perceived. Even when no target is perceived, the participant must still select an interval at random. In a clinical setting, patients are reluctant to guess and don’t feel comfortable with it. It is also likely to weaken the patient’s confidence in the measurement outcome. A second problem is related to the first one. Because people have to randomly select an interval when no target is being perceived, a sequence of coincidental correct guesses can cause a further decrease of the stimulus level below threshold.

The listener is now at risk of engaging in a random walk below threshold which reduces the accuracy of the threshold estimate and creates a *bias* towards lower thresholds (Amitay et al., 2006). Leek and colleagues (1992) used computer simulations to compare different adaptive staircase procedures and found that a 2AFC-procedure produced thresholds which were 2-3 dB lower than the real threshold. Similar findings are reported by Schlauch and Rose (1990). This bias can be avoided only by adding extra trials to a test run.

The use of Levitt’s transformed up-down method implies that a large number of responses are required before the threshold run is terminated and a threshold estimate is yielded. It has been shown that the number of trials in a 2I2AFC, up-down paradigm could be up to double the number of trials in a ML procedure, in order to obtain reliable threshold estimates. Leek and colleagues (2000) reported an average of 53 trials per threshold run in a 2I2AFC 2-down, 1-up procedure. Other researchers report similar findings (Amitay et al., 2006). This is a significant disadvantage when aiming for a fast threshold estimate.

Single-Interval yes-no procedures are less widely accepted compared to forced-choice procedures. In a yes-no procedure the participant is presented with a stimulus and is asked whether the stimulus was heard or not. The main advantage of yes-no procedures is that they are simple and fast. The simplicity lies in the fact that the listener has to indicate whether or not he/she heard the target. Listeners, and more specifically people in a clinical setting, are likely to be very much more at ease in performing this task since it is straightforward and easy to grasp. A commonly used yes-no procedure is Green’s Maximum Likelihood method (ML) (1993).

The significant advantage of this Maximum-likelihood adaptive procedure is that the stimulus level converges onto the threshold rapidly. Green (1993) suggested that threshold estimates could be collected in as little as 12 trials. Leek and colleagues (2000) determined that 24 trials was a sufficient number of trials. Other researchers have been found to report trial numbers below 24 trials (Shelton et al., 1982, Florentine et al., 2001).

A disadvantage of Green's ML method is that it is thought to be sensitive to the listeners' change in criterion, and to listeners' errors (Leek et al, 2000, Gu and Green, 1994). These issues are addressed by estimating how often the listener indicates a stimulus was observed when the stimulus was at a very low intensity level or absent (false alarm). This estimate is subsequently taken into account when finding the psychometric function representing the responses of the listener. However, this concern might be less pressing when the main purpose of the test is to obtain a threshold estimate. Several studies have compared threshold estimates obtained using a 2I2AFC, up-down method and a ML yes-no paradigm. It was shown that both procedures produce very similar threshold estimates and comparable variability for a range of detection tasks (Gu and Green, 1994; Leek et al., 2000, Baker and Rosen, 2001; Marvit et al., 2003; Buss et al., 2001; Shelton et al., 1982).

The fact that the Single interval yes-no procedures consist of a simple task and are considerably faster than multiple-interval forced-choice procedures and that threshold estimates are comparable to more traditional estimation procedures (such as 2I2AFC), convinced us that Green's ML method is the most suitable measurement procedure for our specific purposes.

To make the task even simpler for the listener, it was decided that a cue would be added to the stimulus. The cue reminds the listener what he is listening for and serves as some kind of anchor point which is the same distance from the test target in all participants. Florentine et al. (1999,

2000) suggested a similar cue in a yes-no procedure when trying to develop a clinically viable gap-detection measure.

The method as described above will be referred to as the Maximum-likelihood method.

2.3.2 Simplicity

The second issue we addressed was how to make the procedure as simple and transparent as possible so it would be easily accessible to clinicians. Two adjustments were made to Green's ML in order to fulfil this criterion. A first adjustment concerned estimating the false alarm rate, the second was related to the method for finding the best psychometric function.

One of the features which make ML somewhat cumbersome is the need to estimate the false alarm rate of the listener. A false alarm is a positive response at signals of very low intensity and it functions as an indicator of the change in a listener's criterion. It is generally accepted that estimating the false alarm rate is difficult and requires a large number of trials. Green (1993) reported that the false alarm rate was underestimated when using short threshold runs. He subsequently suggested using catch trials as a basis for the estimates of the false alarm rate (Gu and Green, 1994). Several researchers have adopted this method with satisfactory results (Leek et al., 2000, Dubno and Ahlstrom, 2001).

We propose an alternative use of catch trials. In order to control for criterion shifts, errors and guessing in the listener's performance, we use the catch trials simply to check on our participant. When a catch trial is being presented, and the listener indicates that he heard a target, he is informed that a mistake was made and the threshold run is aborted. Gu and Green (1994) compared thresholds obtained by adopting a liberal criterion with thresholds obtained with a conservative criterion. They showed that the threshold difference when fitting a logistic function was on average 2.2 dB. It is our opinion that this trade off is not problematic when collecting large amounts of data from single individuals. In our calculations of the psychometric function we therefore assume a false alarm rate of 0.

We decided to abandon a second feature of the ML method. This is related to the method used to find the best psychometric function. In the ML method a psychometric function is fitted to all available responses after each trial. This best psychometric function is subsequently used for both the selection of the next stimulus level as the estimate of the threshold. The psychometric function with the best fit has the highest probability of representing the 'true' psychometric function of the listener. It is selected from a set of candidate functions which vary in the stimulus levels associated with the threshold (50% point of the function, *Th*) but with a fixed slope (k = 0.5). After every trial the probability of the best psychometric function is multiplied with the cumulative probabilities of the best functions associated with the previous trials. This method of finding the best psychometric function is complicated, difficult to understand and must be computerized. It is not easily accessible to the wider/clinical public. Therefore we propose an adjusted fitting process. A psychometric logistic function will be fitted to all the responses by using *a least squares best fit* method allowing the slope (k) and the level of the threshold (*Th*) to vary. A least-squares method is a commonly used and understood statistical method among non-statisticians. This two-parameter fit provides us with a equally good representation of the psychometric function of the responses. The fact that the slope parameter k is allowed to vary will also provide us with the possibility of estimating the slope of the psychometric function when required.

2.3.3 Reliability

A number of studies showed that the Maximum Likelihood method is a reliable procedure with standard deviations similar to those found in a 2I2AFC, transformed up-down procedure (Leek et al., 2000). However it has been shown that ML is sensitive to errors early in the threshold run (Gu and Green, 1994). This sensitivity is closely related to the ability of ML to converge on the threshold region very quickly. In an adaptive track, as is ML, the level of the next stimulus is decided on the basis of the previous responses. This has as a direct consequence that the procedure converges on the threshold region very quickly. In our case, the mean (50% point) of the most recently estimated psychometric function is set as the intensity of the next stimulus. However, when an error occurs early in a threshold track (before the level has homed in on the desired region), this error weighs very strongly onto the selection of the stimulus and results in convergence on the wrong stimulus level region. When this situation occurs it is impossible to recover from the error. It requires a large number of trials to move closer to the desired intensity region (the ‘real’ threshold region) and this ‘real’ threshold region will never be sampled effectively. In a procedure where the stop criterion is based on a fixed number of trials this will result in an erroneous final threshold estimate.

An example will clarify what happens in this situation (see Figure 2.3-1). Let us assume we are trying to estimate the threshold for a tone in quiet for a particular listener. Let us further assume that the ‘real’ threshold is 25 dB SPL. A threshold track is started at a stimulus level randomly selected around 40 dB SPL, in this particular case 44 dB SPL. The initial step size is 10 dB SPL. The listener is presented with a stimulus at 40 dB and responds ‘YES, I heard it’ (plus signs). The next stimulus level is presented at 34 dB SPL. For an unspecified reason (maybe inattention), the listener indicates that he did not hear the stimulus (NO-response, minus sign), although it is clearly above threshold. The psychometric function fitted to this YES and no response will have a 50% point associated with a stimulus level of 39 dB SPL. Therefore the next stimulus will be presented at 39 dB SPL. The listener, now responding as expected, responds ‘YES’ and will do so in the consecutive trials. The intensity of the stimulus decreases with every successive trial since the 50% point of the psychometric function moves closer to the intensity of the NO-response. However the intensity will never be set below the intensity of the erroneous NO-response even in the situation where only YES-responses are given following the error. After a fixed number of trials the run is terminated and a final threshold estimate similar to the lowest intensity (here 34 dB SPL) is returned. This is a threshold estimate 9 dB higher than the ‘real’ threshold.

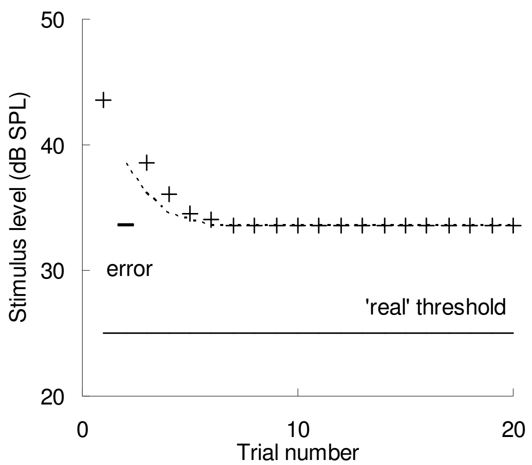


Figure 2.3-1. Illustration of response track when an error occurs early in threshold run. Plus-signs indicate YES-responses, minus-sign indicates NO-response. Dotted line is the threshold estimate after every trial based on all the responses collected until that point. The full line represents the ‘real’ threshold of the listener.

In order to avoid situations as described above, Leek et al. (2000) suggested closely monitoring the performance of the listeners and repeating threshold runs when necessary. Initially, a monitoring system was added to our existing procedure. It was based on the assumption that one would expect a mix of YES and NO-responses in a run where stimulus intensities are being sampled around the ‘true’ threshold. A series of only YES or only NO-responses is a strong indicator that the stimulus is being presented at an intensity level significantly away from the true threshold and will therefore not return a reliable threshold estimate. Specifically, when a threshold run reaches the end, a check is done on the last 80% of the total number of trials (not including trials in the Initial phase and catch trials). If all those responses are identical (either YES or NO), it is decided that this run is not reliable, the run is rejected and restarted. This monitoring system showed to be useful and a substantial amount of the data presented in this thesis was collected using this procedure.

The estimation method incorporating the adjustments described so far will be referred to as the 'Single Interval 50 %-method' (SI 50%)[1].

It was felt that the monitoring system came at an expense of frequent restarts of the threshold run. Furthermore, it did not meet our 'simplicity' requirement. So a different strategy was adopted for adapting the sequence of the stimulus levels. The stimulus selection is no longer based on the 50% point of the psychometric function but follows a simple one-up one-down staircase procedure. In a one-up one-down staircase the stimulus level is adjusted with a fixed step size after each response. A YES-response yields a decrease in the stimulus level, a NO-response results in an increase in the level. In our opinion this is a simpler and more straightforward way of selecting the stimulus, and is also closely related to the selection procedures currently used in audiological clinics. The psychometric function is no longer computed after each trial since the last fitted psychometric function is no longer needed for the selection of the next stimulus level. At the end of the threshold run, only one psychometric function is fitted to all the available responses and a threshold estimate is yielded.

[1] The choice of the term 'SI 50%' is based on the fact that the level of the next stimulus is determined by the 50% point of the last fitted psychometric function.

The staircase adaptive track makes the method simpler and solves the problem of getting stuck in an inappropriate intensity region. It also provides a better estimate of the slope of the psychometric function since the sloping part of the psychometric function is explored more broadly. Obviously there is a trade-off since a small number of extra trials is required to converge on the threshold region. However the time gained by eliminating the need for repeated measurement outweighs the cost of a limited number of extra trials. Introducing the staircase procedure as a method of selecting the level of the next stimulus was the final adjustment in our search for a simple, fast and reliable threshold estimation method.

We refer to this final version of the threshold estimation method as the ‘Single-Interval-Up-Down method’, which has already been described in section 2.2 of this chapter.

2.3.4 Summary of the adjustments

Table 2.3-1 shows a summary of the adjustments described above and how they relate to our goals of speed/ease of use, simplicity and reliability. The adjustments described in this section were implemented over a two-year period. The need for these improvements usually emerged during data collection. An unfortunate consequence of this fine-tuning of the estimation procedure is that data has been collected using two slightly different Single interval versions (ML and SI-50%) before settling for the Single-Interval-Up-Down procedure. However, these adjustments did not have a major effect on the threshold estimates. A short summarizing description of the three procedures is given below. This will be followed by a comparison of detection threshold of a normal listener estimated using the 3 different procedures

Table 2.3-1. Overview of the procedures considered for use in data collection. Areas in italics indicate the adjustment implemented in that specific procedure.

	Interval	**Level control/ Sequence strategy**	**Psychometric function Fitting procedure**	**False alarms**
2I2AFC++	2	2-down 1-up	Mean peaks and troughs	Compensated
Maximum Likelihood, no false alarms	*1 (+cue)*	*50 %*	*Maximum Likelihood*	*None allowed (catch trials)*
Single Interval 50 % (SI-50%)	1 (+cue)	50 %	*Least squares best fit*	None allowed (catch trials)
Single-Interval-Up-Down (SI UD)	1 (+cue)	*1-Up 1- Down*	Least squares best fit	None allowed (catch trials)
	SPEED COMFORT	**RELIABILITY**	**SIMPLICITY**	

Maximum Likelihood method

Task: Participants are asked to indicate whether they heard a target tone or not

Stimulus selection: Start level is set at a level where the participant can hear the target tone, which is a tone intensity of 30 or 40 dB SPL for detection of a tone in quiet, and a masker intensity of 10 dB SPL for detection of the probe in the presence of a variable forward masker. In this initial phase, the level of the stimulus is adjusted in step sizes of 10 dB SPL. For detection in quiet this means a decrease of the tone intensity. For detection of the probe in the forward masking task this entails an increase of the masker intensity. The ‘Initial phase’ ends at the first NO-response (participant can *not* hear the target). This first NO-response is considered as trial 1 and start of the ‘Threshold phase’. From that point onwards a psychometric function is fitted after each response and the intensity of the next stimulus is set at the 50 % point of the psychometric function. The first psychometric function is fitted using the first NO-response and the last YES-response of the Initial phase.

Threshold estimation: A psychometric function is fitted after each response in the Threshold phase using Green's Maximum Likelihood method (1993). A threshold estimate is subsequently produced after each trial corresponding to the 50 % point of the psychometric function. Each threshold run consists of 10 trials and the final threshold (in dB SPL) is set to the 50% point of the last psychometric function.

Catch trials: 20% of trials were catch trials with no test stimulus presented. If a 'YES' response was given during a catch trial, the run was aborted, the participant informed and the run was restarted. In a block of threshold runs, the catch trial rate is reduced gradually after each flawless run to no less than 10 %. Catch trials are not included in the total trial count.

Participants are encouraged to be conservative in their judgments and not indicate the presence of a sound if they are uncertain.

Single Interval 50 % - method

This procedure is the same as the Maximum Likelihood method except for one thing: the psychometric function is estimated using a least squares best fit procedure.

Task: same as Maximum Likelihood Method

Stimulus selection: same as Maximum Likelihood Method

Threshold estimation: A psychometric function is fitted after each response in the Threshold phase using a Least Squares Best Fit procedure. A threshold estimate is subsequently produced after each trial corresponding to the 50 % point of the psychometric function. Each threshold run consists of 10 trials and the final threshold (in dB SPL) is set at the 50% point of the last psychometric function.

Catch trials: same as Maximum Likelihood Method

Single-Interval-Up-Down method

This procedure is the same as the Maximum Likelihood method and the Single Interval LS 50 % method described above except for the stimulus selection and the final threshold estimate.

Task: same as Maximum Likelihood Method

Stimulus selection: The start intensity of the run is set at supra-threshold level (resulting in a YES-response). This is tone intensity of 30 or 40 dB SPL for detection of a tone in quiet, and a masker intensity of 10 dB SPL for detection of the probe in the presence of a variable forward masker. If the response is 'YES' the stimulus level is adjusted by a fixed step size; if the response is 'NO' the level is adjusted by that same step size in the opposite direction. This means that for a detection-in-quiet-task a YES results in a *decrease of the tone intensity*. For detection of the probe in the forward masking task a YES-response entails an *increase of the masker intensity*. In the Initial phase, the level of the stimulus is adjusted in step sizes of 10 dB SPL. The Initial phase ends at the first NO-response (participant can *not* hear the target). This first NO-response is considered as trial 1 and start of the 'Threshold phase'. In the Threshold phase stimulus level is adjusted in smaller step sizes of 2 dB.

Threshold estimation: Each threshold run consists of 10 trials. A single psychometric function is fitted at the end of the threshold run using a Least Squares Best Fit procedure. The threshold estimate (in dB SPL) is set at the 50% point of the psychometric function.

Catch trials: same as Maximum Likelihood Method

Comparison of the 3 Single-interval methods

A short experiment was conducted to confirm that the procedural adjustments implemented during the data collection (and resulting in the 3 Single interval methods described above) did not greatly affect the threshold estimates.

Method

Detection thresholds in quiet were measured in a normal listener using the 3 Single interval methods described above (Maximum-likelihood, Single-Interval 50 % and Single-Interval-Up-Down). The stimulus was a 100-ms, 2000-Hz pure tone. Five threshold estimates were made per method. A threshold run consisted of 10 trials in each condition (not including trials in the Initial phase and catch trials).

Results

Figure 2.3-2 shows the average threshold for each method. The error bars represent 1 standard error. It can be concluded that the threshold estimates are very similar for all procedures. The average thresholds vary between -3.8 dB SPL and -4.9 dB SPL. There is some variability in the standard deviations associated with each procedure. The standard deviations are 4.8, 1.1 and 3.0 dB SPL for ML, SI-50 % and SI UD respectively.

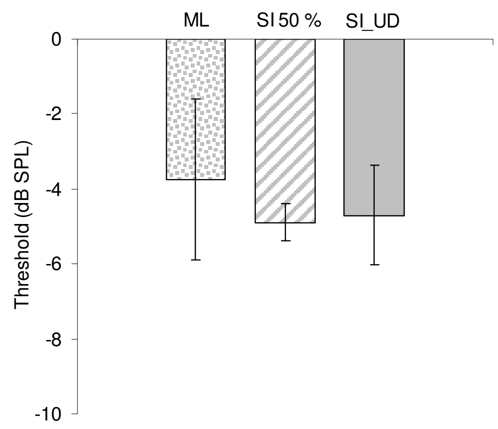


Figure 2.3-2. Threshold estimates for a 2000 Hz, 0.1 s pure tone in a normal listener, obtained using Maximum-likelihood (ML), Single Interval 50 % (SI-50%) and Single-Interval-Up-Down (SI-UD). The thresholds are the average of 5 measurements. Error bars represent 1 standard error.

The SI 50 % method shows a variability which is considerably lower than the variability found in the Maximum-likelihood method and the Single-Interval-Up-Down method. This can possibly be explained by the fact that the listener was a well trained subject. As mentioned before the SI 50 % method selects the level of the next stimulus based on the 50% point of the psychometric function. This method is very reliable when no errors occur, which was the case for the participant in this experiment. When errors do occur however, the threshold estimate is severely affected. The Single-Interval-Up-Down method is less vulnerable to these errors. The fact that fixed step sizes are used to determine the level of the stimulus does add some variability to the threshold estimates, resulting in standard deviations higher than the SI-50% method. Nevertheless, the SI-UD procedure is the preferred method to be used in this project since it is the simpler and more robust method.

2.4 Conclusion

This chapter is concerned with the development of a simple, fast and reliable threshold estimation procedure. Three important criteria were considered when selecting the estimation method. The first criterion is user friendliness. This means that the task must be easy and the thresholds must be collected in a short amount of time. The second criterion is robustness, meaning that the threshold estimates must be reliable. Finally, it is important that the estimation procedure is simple and straightforward.

The Single-Interval-Up-Down procedure was presented and the development of this procedure was discussed. The adjustments made to meet the 3 criteria were explained in detail, highlighting the issues surrounding single interval procedures.

CHAPTER 3

Predicting the accuracy in a Single Interval Procedure

3.1 Introduction

In this project we are confronted with the problem of obtaining fast but reliable threshold estimates. If the emphasis of the experiment lies on obtaining very precise measurements, a large number of responses, and therefore trials, will be required. When the accuracy of the estimate is less crucial, one can consider eliminating trials in order to speed up the data collection. Obviously, fewer trials will result in a reduced accuracy of the threshold estimate. An important question to ask is ‘What is the smallest number of trials which will still guarantee a given level of reliability?’ Green (1993) pointed out that the standard deviation of the threshold estimate (its unreliability) is related to the reciprocal of the number of trials. In other words, the standard deviation of a threshold estimate will decrease with an increasing number of trials. Pentland (1980) compared the improvement in accuracy as a function of trial number for a number of procedures (standard staircase, PEST and Best PEST). In all procedures, the accuracy improved as the number of trials was increased. A significant difference in the overall accuracy between the procedures was found, with the Best PEST procedure yielding the lowest standard deviations.

One of the advantages of a Single Interval procedure is the possibility of ending a threshold run at any arbitrary point. This is possible because a ‘best’ psychometric function can be found after each trial to provide a threshold estimate. It is therefore perfectly plausible to set the stop rule of the procedure at a fixed number of trials. In a 2I2AFC procedure this is usually not the case and a fixed number of reversals is required before a threshold run ends. These reversals are more widely spaced with an unspecified number of trials between reversals, yielding variable trial lengths.

In this chapter, the Single-Interval-Up-Down procedure (described in Chapter 2) will be used to demonstrate how the accuracy of a threshold estimate changes with the number of trials used in a threshold run for human listeners. Subsequently a formula predicting the number of trials needed for a required level of accuracy will be presented. Computer simulations and additional human data will be used to evaluate this predictive formula. Finally the formula will be applied to specify the number of trials which will be used when making the measurements needed for the development of the hearing profiles of normal and impaired listeners, presented further in this thesis.

3.2 Accuracy as a function of number of trials

3.2.1 Human listener

The variability of the threshold estimates was explored by repeatedly measuring the same threshold for a tone in a human listener.

Method

Thresholds were measured for a 2000-Hz tone with a duration of 100 ms using the Single-Interval-Up-Down procedure. The threshold measurement was repeated 20 times and each threshold run continued for 30 trials. The Initial step size was set at 10 dB SPL, the step size was 2 dB SPL. Threshold estimates were made after each trial. The accuracy of the estimate was assessed in terms of the standard deviations of the threshold estimates, Th, after each trial across the 20 runs. A smaller standard deviation of the threshold estimates should be interpreted as a better accuracy.

Results

The standard deviation of the threshold estimates is used to specify the accuracy of the procedure and is referred to as the 'error'. Within each run, the threshold was estimated after each trial so that the error could be measured as a function of the trial number. This procedure is illustrated in Figure 3.2-1 where each diamond indicates the standard deviation of the threshold estimate after an increasing number of trials for a human listener (CM). As the run proceeds, the accuracy of the estimates increases and this is reflected in a reduction of the standard deviation.

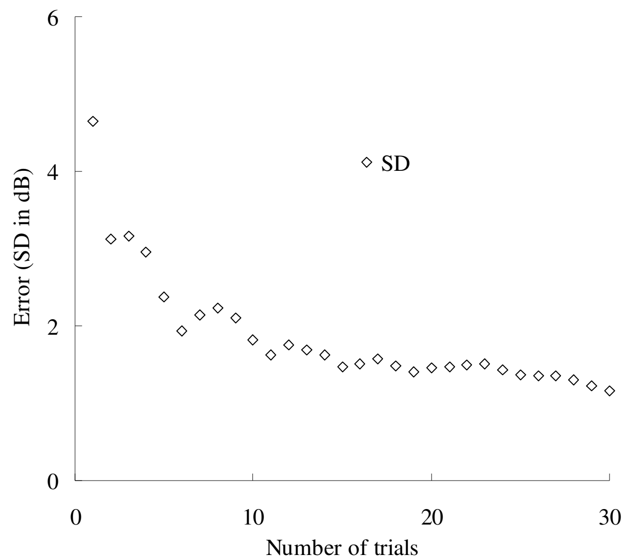


Figure 3.2-1. Standard deviation of threshold estimates (in dB) as a function of the number of trials for listener CM (open diamonds). Each data point is based on 20 threshold runs.

An important predictor of the error is the width of the psychometric slope of the listener. In principle, it should be possible to predict the accuracy of the threshold estimate if we know the width of the psychometric slope. This can be estimated from the data by fitting a psychometric function to the responses of the listener.

Estimating the psychometric slope

The Single Interval Procedure is based on the assumption that a listener's responses to a tone are based on an underlying psychometric function. In our case this is a logistic function of the form

$$p = \frac{1}{1 + e^{-k\,(L-Th)}} \qquad (1)$$

The function is fitted to the YES/NO- responses recorded from the human listener CM, where *p* is the proportion of YES-responses, *L* is the level of the presented stimulus, *k* is a slope parameter and *Th* is the estimated threshold, i.e. the level of the stimulus when the proportion of detections is estimated by the function to be 0.5.

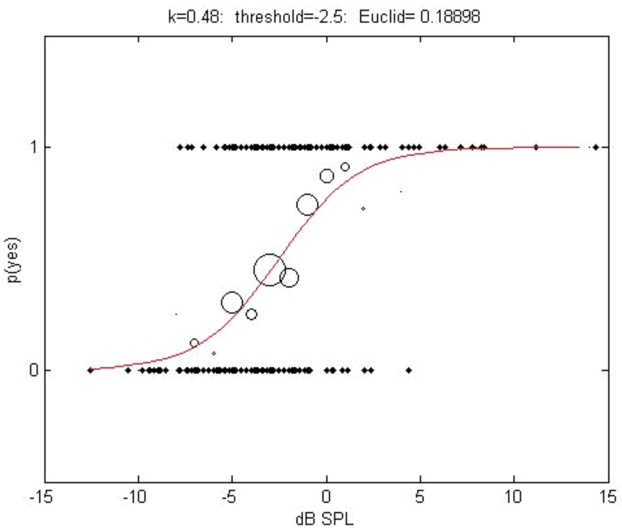
The two important parameters in a psychometric function are the *mean (Th)* of the psychometric function which gives us the detection threshold and *the slope (k)*. The slope of the psychometric function have a strong influence on the variability of this threshold estimate but does not affect the threshold estimate (Green, 1993). A shallow slope (small k) will yield more variable thresholds than a steep slope (large k).

Since we are looking at the variability of the threshold estimates, the slope is an important factor we need to take into account. Green (1993) suggested that a slope of 0.5 is a good approximation of the slope of a human psychometric function when measuring absolute thresholds. The psychometric function of the human listener was subsequently estimated to determine the slope.

Method. The psychometric function of the human listener was estimated using the 620 YES/NO-responses obtained in the 20 threshold runs presented above. The logistic function represented by equation (1) was fitted to the responses using a least-squares method where k and Th are free parameters.

Results. The responses used to estimate the slope of the psychometric function are displayed as dots in Fig. 3.2-2. The size of the circles is used to indicate the relative number of responses at the corresponding level. The up/down tracking procedure generates more stimuli in the levels close to the mean of the function which is reflected in the larger circles around the mean of the function. The best fit function is shown as the continuous line through the data points.

The best fit function has a threshold (*Th*) of -2.5 dB SPL and a slope parameter k of 0.48. This finding is in line with Green's assumption that the slope parameter for human psychometric functions in a detection task is 0.5.



Threshold (*Th*) = - 2.5 dB SPL

Slope parameter k = 0.48

Figure 3.2-2. Psychometric function for listener CM. The fit of the function is based on the 620 YES/NO-responses obtained in the 20 threshold runs presented above. The threshold (*Th*) is the stimulus level associated with the 0.5-point of the psychometric function. The slope parameter k represents the width of the function. The Euclidian distance refers to the RMS error between the function and the data.

3.2.2 Computer simulations

Subsequently, Monte Carlo simulations were used to further investigate how the accuracy of the threshold estimates changes with an increasing number of trials.

Method

The participant's response was simulated using an assumed logistic function with parameter Th always fixed at 15 dB SPL and a slope value k of 0.5. The choice of the slope parameter was based on the slope parameter found in the human listener discussed in the previous section. For each trial, the stimulus level, L, was used in equation (1) to generate the probability p that a YES-response had occurred. A random number generator was used to generate a value between 0 and 1. If the random number was less than p, a YES-response was judged to have occurred on that trial otherwise a NO-response was assumed. Cue tones and catch trials do not feature in the computer simulations.

Simulation thresholds were measured using the Single-Interval-Up-Down procedure. The initial step size was set at 10 dB and the step size was set at 2 dB. The simulation consisted of 100 runs. Each run continued for 50 trials. Threshold estimates were made after each trial. The accuracy of the estimate was assessed in terms of the standard deviations of the threshold estimates, Th, after each trial across the 100 runs. A smaller standard deviation of the threshold estimates should be interpreted as a better accuracy.

Results

The individual data points in Fig. 3.2-3 show how the accuracy increases as the trials progress. In other words the standard deviation of the estimate decreases with increasing trial numbers. This result is in line with the accuracy found in the normal listener presented above.

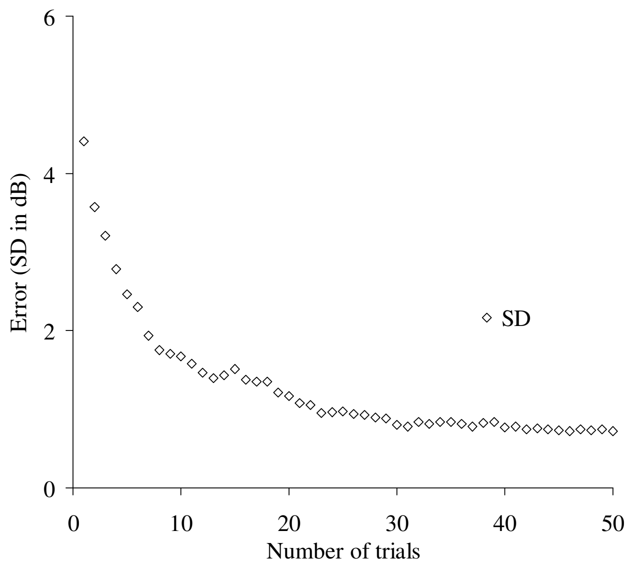


Figure 3.2-3. Standard deviation of threshold estimates as a function of the number of trials based on computer simulations (open diamonds), assuming a psychometric slope of 0.5.

This computer simulation can now be used to evaluate a statistical formula which will enable us to predict the standard deviation associated with a specific number of trials.

3.3 Predicting variability

3.3.1 Predictive formula

The variability of the threshold estimates is related to the variability of the assumed underlying psychometric function (Chapter 2). The assumed psychometric function is a logistic function of the form

$$p = \frac{1}{1 + e^{-k\,(L - Th)}} \qquad (1)$$

where P is the proportion of 'YES' responses, L is the level of the presented stimulus, k is a slope parameter and Th is the estimated threshold (in dB SPL), i.e. the level of the stimulus at which the proportion of YES-responses is 0.5.

The standard equation for estimating the variance of the probability density function of the logistic distribution (Hastings and Peacock, 1975) is

$$\text{variance}_{\text{logistic}} = \frac{\pi^2}{3 * k^2} \qquad (2)$$

The variance of the mean is related to the variance of the logistic function and to the reciprocal of the number of trials. The standard deviation of the mean is therefore determined by the equation

$$SD_{\text{mean}} = \sqrt{\frac{\text{variance}_{\text{logistic}}}{\text{number of trials}}} \qquad (3)$$

Based on equation (2) and (3), the increase in accuracy as a function of the number of trials can be predicted using the equation:

$$SD_{\text{mean}} = \frac{\pi}{\sqrt{3} * k * \sqrt{\text{no of trials}}} \qquad (4)$$

where SD_{mean} is the standard deviation of threshold estimates (error) and k is the slope of the psychometric function.

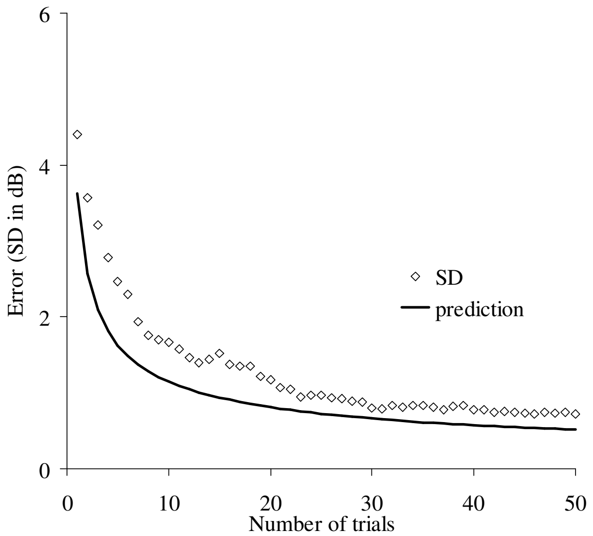
This predictive formula (4) will now be evaluated on the basis of the simulated data.

3.3.2 Does the function fit?

Function fit to simulated data

A prediction of the standard deviations of the threshold estimates based on formula (4) was compared to the standard deviations obtained in the computer simulations presented in section 3.2.2. The slope parameter *k* was set at 0.5 in equation (4) to match the slope parameter used in the computer simulation.

Figure 3.3-1 shows the standard deviations obtained in the computer simulations (open diamonds) and the predicted standard deviations assuming a slope parameter *k* of 0.5 (full line). The predictive formula produces an *underestimate* of the variability of the threshold estimates. The additional variance produced in the computer simulations is thought to originate from the granularity (finite step size) of the measurement procedure.



$$SD_{mean} = \frac{\pi}{\sqrt{3} * k * \sqrt{\text{no of trials}}}$$

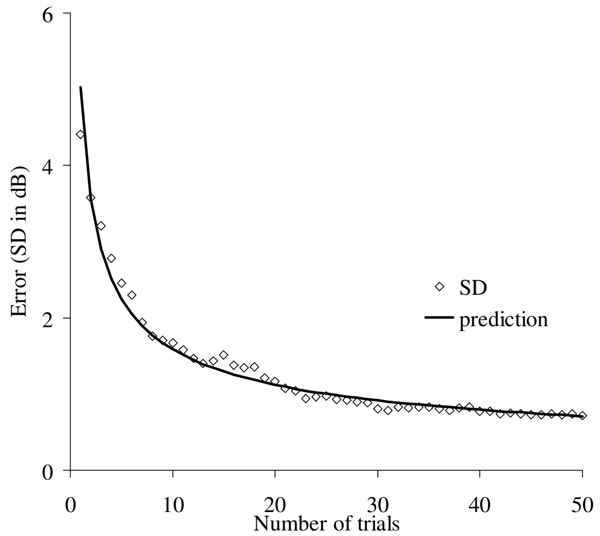
Figure 3.3-1. Standard deviation of threshold estimates as a function of number of trials based on computer simulations assuming a psychometric slope *k* of 0.5 (open diamonds). Full lines represent the predicted standard deviations equation (4) with $k = 0.5$.

To accommodate this additional variance, a scalar *j* is introduced to equation (4). The equation is now of the form

$$SD_{mean} = \frac{j * \pi}{\sqrt{3} * k * \sqrt{\text{no of trials}}} \qquad (5)$$

The predictive function (5) is again fitted to the simulated data using a least-squares best fit procedure with *j* as a free parameter. A *j*-parameter larger than 1 results in a vertical, upward shift of the predictive function.

Figure 3.3-2 shows the standard deviations obtained in the computer simulations (open diamonds) and the predicted standard deviations assuming a slope parameter *k* of 0.5 (full line). A *j*-value of 1.4 was found to produce the best fit to the data. The prediction fits the simulation data very well (RMSlog = 0.04 dB).



$$SD_{mean} = \frac{j * \pi}{\sqrt{3} * k * \sqrt{\text{no of trials}}}$$

with $j = 1.4$

Figure 3.3-2. Standard deviation of threshold estimates as a function of number of trials based on computer simulations assuming a psychometric slope *k* of 0.5 (open diamonds). Full lines represent the predicted standard deviations using formula (5) with $k = 0.5$ and j = 1.4.

Function fit to human data

Formula (5) has shown to provide a satisfactory prediction of the simulated standard deviations when a *j*-value of 1.4 is used. The accuracy-data found in the human listener CM (section 3.2.1) is similarly fitted with the equation (5) assuming a slope *k* of 0.48 (as established above) and adopting the *j*-value of 1.4 established for the prediction of the simulated standard deviations.

Figure 3.3-3 shows the standard deviations as a function of the number of trials for the human listener (open diamonds) and the predicted standard deviations (full line) using the true slope of the listener ($k = 0.48$) and assuming a *j* of 1.4.

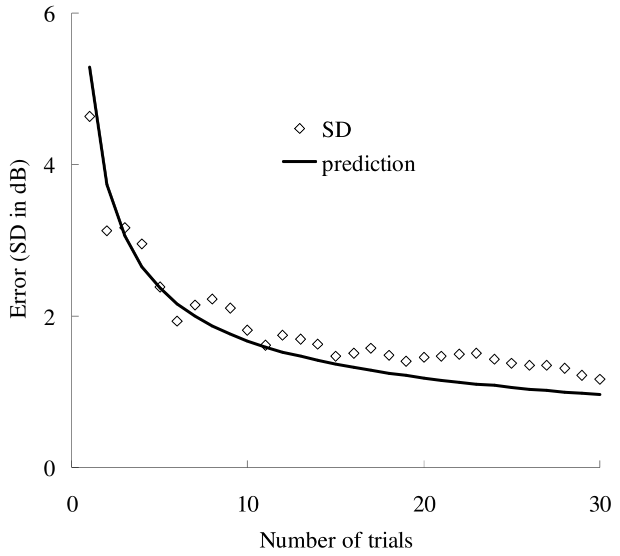


Figure 3.3-3. Standard deviation of threshold estimates as a function of the number of trials for listener CM (open diamonds). Full line represents the prediction of the standard deviations using the true slope value (k=0.48) of the listener and a j-value of 1.4.

A good fit was found between the predicted standard deviations and the observed standard deviations (RMSlog = 0.08 dB). The observed standard deviations are larger than the predicted error for increasing trial numbers. This larger error can be explained in terms of factors such as fatigue, lapses of attention, etc. Any extra source of variance must result in an *addition* to the values predicted by the formula. On this basis we would expect human variance to be larger than the predicted variance.

The fit is nevertheless successful and shows that the formula can be used to predict the accuracy of the estimation procedure for a given number of trials.

3.3.3 Accuracy as a function of psychometric slope

The statistical formula showed to successfully predict the accuracy in human listeners. The prediction however is a function of the slope of the psychometric function. This psychometric slope may vary from participant to participant. This would have a strong influence on the accuracy of the procedure. Green (1993) suggested that a slope of 0.5 is a good approximation of the slope of a human psychometric function in this detection task His suggestion is based on data by Watson et al. (1972) who found that a psychometric function with a 8 dB-width was a good representation of a human psychometric function in a detection task. If we take 'width' to refer to the difference in level between the 5% and the 95% point of the logistic function, 8 dB is the expected width of a logistic function where k = 0.5. The slope of the psychometric function represents a range of stimulus levels at which the listener is uncertain whether he heard the stimulus or not. A shallow slope represents a wide range of levels at which uncertainty occurs, a steep slope will only consist of a limited range of levels.

Computer simulations were used to explore the effect of the slope on the accuracy.

Method

Additional computer simulations were run for 2 more slope parameters, k, of 0.3 and 0.7. The simulation procedure is similar to the procedure used for k = 0.5. The calculation of standard deviations was similar to the previous experiments. The data representing standard deviations for k = 0.5 is the same data as presented in Figure 3.2-3. K-values *lower than 0.5* will represent a psychometric slope which is *shallower* than a function with a k-value of 0.5. Similarly, a k-value *higher than 0.5* will represent a *steeper* slope.

Results

The individual data points in Figure 3.3-4 show simulated standard deviations as a function of the number of trials assuming a slope parameter k = 0.5 (open diamonds), k = 0.3 (open triangles), k = 0.7 (crosses).

The standard deviation of the threshold estimates increases when the psychometric slopes becomes shallower (lower k). This is because the range of 'uncertain' stimulus levels is wider and will therefore yield threshold estimates with a larger variability.

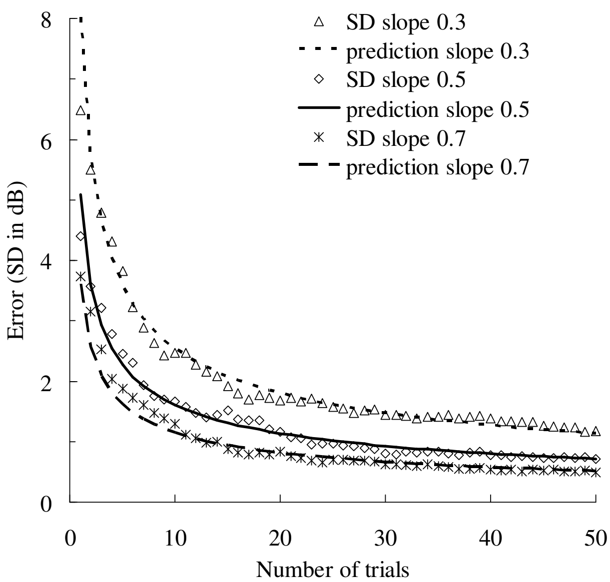


Figure 3.3-4. Standard deviation of threshold estimates (in dB) as a function of the number of trials assuming different slopes of the psychometric function. The symbols represent data points generated by computer simulation assuming a slope parameter $k = 0.3$ (open triangles), 0.5 (open diamonds) and 0.7 (crosses). The straight lines are the predicted functions obtained using equation (4) and a j of respectively 1.3, 1.4 and 1.4 .

The lines in Figure 3.3-4 are least-squares best fit applications of equation (5) to the simulated data using the respective slope parameter *k* and by allowing *j* as a free parameter. The predictive functions is shown to fit the data well (RMSlog = 0.03, 0.04, 0.03 dB for slope condition 0.3, 0.5, 0.7 respectively). The *j*–values found are 1.3, 1.4 and 1.4 for k = 0.3, 0.5, and 0.7 respectively.

The j-values found in the 3 different slope conditions are all very similar. It is therefore concluded that assuming a *j*-value of 1.4 is appropriate when predicting the standard deviation of threshold estimates in listeners.

3.4 Predicting the accuracy in human listeners

3.4.1 Accuracy in normal listeners

The accuracy in human listeners was explored more extensively using the same procedure as for the accuracy of the human listener reported above (see section 3.2).

Method

Thresholds were measured for a 2000-Hz tone with a duration of 100 ms using the Single-Interval-Up-Down procedure. A 1000-Hz tone was used for 1 listener (RM). The threshold measurement was repeated 20 times and each threshold run consisted of 30 trials. The data was collected in a single session with a two-minute break after every third run. The Initial step size was set at 10 dB, the step size was set at 2 dB.

The participants were four listener aged between 21 and 64 years. All listeners had audiometric thresholds below 20 dB HL for the test frequency. Listener RM has a mild to moderate hearing loss from 2000 Hz onwards. Listener CM (reported in section 3.2) is one of the four listeners and his data is the same data as presented in section 3.2.

Results

Figure 3.4-1 shows the accuracy as a function of trial number in the 4 listeners. Similar to standard deviations presented in previous sections, all listeners show a decrease in the standard deviation of the threshold estimates as a function of number of trials. The rate of decrease is similar in all listeners.

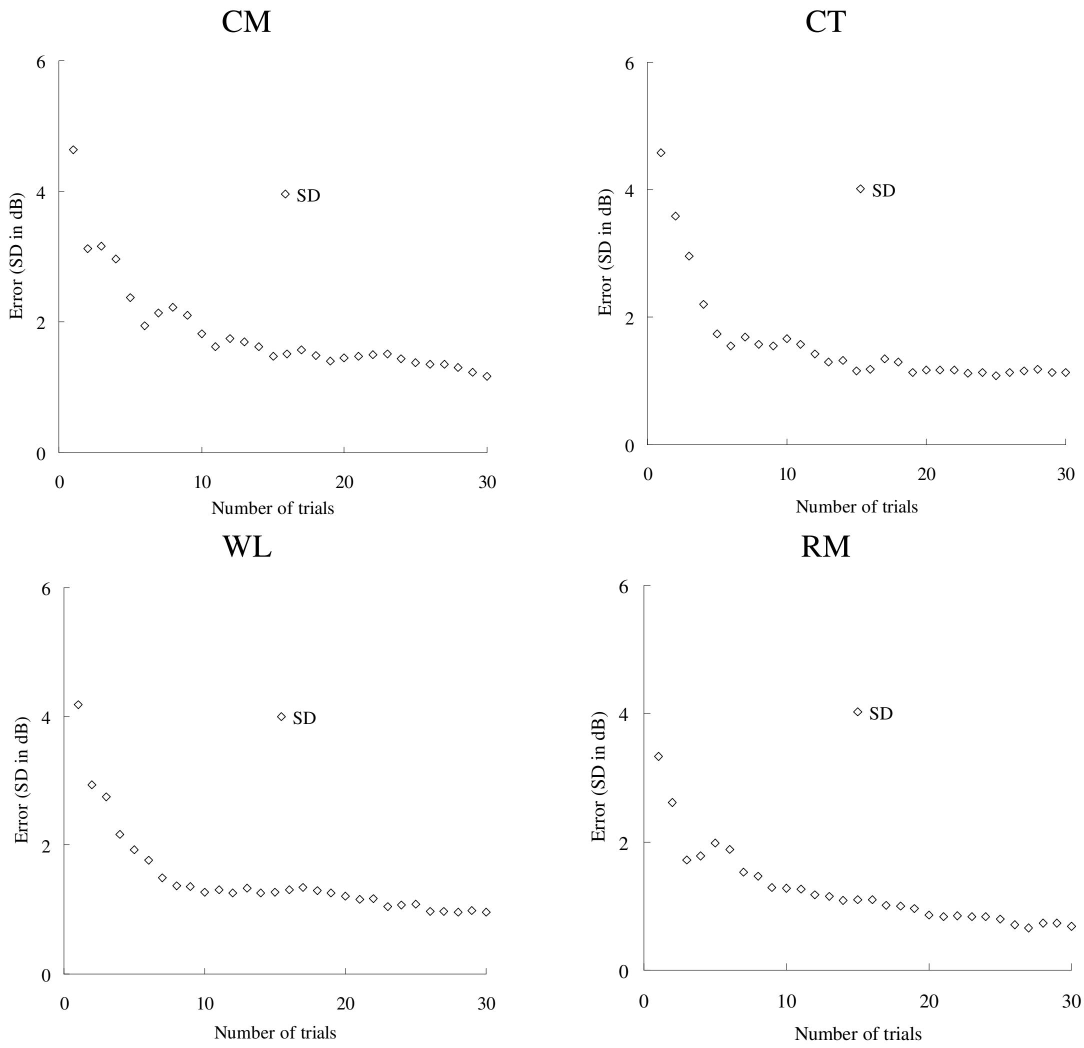


Figure 3.4-1. Standard deviations of threshold estimates (in dB) as a function of number of trials for normal listeners. Frequency of the tone was 2000 Hz for participant CM, CT, WL and 1000 Hz for RM.

3.4.2 Psychometric slopes in normal listeners

A precise prediction of the standard deviation in human listeners requires an exploration of the psychometric slopes. The psychometric slopes were estimated using the same method as described for the estimation of slope of the human listener CM (see section 3.2).

Method

The psychometric function of the human listener was estimated using the responses obtained in the 20 threshold runs presented above. The data used to obtain the slope of the psychometric function are displayed as dots in Figure 3.4-2. The relative size of the circles is used to indicate the relative number of responses at the corresponding level. The slope is found by fitting a logistic function to the 620 YES/NO-responses using a least squares best-fit procedure. The best fit function is shown as the continuous line through the data points. For every listener the *k*-value, threshold *Th* and the RMS (Euclid) are shown above the psychometric function.

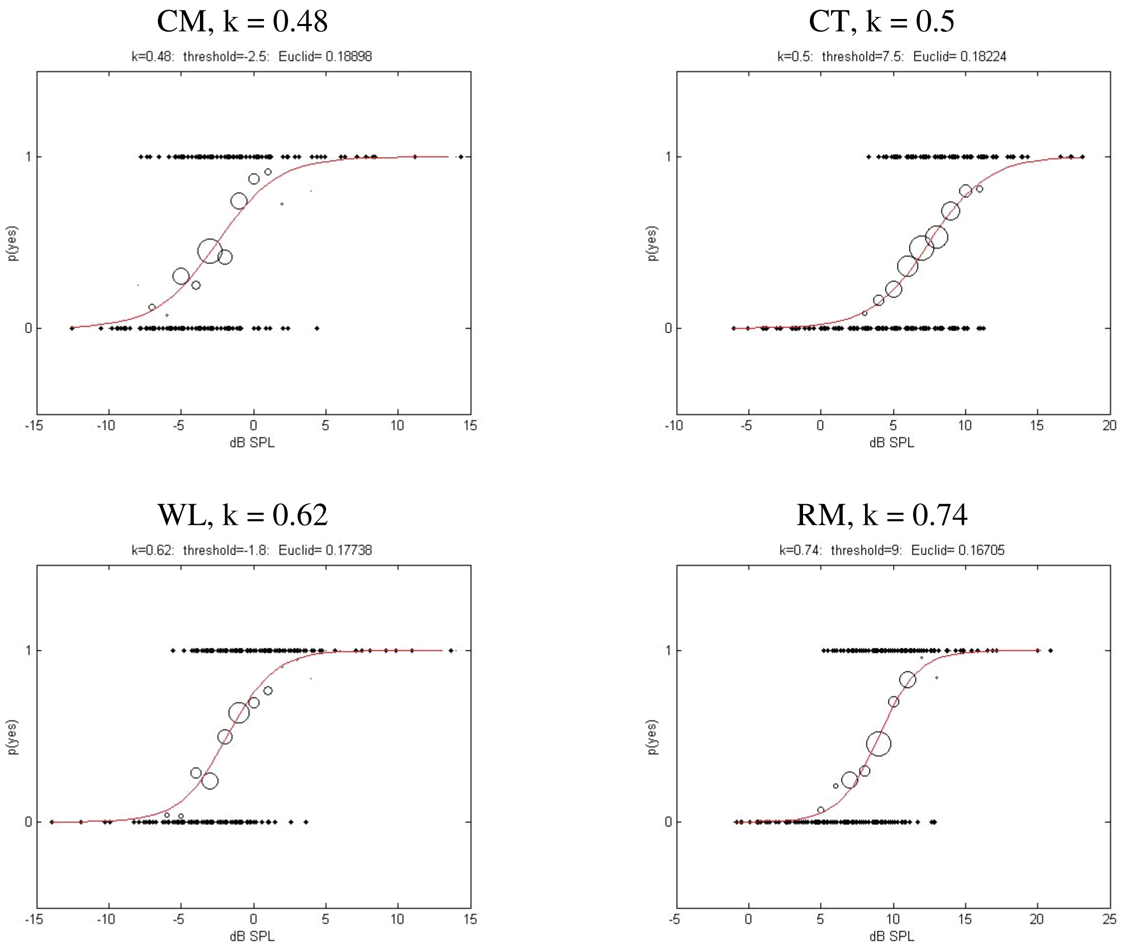


Figure 3.4-2. Psychometric function of 6 human listeners based on 620 responses. The size of the circles represents the number of responses contributing to that point of the psychometric function. Frequency of the tone was 2000 Hz for participant CM, CT, WL and 1000 Hz for RM.

Results

The slope estimates range from 0.48 for listener CM to 0.74 for listener RM. In two normal listeners a slope parameter k of 0.5 is found which is in agreement with slope parameters suggested by Green (1993). The slope found in listener RM is steep compared to the slopes of the other normal listeners. This listener has a mild to moderate hearing loss from 2000 Hz onwards. It could be speculated that the hearing loss at the neighbouring frequencies affects the measure of the psychometric slope and that listener RM probably does not truly fit in the 'normal' category. The difference in test frequency is not thought to be a contributing factor to the steeper slope of listener RM. Watson and colleagues (1972) suggested that the slope of the psychometric function changes as a function of the stimulus frequency and found shallower slopes at lower frequencies. This relation is in the opposite direction of the one observed in listener RM where a steeper slope is found for a test stimulus at a lower frequency. It is therefore concluded that the different stimulus frequency used when RM was tested can not explain the steep slope found in this listener.

The slope parameters found in the 4 listeners suggest that using a slope parameter k of 0.5 is a good basis for predicting the accuracy of the threshold estimates.

3.4.3 Predicting the accuracy using the true slope of the listener

The variability of the threshold estimates was estimated using equation (5). The true slope of each listener was used along with a j-value of 1.4. The comparison between the real and the predicted standard deviations can be seen in Figure 3.4-3. The predictive function fits the data very well. This shows that the formula is a good predictor of the accuracy of the threshold estimates for a given number of trials, when the true slope of the listener is used.

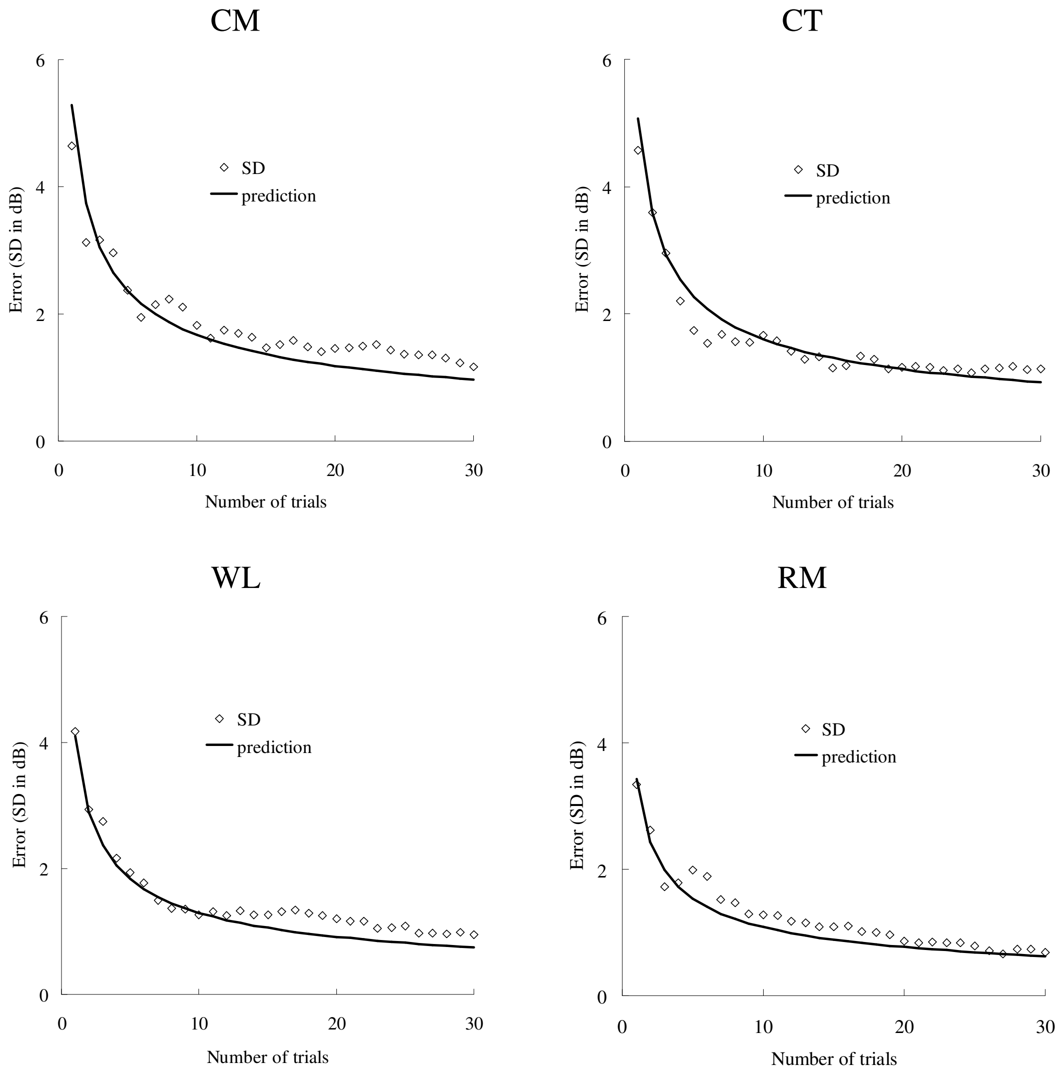


Figure 3.4-3. Standard deviation of threshold estimates as a function of the number of trials for 4 normal listeners (open diamonds). Full line represents the prediction of the standard deviations using the true slope value of the listener (see Figure 3.4-2) and a *j*-value of 1.4.

3.4.4 Predicting the accuracy with a generic slope of 0.5

In the previous section standard deviations in human listeners were predicted using the true slope of the listener which was obtained by generating the psychometric function of the listener. Unfortunately, in a clinical setting, time to measure the psychometric slope will not be available. However, a standard set of values for *k* and *j* will allow us to approximate the accuracy found for a given number of trials. The human data is subsequently used to evaluate the predictive equation when a standard value for the slope parameter *k* and scalar *j* is assumed.

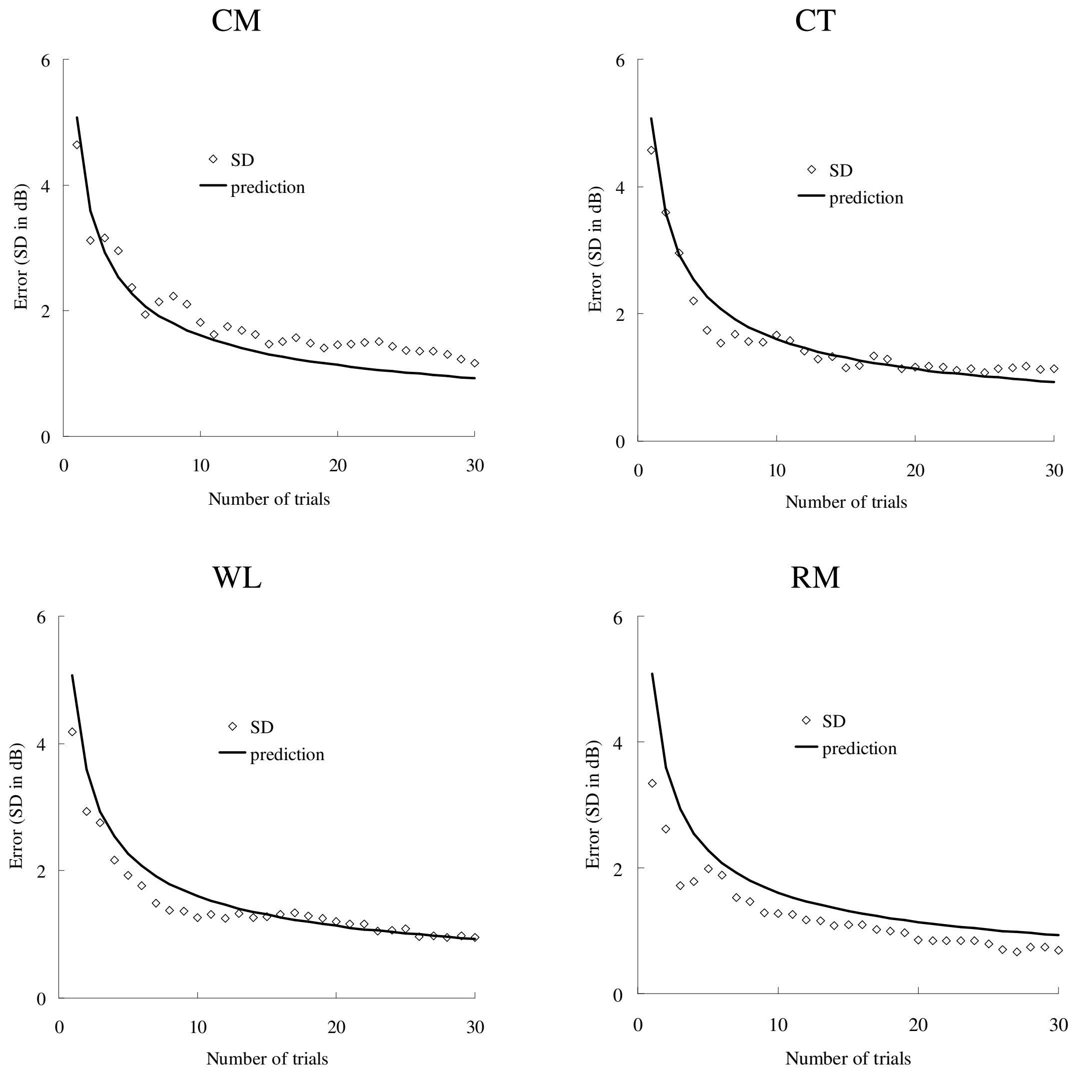


Figure 3.4-4. Standard deviation of threshold estimates as a function of the number of trials for 4 normal listeners (open diamonds). Full line represents the prediction of the standard deviations using the a generic slope value k = 0.5 and a j-value of 1.4.

Figure 3.4-4 shows the predicted accuracy for the human listeners when assuming a slope parameter k = 0.5 and a j-value of 1.4. The predictive function fits the data well in all listeners. For listener RM, the prediction overestimates the actual variability. This is because the real slope of this listener is steeper than the assumed slope in the prediction.

It is concluded that equation (5) offers a good prediction of the accuracy as a function of number of trials in a threshold run, when assuming a standard psychometric slope of 0.5 and a j-value of 1.4.

3.5 Specifying the number of trials

In this chapter we have established that equation (5) is a good predictor of the decrease of the variability of the threshold estimates as more trials are included in a threshold run. The prediction allows us to make an informed decision on how many trials need to be including in a threshold run to obtain a certain level of accuracy.

The accuracy of the threshold estimates as a function of the number of trials can be predicted using equation (5)

$$\mathrm{SD} = \frac{\mathrm{j} * \pi}{\sqrt{3} * k * \sqrt{\text{no of trials}}} \qquad (5)$$

The number of trials needed to acquire a certain level of error (SD) can be calculated by rearranging equation (5)

$$\mathrm{n^o \text{ of trials}} = \frac{\mathrm{j}^2 * \pi^2}{3 * k^2 * SD^2} \qquad (6)$$

where SD is the required level of accuracy

A number of examples will illustrate the principle:

A. What is the number of trials required to estimate the threshold with an accuracy corresponding to a standard deviation of 1 dB (SD = 1 dB SPL) assuming a standard psychometric slope (*k=0.5*) and *j=1.4*? Substituting these values for the SD, *k* and *j* in equation (6) results in

$$\mathrm{n^o \text{ of trials}} = \frac{1.4^2 * \pi^2}{3 * .05^2 * 1^2} = \mathbf{26\ trials}$$

B. What is the number of trial required to estimate the threshold with an accuracy corresponding to a standard deviation of 1.5 dB (SD = 1.5 dB SPL) assuming a standard psychometric slope (k=0.5) and $j=1.4$? Substituting these values for the SD, k and j in equation (6) results in

$$\text{n}^{\text{o}} \text{ of trials} = \frac{1.4^2 * \pi^2}{3 * .05^2 * 1.5^2} = \textbf{11 trials}$$

A standard deviation of 1.5 dB SPL represents a level of accuracy that is satisfactory for the threshold measurements included in the hearing profiles discussed further in this thesis. All measurements will therefore be performed using threshold runs consisting of 10 ‘threshold’ trials (not including initial trials and catch trials).

3.6 Conclusion

In this chapter, we explored how the accuracy of threshold estimates depends on the number of trials used in a threshold run. The Single-Interval-Up-Down method was used to generate threshold estimates in human listeners and using computer simulations. An analysis of the threshold estimates on a trial-by-trial basis showed how the standard deviation of the threshold estimates decreases as the number of trials is increased.

Subsequently a formula was derived to predict how the accuracy improves as more trials are included in a threshold run. The formula was shown to be consistent with the data from human listeners and computer simulations.

The formula can be used to specify the number of trials for a given level of accuracy. This is important since not all situations ask for the same level of accuracy in the threshold estimates. Researchers/clinicians who work in settings where a meticulously measured threshold estimate is less important can now make an informed decision on the required accuracy and on the number of trials they need to acquire this level of accuracy. On the other hand, when more precision is needed, the formula will indicate the need for more trials.

The formula predicts that for a standard deviation of 1.5 dB SPL, only 10 trials (not including catch and initial phase) are needed in an absolute threshold measurement. It was consequently decided that 10 trials would be used for all threshold measurements conducted for the development of the hearing profiles.

The predictive formula was evaluated only for the measurements of absolute thresholds and not for forward-masked thresholds. Unfortunately, not a lot of information is available about the slope of the psychometric function in a forward-masking task and no suggestions concerning a slope parameter *k* have been found in the literature. It has been suggested that the slope of psychometric function is a function of the frequency ratio between the masker and the probe (Yasin and Plack, 2008, Schairer et al., 2003). This is explained in terms of the effect of compressive non-linearity on the masker. This implies that different psychometric slopes will apply in the forward-masking measurements performed for the hearing profiles since different frequency-ratios will be used. A prediction of the accuracy could therefore be misleading. Nevertheless, it was decided that 10 trials would also be the standard setting for measurements of forward-masked thresholds.

There is evidence that the psychometric slope is steeper in impaired listeners (Arehart et al., 1990, Carlyon et al., 1990). A better accuracy can therefore be expected for the threshold measurements of these listeners. The stop criterion of 10 trials, set on the basis of the analysis presented in this chapter, will consequently also be suitable for threshold measurements of impaired listeners.

CHAPTER 4

Evaluation of the Single-Interval-Up-Down procedure

4.1 Introduction

In Chapter 2 of this thesis a threshold estimation procedure, the Single-Interval-Up-Down method has been presented. Day-to-day experience with this method has been extremely satisfactory in a number of ways. First, the method is user-friendly both for the researcher/clinician as for the participants/patients: the tasks are simple to explain, participants are very comfortable performing the tasks and settle into the testing easily. Therefore little training is required. Secondly, the threshold estimates are easily measured and have shown to be reliable. Finally, the method is fast and little time is needed to obtain a threshold. It is also our opinion that the method is as valuable as traditional Forced choice methods.

For presentational purposes, this chapter will describe 3 short experiments which will systematically address the issues of training, reliability and speed. Furthermore a comparison of the Single-Interval-Up-Down method with a traditional 2I2AFC a two-down, one-up procedure will be presented. A similar comparison has already been reported by Leek and colleagues (2000).

In these experiments we are interested in:

- How much training is required for a detection task in quiet and for detection in the presence of a forward masker?

- Is there a substantial threshold difference between a 2I2AFC method and our simple SI-UD method?
- How fast is the Single-Interval-Up-Down method and how does this compare to the time needed for a threshold estimate in a 2 AFC ++ method?

4.2 Training

A training effect can be defined as an improvement of performance between the moment of introduction to a new task and later executions of that task. This improvement can come in the form of *more sensitive* threshold estimates but can also be considered to be *more stable* estimates.

In many labs, participants go through numerous hours of training before taking part in the actual experiment. In order for our procedure to be useful in a clinical environment however, it is important that a limited amount of training is required before test results are considered reliable. In other words, the threshold estimates should be stable after a minimum number of runs.

A number of normal listeners were introduced to the Single-Interval-Up-Down method. Those 'introductory' measurements were analysed to explore possible changes in performance and to identify the possible presence of a training effect. The first experiment evaluates the training effect for threshold detection in quiet. Experiment 2 does the same for the estimation of forward-masked thresholds. The data was collected by a masters-student as part of the masters-dissertation.

4.2.1 Detection in quiet

The first experiment is aimed at exploring the training effect in a detection task in quiet (see section 2.2).

Method

Stimulus. Absolute thresholds were measured for a 1-kHz pure tone. The duration of the tone was 100 ms. The stimulus was preceded by a cue consisting of a 1-kHz pure tone which was 10 dB SPL more intense than the test tone.

Procedure. Thresholds were measured using the Single-Interval-Up-Down method over 5 successive runs. Each run contained 10 trials (not including catch trials and the trials in the initial phase). The initial step size was set at 10 dB SPL, the small step size was set at 2 dB SPL. After receiving verbal instructions on the task, subjects entered the booth and completed all 5 runs consecutively.

Listeners. All listeners were student volunteers. Subsequent testing measurement of their absolute thresholds confirmed that they all had normal hearing. The listeners had no previous experience of psychoacoustic experiments and had not previously performed any tasks using the Single-Interval-Up-Down procedure.

Task. In this test participants were asked to count the number of tones heard. If the participant reported 2 tones, it was assumed that the test stimulus had been heard. If the participant reported only 1 tone, it was assumed that only the cue had been heard and the test tone had not been heard. The level of the tones changed between trials according to the participant’s response.

Results

Figure 4.2-1 shows the threshold estimates as a function of the threshold run for individual participants. The data points represent threshold estimates for each participant. The full line represents the average threshold across all participants as a function of threshold run.

The mean threshold across participants does not vary across runs. There is no apparent trend in the threshold estimates across runs.

As a second check, an analysis of the standard deviations of the threshold estimates across listeners was performed. There is a small tendency towards a reduced spread in the threshold estimates from the 4th run onwards. The mean standard deviations of the individual thresholds per threshold run are 5.2, 5.0, 5.2, 3.9 and 4.0 dB SPL for run 1 to 5 respectively.

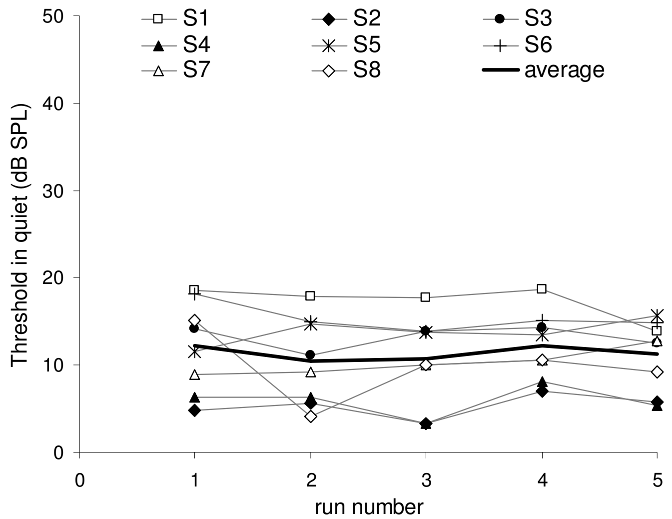


Figure 4.2-1. Threshold estimates as a function of threshold run. Data points (connected by thin grey lines) represent individual threshold estimates as a function of run number. The thick black line represents the mean threshold as a function of run number across all participants.

Discussion

The threshold estimates as a function of threshold run presented in Experiment 1 indicate that no substantial training effect is present in a detection task in quiet when using the Single-Interval-

Up-Down method. The data reflects the normal experience that the threshold estimates do not vary substantially after the first run. The variability found in the thresholds per run shows a minor reduction after 3 runs which could indicate that threshold estimates become more consistent after 3 threshold runs.

4.2.2 Detection in the presence of a forward masker

The second experiment is similar to experiment 1 but looks at the training effect in a detection task in the presence of a forward masker (see section 2.2).

Method

Stimulus. Masker thresholds were measured in a forward masking task. The stimulus consisted of a 108-ms masker tone followed by an 8-ms probe tone. The masker and probe were separated by a 10-ms gap. The frequency of both probe and masker was 1 kHz. The test stimulus was preceded by a cue which is the same as the test stimulus but the masker level was 10 dB SPL less intense.

Procedure. The Single-Interval-Up-Down method was used over 5 successive runs. Each run contained 10 trials (not including catch trials and the trials in the Initial phase). The initial step size was set at 10 dB SPL, the small step size was set at 2 dB SPL. These forward masking measurements were made after the measurements in experiment 1. After receiving verbal instructions on the task, subjects entered the booth and completed all 5 runs consecutively.

Task. In this test participants were asked to count the number of probe tones heard, while ignoring the masker tones. If the participant reported 2 probes, it was assumed that the test probe had been heard. If the participant reported only 1 probe, it was assumed that only the cue probe had been heard and the test probe had not been heard. The masker levels were adjusted between trials according to the participant's response.

Listeners. All listeners were student volunteers and had previously participated in experiment 1. All had normal hearing.

Results

Figure 4.2-2 shows the masker threshold as a function of the threshold run. The individual data points represent threshold estimates for each participant as a function of threshold run. The full line represents the mean threshold across all participants as a function of threshold run.

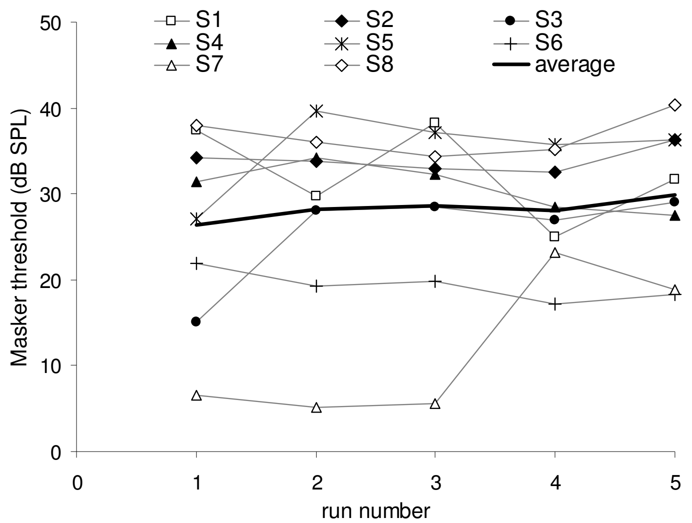


Figure 4.2-2. Masker thresholds as a function of threshold run. Data points (connected by thin grey lines) represent threshold estimates for each participant. The full line represents the mean threshold across all participants.

There is a 5-dB SPL increase in the mean masker threshold across trials. This trend can be explained in terms of two participants who had difficulty in performing the task in the first runs. One listener shows very low threshold estimates in the first 3 runs. A second listener showed a similar pattern for the first run. However, this listener shows more realistic and stable masker thresholds from the 2nd run onwards. The six remaining listeners did not show this pattern and displayed stable masker thresholds from the first run onwards.

Discussion

It can be concluded that some training effect is present at an early stage of performing the forward-masking task. The task is more complex than a simple detection task in quiet since the listener is asked to detect a probe while ignoring a preceding masker. Six listeners were able to perform this task adequately straight away yielding stable masker thresholds in all 5 threshold runs. Two listeners needed 1 and 3 threshold runs to generate acceptable threshold estimates. It is assumed that those listeners were confused about what they were listening for reporting the presence of the masker at some occasions rather than the presence of the probe. When listeners have difficulties with the task, the masker thresholds are unrealistically low and this is noticable early in a threshold run. This is encouraging since it implies that difficulties can be spotted easily if they would occur, after which some repeated measurements can be conducted.

4.3 Comparison with a Forced-choice-procedure

A commonly used threshold estimation procedure is a two-interval, two-alternative forced choice paradigm (2I2AFC) combined with a one-up, two-down adaptive tracking procedure (Levitt, 1971). This procedure is widely accepted as a reference procedure when comparing different measurement procedures. A comparison between a 2I2AFC-procedure and a Single interval procedure has previously been reported by Leek et al. (2000). They found an excellent agreement between the thresholds estimated using an ML yes-no procedure and a 2I2AFC-procedure. A small difference in variability was found, with the lowest standard deviation in the ML-procedure. The standard deviation for the 2I2AFC procedure were 1.56 dB SPL, compared to 0.97 dB SPL in the ML-procedure. The time needed to obtain a threshold estimate was considerably higher for the 2I2AFC-procedure. A threshold run in the ML-condition had an average duration of 45 seconds while 2 minutes and 13 seconds were needed for a similar run in the 2I2AFC-condition. This corresponded with an average of 24 trials and 53 trials per threshold run for the ML- and 2I2AFC-condition respectively.

In this section, the Single-Interval-Up-Down method is compared with a conventional 2I2AFC one-up, two-down procedure. We looked at threshold estimates in quiet, standard deviations of the estimates and time needed per measurement for each of the 2 procedures.

4.3.1 Method

Stimuli. Absolute thresholds were measured for a 1-kHz pure tone. The duration of the tone was 100 ms. In the Single-Interval-Up-Down–condition, the stimulus was preceded by a cue consisting of a 1-kHz pure tone which was 10 dB SPL more intense than the test tone. In the 2I2AFC-condition, no cue was presented.

Procedure. Thresholds were measured using the Single-Interval-Up-Down method and using a 2I2AFC method. Thresholds were measured over 5 successive runs in each condition. The threshold estimates reported are the mean of 5 measurements.

In the *Single-Interval-Up-Down-condition*, each run contained 10 trials (not including the catch trials and the trials in the initial phase). The initial step size was set at 10 dB SPL, the small step size was set at 2 dB SPL.

In the *2I2AFC-condition*, thresholds were measured using an one-up, two-down adaptive procedure (Levitt, 1972). Each run was terminated after 8 reversals and the threshold was calculated from the average level of the last 6 reversals. The step size was initially set at 10 dB SPL and reduced to 2 dB SPL after the 2 first reversals.

The time to obtain each threshold was recorded as was the total number of trials for the 2I2AFC-procedure.

Listeners. All listeners were student volunteers. Measurement of their absolute thresholds at different frequencies confirmed that they all had normal hearing. Six listeners (S1-6) had also participated in experiment 1 and 2 described in section 4.2.

Task. In the *Single-Interval-Up-Down-condition*, participants were asked to count the number of tones heard. If the participant reported 2 tones it was assumed that the test stimulus had been heard. If the participant reported only 1 tone it was assumed that only the cue had been heard and the test tone had not been heard. The level of the tones changed between trials according to the participant’s response.

In the *2I2AFC-condition*, participants were presented with two successive visual intervals and the signal was presented in one of the two intervals. The listener had to indicate in which interval the signal occurred.

4.3.2 Results

Threshold estimates and standard deviations. In the 2I2AFC-condition, three data points were omitted. The data points originated from 2 different listeners and were omitted on the basis that they were more than 5 SDs (typical SD within participant) away from the mean. No data points were omitted in the Single-Interval-Up-Down-condition.

To enable a direct comparison of the threshold obtained in the two procedures, an adjustment of the 2I2AFC thresholds, suggested by Leek and colleagues (2000), was necessary. The psychometric function used in our Single Interval Procedure (see section 2.2) targets a probability scale from 0 to 100%. The 2I2AFC, two-down, one-up procedure however targets the 70.7 % point on a 50-100% scale. We rescaled the 2I2AFC range as suggested by McKee et al. (1985) and found that the 70.7% point on a 50-100% scale is equivalent to the 41.4 % point on a psychometric function with a scale ranging from 0-100 %. The level difference corresponding to a probability change from 41.4% to 50% is 0.7 dB when assuming a slope parameter *k* of 0.5. Therefore 0.7 dB had to be added to the 2I2AFC-thresholds in order to make a comparison possible.

Table 4.3-1. Threshold estimate (in dB SPL) in a Single-Interval-Up-Down method and a 2I2AFC++ method. First rows show the mean threshold for the individual participants. This threshold is based on 5 threshold estimates. Standard deviations of the within-subject estimates are between brackets. Bottom row (bold) shows the mean thresholds (in dB SPL) across all participants. The mean standard deviation across participants is shown between brackets.

Participant	SI UD	2I2AFC++
S1	16.6 (2.0)	17.0 (4.6)
S2	8.2 (1.9)	7.1 (1.7)
S3	10.9 (1.8)	10.7 (1.9)
S4	-2.4 (2.3)	-0.4 (2.2)
S5	11.7 (1.9)	5.7 (6.7)
S6	5.4 (1.3)	3.8 (3.1)
S7	9.0 (2.2)	9.1 (1.6)
S8	8.9 (1.9)	9.9 (4.3)
S9	5.2 (0.7)	4.9 (4.4)
MEAN THRESHOLD (mean SD)	**8.2 (1.8)**	**7.6 (3.4)**

Table 4.3-1 shows the mean threshold estimates obtained using the Single-Interval-Up-Down method and using the 2I2AFC two-down, one-up method for each of the 9 participants. The standard deviation of the threshold estimates within a listener is shown between brackets. The bottom row shows the mean thresholds across participants for the 2 methods with the mean standard deviation across participants between brackets. The thresholds obtained using the Single-Interval-Up-Down method are very similar to the thresholds obtained in the 2I2AFC two-

down one-up procedure. The standard deviations of the threshold in the Single Interval procedure (1.8 dB SPL) are lower than the standard deviations of the thresholds in the 2I2AFC-procedure (3.4 dB SPL).

Time needed per threshold run. Table 4.3-2 shows the mean time needed per threshold run (in seconds) for both measurement procedures for each of the 9 participants. The bottom row represents the mean time needed per threshold run across all listeners. The mean time needed per threshold run in the Single-Interval-Up-Down-condition was 61 seconds opposed to 160 seconds in the 2I2AFC-condition.

Table 4.3-2. Time needed per threshold run (in seconds) in a Single-Interval-Up-Down method and a 2I2AFC one-up, two-down method. First rows show the mean time (in sec) per threshold run for the individual participants. Bottom row (bold) shows the mean time needed per threshold run across all participants.

Participant	SI UD	2I2AFC++
S1	50	176
S2	75	183
S3	56	160
S4	99	181
S5	50	117
S6	55	148
S7	61	156
S8	55	183
S9	50	138
MEAN TIME NEEDED PER RUN	**61 sec**	**160 sec**

The mean total number of trials needed in the 2I2AFC-condition was 32 trials across all listeners. On the other hand, the number of trials used in the Single-Interval-Up-Down-condition was fixed at 10 trials (not including the catch trials and the trials in the initial phase) at the start of the experiment.

4.3.3 Conclusion

The results presented above have shown that the agreement between thresholds obtained with the Single-Interval-Up-Down method and results from the 2I2AFC-method is excellent. The threshold estimates found in the two threshold estimation procedures are very similar and indicate that both procedures are equally sensitive. The initial concern that the Single Interval procedure does not control for a criterion shift (section 2.3) and therefore could yield different threshold estimates was shown to be unfounded.

The accuracy provided by the Single-Interval-Up-Down method is better than the accuracy found in the 2I2AFC-method with lower standard deviations of the threshold estimates in the Single-Interval-Up-Down-condition. There is no evidence that a criterion shift has occurred in the Single-Interval-Up-Down-method.

As expected, the time needed to obtain a threshold estimate is limited when using a Single interval Procedure. The Single interval method is faster than the 2I2AFC-procedure in all our participants. The difference in time needed per threshold run can be explained by the difference in number of trials needed to find a threshold estimate. In the Single Interval procedure a threshold estimate is generated after 10 ‘threshold’ trials opposed to the 2I2AFC-condition where a fixed and substantial number of reversals is required before a threshold estimate is yielded.

4.4 Discussion

In this chapter, an evaluation of the Single-Interval-Up-Down method has been presented in terms of user friendliness, accuracy and speed. The first two experiments looked at the training effect associated with two different detection tasks using the Single-interval-Up-Down method.

No training effect was found when measuring absolute thresholds. Listeners had slightly more trouble with forward-masked thresholds. This effect however is minor and after 3 training runs the results appear to be stable.

In a third experiment, we compared threshold estimates obtained using the Single-Interval-Up-Down method with the more traditional 2I2AFC, two-down, one-up procedures. No major differences in thresholds were found between the two procedures, which is a finding in agreement with the results reported by Leek and colleagues (2000).

The variability of the thresholds in the Single-Interval-Up-Down-condition was better than in the 2I2AFC-condition. This is in agreement with the findings of Leek and colleagues and Amitay et al. (2006). The overall variability in Leek's study was lower than the variability reported above since the threshold runs consisted of more trials in both conditions. In our experiment the threshold runs were relatively short in both conditions resulting in larger standard deviations. This might also explain the larger variability in the 2I2AFC-condition and suggest the need for even longer runs when using forced-choice paradigms.

The time needed per threshold run is considerably shorter in the Single-Interval-Up-Down-condition than in the 2I2AFC-condition. The Single interval-condition showed to be over 2.5 times faster than the 2I2AFC-condition. This result also agrees with the findings reported by Leek and colleagues. The overall time per threshold run in Leek's study was shorter than in the experiment reported here since Leek used a stimulus which was only 20 ms in duration compared to the stimulus used in our study which was 100 ms long. The difference in time between the two studies can thus be explained in terms of differences in the stimulus presentation time.

It is concluded that the Single-Interval-Up-Down method is a simple, reliable and fast procedure for threshold estimates, which makes it an adequate threshold estimation procedure to be used for the measurements used for the hearing profiles in this thesis and in a clinical setting.

CHAPTER 5

Assessment Procedure

5.1 Introduction

In the previous chapters (Chapter 2, 3 and 4) an efficient and reliable threshold estimation procedure was presented. This measurement procedure will be employed when collecting data on various measures of normal and impaired listeners. The problem which presents itself at this point is the question as to which measures will be most informative and efficient when developing a profile of a person's hearing. This comprehensive set of data will subsequently be used to develop a computer model of this individual's hearing. Ideally, the profile will offer an insight into the underlying pathology, information to build the computer model of this individual's hearing and offer clues on how to remediate the hearing problem.

Each participant was subjected to a general audiological screening and a set of psychoacoustic tests that provides measures of auditory sensitivity, frequency selectivity and auditory compression. This chapter will present each test separately; discuss the general principles, the normal and impaired manifestations and the potential conclusions that might be drawn from different patterns. In the past, each psychoacoustic test on itself has been studied in detail by many researchers. It is however beyond the scope of this project to give a detailed review of each measure. The presentation of the different measures will therefore focus on the features relevant only to this project.

The tests presented in this next chapter are split up in 2 sections:

1. Clinical screening consisting of otoscopy, Rinne tuning fork test, Tympanometry, DP OAE measurements and Pure Tone Audiometry.
2. Psychoacoustic measures include absolute threshold, threshold as a function of duration, Iso Forward Masking Curves and Temporal Masking Curves.

5.2 Clinical screening

The clinical screening aims to formulate a clinical diagnosis at the start of the participant's testing series. It is a baseline measurement against which the psychoacoustic measures are evaluated. The clinical measures are presented in a general manner and will not be discussed in great detail.

5.2.1 Otoscopy

Otoscopy is a visual inspection of the tympanic membrane and ear canal by means of an otoscope. The inspection is necessary to identify possible malformations or obstructions of the ear canal, look at the condition of the tympanic membrane and identify signs of infection behind the tympanic membrane (Katz, 2002, p. 16).

5.2.2 Rinne tuning fork test

The Rinne tuning fork test is a quick test to evaluate the presence of a conductive hearing loss. A vibrating tuning fork is placed behind the ear, on the mastoid bone. A tone should be audible to the participant. When the participant can no longer hear the vibration, the tuning fork is held in front of the ear. The participant should once more be able to hear the tone. If nothing is heard, it is an indication that a conductive hearing loss is present. If hearing is normal or a sensorineural hearing problem is present, the participant will be able to hear the tone again when the tuning fork is subsequently held in front of the ear (Katz, 2002, p. 79).

5.2.3 Tympanometry

Tympanometry is an objective measure of the middle ear function. It tests the condition of the middle ear and the mobility of the tympanic membrane and the ossicular chain (the 3 ear bones) by creating variations of air pressure in the ear canal.

A tympanogram can identify middle ear problems such as perforation of the tympanic membrane, rigidity of the tympanic membrane, fluid in the middle ear cavity or reduced mobility of the tympanic membrane (due to the presence of scar-tissue) or the ear bones (due to otosclerosis) (Katz, 2002, p. 175).

The tympanograms measured in the participants discussed in Chapter 7 and 8 were measured using a Kamplex AA222 combination audiometer/impedance test station (PC Werth, London, UK).

5.2.4 Distortion Product Otoacoustic Emissions (DP OAEs)

Otoacoustic emissions (OAEs) are sounds of cochlear origin, which can be recorded by a microphone fitted into the ear canal. They provide a simple, efficient and non-invasive objective indicator of healthy cochlear function. OAEs are known to disappear with inner ear damage (Kemp, 2002).

Distortion Product Otoacoustic Emissions (DP OAEs) are generated in the inner ear by presenting two tones close in frequency. The distortion products are generated as a result of the nonlinear response properties of the basilar membrane and are therefore an indicator of the presence of compression in the auditory system (Katz, 2002, p. 447).

The absence of DPOAEs is generally associated with reduced or absent compression which has been attributed to OHC damage. For a hearing loss greater than 50 dB SPL, no DPOAEs are expected to be recorded regardless of the underlying pathology (Gorga et al., 1997).

The DP OAE results reported in Chapter 7 and 8 were recorded using a dual channel ILO292 Echoport USB-II (Otodynamics Ltd, Hatfield UK). DPOAE responses are recorded at frequencies 1, 1.4, 2, 2.8, 4, 6 and 8 kHz. The frequency separation of the primaries (f2/f1) is set at 1.22, with primary levels of 55 and 65 dB SPL. The responses are presented in SNR-levels as a function of frequency (Figure 5.2-1) and are evaluated using the normative data suggested by Gorga et al. (1997). The dotted line in Figure 5.2-1 represents the 5th percentile of responses of the normal listeners. Gorga and colleagues found considerable overlap at these levels with the responses of impaired listeners creating a grey zone of responses found both in normal and impaired listeners. The standard values presented in the figure will therefore only be useful when responses below these values are found. In that case we can conclude that the response is lower than the bottom 5% of the normal listeners and that there is a strong indication of a reduced/abnormal response.

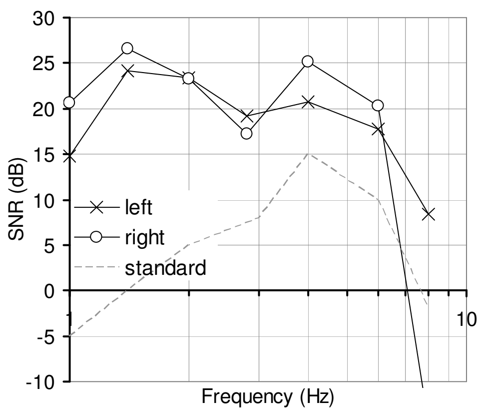


Figure 5.2-1. DP OAE responses of a normal listener (CM) expressed in SNR at 1, 1.4, 2, 2.8, 4, 6 and 8 kHz, in the left and right ear. Responses are expressed in SNR (dB). The dotted grey line represent the 5th percentile of responses of normal listeners reported by Gorga et al. (1997).

5.2.5 Pure tone Audiometry (PTA)

Pure Tone Audiometry is the key hearing test used in the clinic. It is a subjective measure aimed at the identification of hearing thresholds at different frequencies. An audiometric threshold can be defined as the lowest intensity at which the listener can identify the presence of a signal at least 50% of the time (Katz, 2002).

The thresholds are plotted on an audiogram which is a standardized way of representing the hearing levels (dB HL) as a function of frequency (kHz). The hearing levels are specified relative to the published average threshold at various frequencies of healthy listeners with normal hearing. This 'normal' average is set at 0 dB HL. The average values of normal hearing are standardized values determined in BS EN ISO 389-1:2000. Normal hearing is between -10 and 15 dB HL. The audiogram forms the basis for determining the degree, type and configuration of a hearing loss. Figure 5.2-2A illustrates a typical normal audiogram. A hearing loss is associated with an increase in threshold.

Two types of thresholds are determined in a standard PTA assessment:
- Air conduction thresholds: measured using sounds presented through headphones. The sound reaches the inner ear via the outer and middle ear.
- Bone conduction thresholds: measured using sounds presented through a bone vibrator which is usually placed behind the ear on the mastoid bone. The sounds are transmitted through the bones of the skull and reach the inner ear directly bypassing the outer and middle ear. Increased bone conduction thresholds indicate a problem in the inner ear.

Impaired air conduction thresholds can be due to a problem in either or both the outer/middle ear and the inner ear. Increased bone conduction thresholds however can only be due to a problem in the inner ear (sensorineural loss). Normal bone conduction thresholds combined with impaired air conduction thresholds indicate a conductive hearing loss. The difference between the bone and the air conduction thresholds is called the ‘air-bone gap’.

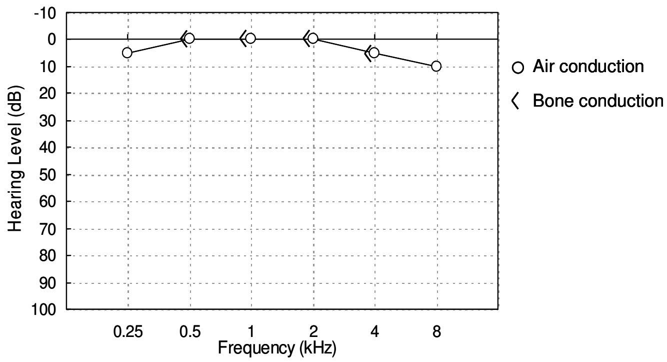
Types of hearing loss

Conductive hearing loss occurs when sound is not conducted normally through the outer or middle ear. Since sound is picked up by the sensitive inner ear even if the outer of middle ear is not functioning normally, a conductive hearing loss often causes mild to moderate impairment. Hearing thresholds are not expected to rise above 55-60 dB from outer or middle ear problems alone. The hearing loss acts as an attenuator but does not affect the quality of the sounds (Møller, 2000). Figure 5.2-2B shows a typical audiogram for a conductive hearing loss. The bone conduction thresholds are normal but a loss occurs in the air conduction thresholds.

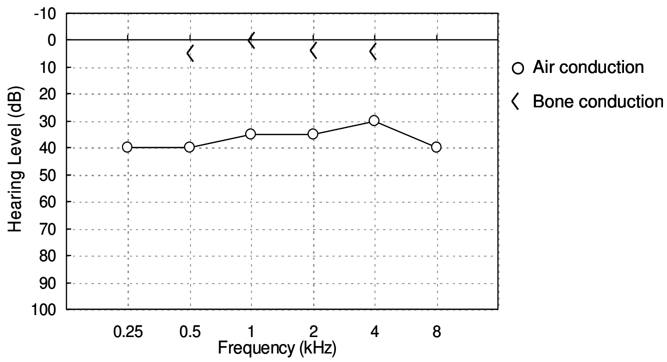
Sensorineural hearing loss is caused by problems in the inner ear, the auditory nerve or the entire auditory nervous system. The sensorineural losses discussed in this thesis will mainly involve impairments originating in the cochlea. In cochlear hearing loss, impaired functioning of the hair cells is the most common cause. Sensorineural hearing loss can be mild, moderate, severe or profound to the point of total deafness (Møller, 2000). Figure 5.2-2C shows an example of an audiogram for a sensorineural hearing loss. The air and bone conduction thresholds are equally raised.

A mixed loss is a combination of a conductive and sensorineural hearing loss (Figure 5.2-2 D). The audiogram shows raised air and bone conduction thresholds with a large air-bone gap between the two sets of thresholds.

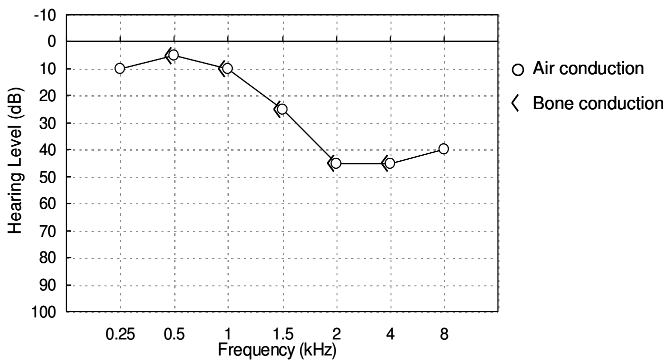
A. normal hearing



B. Conductive hearing loss



C. Sensorineural hearing loss



D. Mixed hearing loss

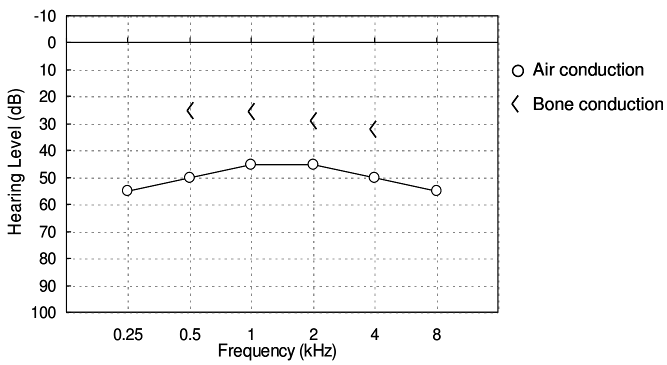


Figure 5.2-2. Schematic representation of audiograms associated with A. normal hearing, B. Conductive hearing loss, C. Sensorineural hearing loss, D. Mixed hearing loss. The circles represent air conduction thresholds, brackets show bone conduction thresholds.

Degree of hearing loss

The severity of the hearing loss is established based on the air conduction thresholds. Table 5.2-1 shows a classification for the degree of hearing loss, proposed by Goodman (1965) and adapted by Clark (1981)(taken from Katz, 2000, p. 82).

Table 5.2-1. Classification for the degree of hearing loss.

Hearing level (dB HL)	Classification
-10 to 15	Normal hearing
16 to 25	Slight hearing loss
26 to 40	Mild hearing loss
41 to 55	Moderate hearing loss
56 to 70	Moderately severe hearing loss
71 to 90	Severe hearing loss
> 90	Profound hearing loss

Testing procedure

The testing procedures were performed following the BSA recommendations specified in BS EN ISO 8253-1. Air conduction thresholds are measured when normal listeners are tested. For the impaired listeners both air and bone conduction thresholds are established. Masking is performed where required. In cases where one ear is clearly better than the other, masking the good ear with noise prevents it from responding to the sounds presented to the test (worse) ear.

The audiograms presented in Chapter 7 and 8 were measured using a Kamplex AA222 combination audiometer/impedance test station (PC Werth, London, UK) using Telephonics TDH 39P headphones.

5.3 Sensitivity: Absolute threshold

Absolute thresholds are an indication of the sensitivity of the auditory system and can be defined as the intensity of the tone necessary to detect it in 50% of the presentations. The main difference between audiometric thresholds and absolute thresholds is the fact that absolute thresholds are expressed in dB SPL as opposed to the audiometric thresholds discussed in the clinical screening where dB HL is used. Audiometric thresholds are actually absolute thresholds converted into dB HL relative to a normative set of thresholds reflecting the average hearing performance.

In the absolute threshold measure, thresholds for pure tones are measured at 250, 500, 1000, 2000, 4000 and 8000 Hz and for tone durations of 8 and 500 ms. Thresholds for the long tone (500 ms) are compared to standardized thresholds (ISO 389-8[2]) to establish the presence or absence of an impairment (Figure 5.3-1). Thresholds for a long tone above the standard are interpreted as impaired.

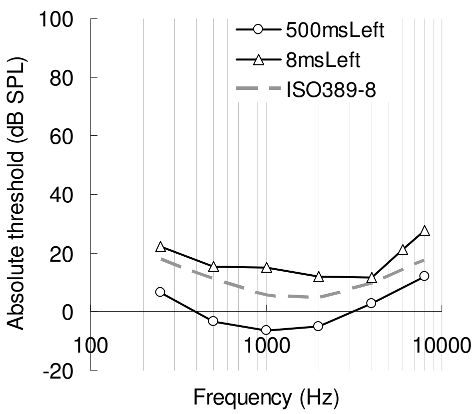


Figure 5.3-1. Example of absolute thresholds (dB SPL) as a function of frequency for pure tones with tone durations of 8 ms (triangles) and 500 ms (circles) at frequencies 250, 500, 1000, 2000, 4000 and 8000 Hz. Data shown is obtained in normal listener (CM).

[2] ISO 389-8 Reference zero for the calibration of audiometric equipment -- Part 8: Reference equivalent threshold sound pressure levels for pure tones and circumaural earphones

The normal pattern of the thresholds as a function of frequency on a dB SPL-scale is a bowl-shaped function. The auditory system is most sensitive at the mid-range frequencies. The low frequencies are attenuated while passing through the middle ear resulting in higher thresholds (Moore, 2004). The shape of the function is independent of tone duration so in normal hearing the 500 ms and the 8 ms function are parallel to each other. A different pattern could potentially pinpoint an anomaly in the individual's hearing.

It is generally established that thresholds change as a function of the duration of the stimulus. Thresholds for an 8-ms tone are normally higher than for the long tone. The difference in threshold between a short and a long tone is smaller in impaired listeners compared to normal listeners. It is our opinion that this difference in threshold is of diagnostic value. This issue will be discussed in depth in Chapter 6.

5.4 Frequency selectivity: Iso Forward Masking Curves

5.4.1 What is frequency selectivity?

Frequency selectivity is the ability of the auditory system to separate tones of different frequencies. It is of specific interest in this project because it reflects how a tone of a particular frequency interferes with a tone with a different frequency. Failure to separate tones is a particular problem for many impaired listeners. The smearing among frequency components makes it difficult to listen to speech in a noisy environment.

5.4.2 Iso Forward Masking Curves: a measure of frequency selectivity

Forward masking technique is used to investigate how tones interfere with each other. We have adopted a paradigm which we will refer to as Iso Forward Masking Curves (IFMC). The procedure is slightly different from the commonly known and used Psychophysical Tuning Curves (PTCs). The difference between the 2 paradigms will be discussed below.

IFMCs are in fact iso-response curves representing the level of pure-tone maskers at different masker frequencies necessary to just mask a pure-tone fixed-level probe at a specific frequency. In other words, the IFMC connects points with equal capacity to mask a probe in a forward masking task, hence the term Iso Forward Masking Curve. The probe frequency is fixed for each IFMC. The maskers are varied in frequency around the probe frequency (f_p) and are represented as a ratio of the probe frequency. The masker frequency associated with the lowest masker threshold is referred to as the tip frequency. This tip frequency represents the frequency of the masker that is most effective in masking the probe. Figure 5.4-1 shows a schematic representation of a normal IFMC. The masker frequencies are 0.5, 0.7, 0.9, 1, 1.1, 1.3 and 1.6 * f_p.

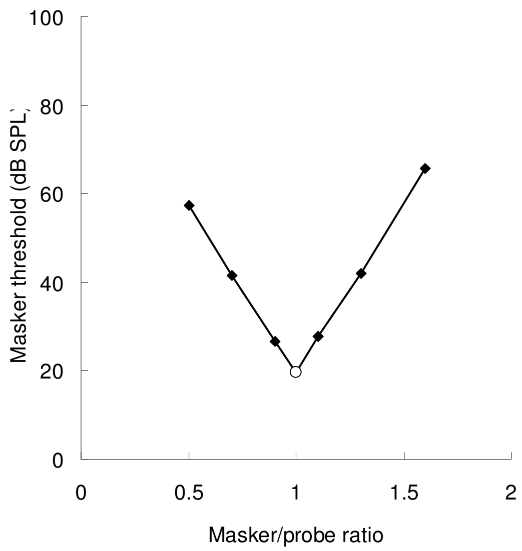


Figure 5.4-1. Schematic representation of a normal IFMC. Masker thresholds (filled symbols) are presented as a function of masker frequency. The masker frequency is represented as a ratio of the probe frequency. In this case the masker frequencies (f_m) are 0.5, 0.7, 0.9, 1, 1.1, 1.3 and 1.6 * probe frequency (f_p). The open circle represents the masker threshold at probe frequency (i.e. $f_m = f_p$).

IFMCs differ from PTCs in two aspects. The first aspect refers to the objective of the measurements. PTCs are aimed at measuring the frequency selective properties of the BM and represent single BM site tuning curves. IFMCs however are aimed at measuring the interference between sounds without claiming to measure BM tuning curves. A second aspect is concerned with the fact that the IFMC paradigm does not control for off-frequency listening. Off-frequency listening refers to the situation where the detection of a signal is based upon activity on the BM at a site with a BF different to the frequency of the signal (a more favourable filter). In most PTC-procedures, a lot of effort has been put in eliminating situations where off-frequency listening can occur usually by adding a notched noise, centred around the frequency of the signal (Patterson, 1976, Patterson and Nimmo-Smith, 1980, Rosen et al. 1998, Glasberg and Moore, 2000). The added noise ensures that the listener is using the target filter (and not necessarily the most favourable filter) resulting in more reliable PTCs. The IFMC-paradigm does include noise maskers because the aim is not to quantify BM tuning, rather the interest lies in how the signals are processed and at which site of the BM they are being detected. In an every day situation people do not have the opportunity to switch off their ‘off-frequency listening’ which is why we do allow for the most favourable filter to detect the target if this should be necessary. Evidence of off-BF listening is a potential indicator of pathology.

5.4.3 Testing procedure

Masker thresholds are measured for *probe frequencies* (f_p) 250, 500, 1000, 2000, 4000, 6000 and 8000 Hz. The *masker frequencies* (f_m) are 0.5, 0.7, 0.9, 1, 1.1, 1.3, and 1.6 * f_p. The *gap between the masker and the probe* is set at 0.01 s and is defined as the duration of the silence period between the masker offset and the probe onset.

Masker and *probe durations* are 0.108 s and 0.008 s respectively. In the 250 Hz condition, masker and probe duration are 0.108 s and 0.016 s respectively. The probe duration is increased to allow at least 4 cycles in the stimulus. The masker and probe are ramped with raised cosine onset and offset times of 0.004 s. For a schematic representation of the stimulus, see Figure 5.4-2.

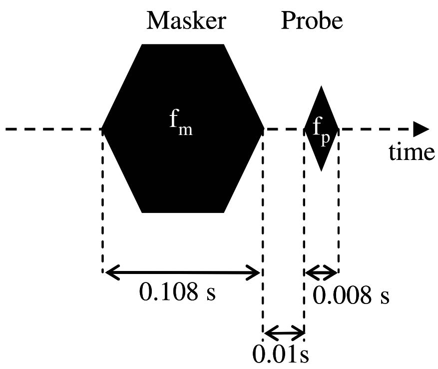


Figure 5.4-2. Schematic representation of the stimulus used in the IFMC measurements. (f_m = masker frequency, f_p = probe frequency).

For all conditions, the *probe level* is fixed at 10 dB above the 8-ms threshold (in dB SPL) of the probe. It is important to present the probe at a low level to avoid spectral splatter (Moore, 1981). Figure 5.4-3 shows an example of the IFMCs for 7 different probe frequencies in a normal listener.

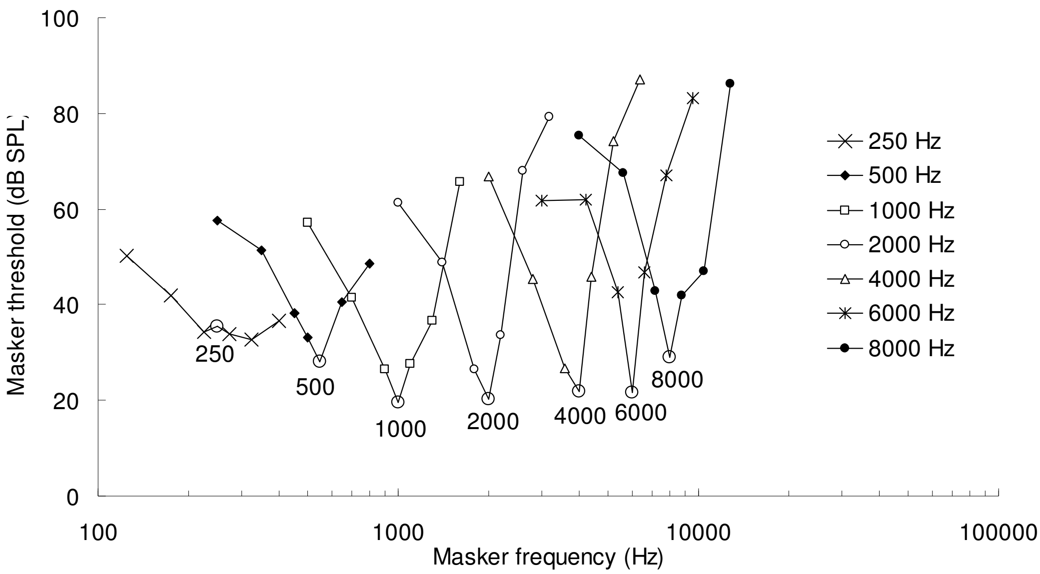


Figure 5.4-3. Example of IFMCs in a normal listener for 7 different probe frequencies. The masker frequency (f_m) are 0.5, 0.7, 0.9, 1, 1.1, 1.3 and 1.6 * probe frequency (f_p). The open circle represents the masker threshold at probe frequency (i.e. $f_m = f_p$). The accompanying value indicates the probe frequency.

5.4.4 The shape of the IFMC in normal and impaired listeners

In *normal ears* the IFMC is V-shaped with a sharp downward tip at the point where the masker frequency equals the probe frequency ($f_m = 1 * f_p$). The frequency associated with the lowest masker threshold is referred to as the tip frequency. This means that a masker is most effective when its frequency is equal to the frequency of the probe. When the masker frequency is moved away from the probe frequency, the masker has to be more intense to mask the probe and masker thresholds are increased resulting in a V-shaped curve. The sharpness of the V-shape is an indicator of the ability of the system to ignore interfering sounds when listening for a probe of a particular frequency.

When the ability to separate tones is affected, this will affect the IFMCs. The anomalies found in these IFMCs can be divided into 2 categories. Either or both categories can occur in any anomalous IFMCs. A first anomaly represents the situation where the frequency selectivity is merely reduced. The interference-potency of the maskers with frequencies different from the probe frequency is increased which means that these maskers need to be less intense to mask the probe. These lower masker thresholds (at masker frequencies away from the probe frequency) result in shallow to flat IFMCs. The masker at probe frequency remains the most effective masker which means the tip frequency is still at or very close to probe frequency (Figure 5.4-4 A & B). Shallow IFMCs are generally raised due to the raised thresholds in the impaired listener. The second type of IFMC-anomalies is characterized by a tip frequency which is shifted away from the probe frequency. A shifted tip frequency indicates that a masker with a frequency different from the probe frequency is most effective at masking the probe (Figure 5.4-4 C). This can be interpreted to mean that the probe is being detected at a BM site with a BF different from the probe frequency. The shift in tip frequency is caused by off-frequency listening and the tip of the IFMC identifies the BF of the most favourable filter for the low-intensity probe.

Figure 5.4-4 D shows a more severe example of tip frequency shift. In this IFMC, 2 tips can be observed. This suggests that there is no active filter on the site of the BM with a BF similar to the probe frequency and that the probe is being detected at adjacent sites on the BM. In this specific case these adjacent sites have a BF of 0.7 and 1.3 * f_p.

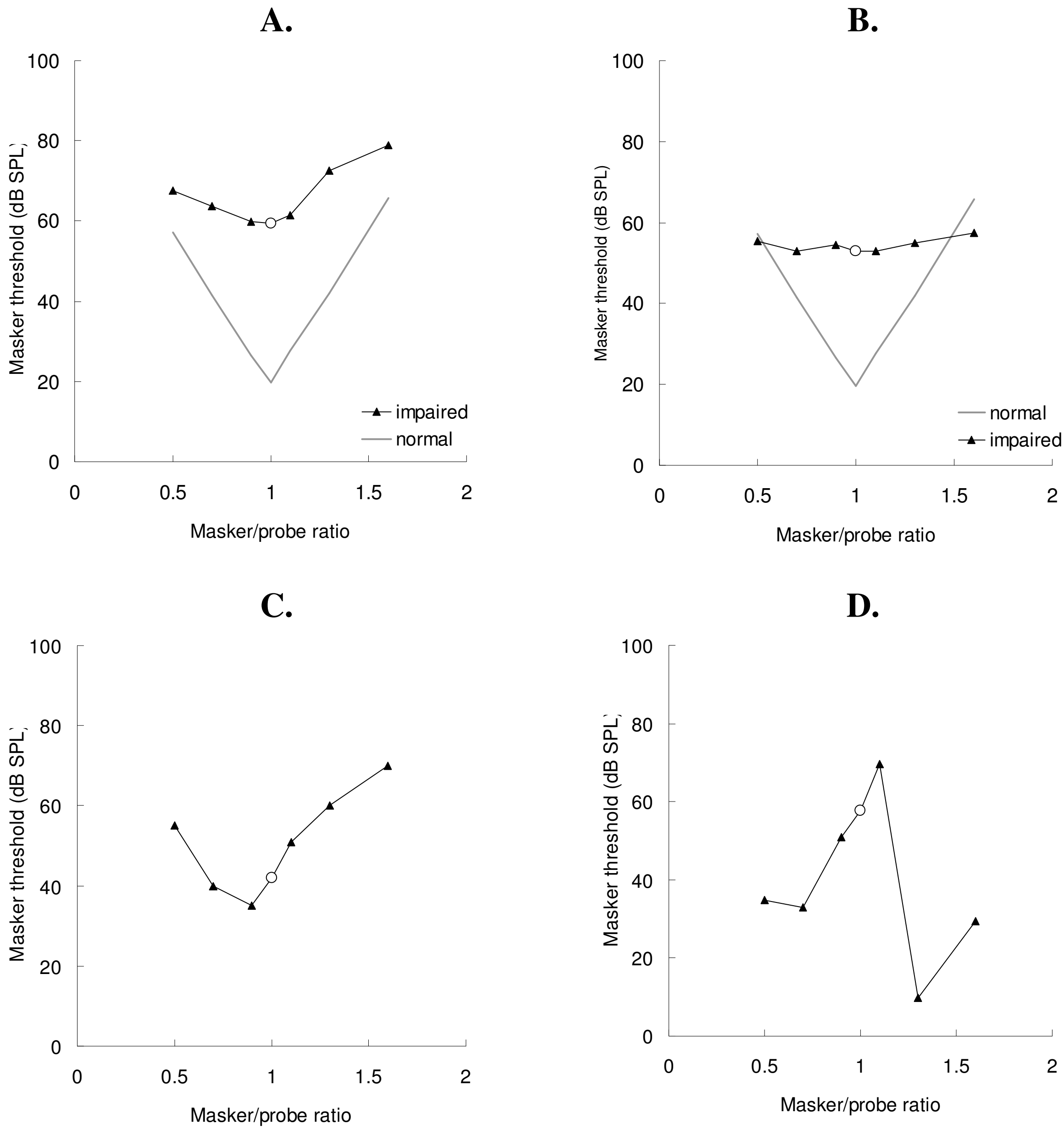


Figure 5.4-4. Illustrations of impaired IFMCs. The large open circle represents the masker thresholds at probe frequency. A and B. Frequency selectivity is reduced but the tip frequency is at probe frequency. C and D. The tip frequency is shifted away from probe frequency and reveals the most ‘favourable’ filter.

5.4.5 Quantification of IFMCs

The IFMCs are aimed at representing how sounds interfere with each other. The aim is not to quantify the frequency selectivity of the auditory system or the bandwidth of the basilar membrane tuning curves. However, to allow a meaningful interpretation of the IFMCs, some measure is needed to evaluate the sharpness of the IFMC.

The sharpness of an IFMC is evaluated on the basis of the 10 dB-width. This is the difference between the 2 frequencies where the masker threshold is increased by 10 dB relative to the tip of the IFMC. The Q10 (Q for quality factor) is a measure of this 10 dB-width relative to the probe frequency (see equation 1).

$$\text{Q10} = \frac{\text{probe frequency}}{\text{10 dB - width}} \qquad (1)$$

The 10 dB-width is estimated by fitting 2 straight lines to the tip of the IFMC. The difference between the 2 masker frequencies associated with the masker level 10 dB above the minimum masker threshold is established and Q10 values are calculated. See Figure 5.4-5 for a schematic representation of this estimation procedure. The lines are fitted using the following data-points: the lowest masker threshold (at BF), the first threshold to exceed minimum threshold + 10 dB on both sides of the curve and all the masker thresholds in between those two +10-dB maskers. In the example pictured in Fig. 5.4-5 this means the data-points used are the tip and 2 data-points (associated with higher masker thresholds) on each side of the function. The 2 highest thresholds are not included in the fit. In the case where there are no masker thresholds available 10 dB above the tip-threshold, the 2 straight lines are fitted to all the data-points and the masker thresholds needed to establish the 10 dB-width are derived from the straight line functions.

The Q10 measure is highly sensitive to changes in parameters and testing procedures (Stelmachowicz & Jestaedt, 1984; Vanden Abeele et al., 1992) so values reported in the literature can not serve as comparative values or as normative data. The Q10 values obtained for the normal listeners in this thesis will be used as the basis or the norm when evaluating Q10 values in the impaired listeners.

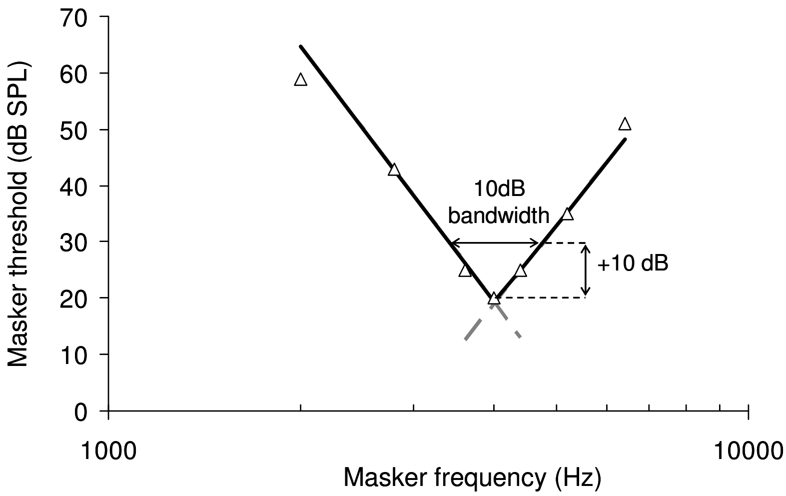


Figure 5.4-5. Schematic representation of Q10-fits. Straight lines are fitted to the data points in order to derive 10 dB-width.

5.4.6 Causes of reduced frequency selectivity

It has been suggested that the loss of frequency selectivity is the most important consequence of cochlear hearing dysfunction and is associated with damage or loss of the outer and inner hair cells(Moore, 2007). The OHCs are responsible for sharp tuning and high sensitivity on the BM and are extremely vulnerable to damage. Disturbances of the function of the hair cells can be caused by a number of problems such as damage or loss of stereocilia and metabolic disturbances.

Hair cell and stereocilia damage

Liberman and Dodds (1984) used noise exposure and ototoxic drugs to produce a variety of types of hair cell damage in cats. They measured neural tuning curves in single neurons, and then traced the hair cells this neuron synapses to see which type of hair cell damage could be associated with specific shapes of tuning curves.

They found that in cases where severe to total OHC damage was present but the IHCs were still intact, neural tuning curves showed an elevated or absent sharp tip resulting in a broad tuning curve (Figure 5.4-6A & B). When both OHC and IHC were damaged, tuning curves were raised and broad (Figure 5.4-6C). Since OHC seem to be more vulnerable to damage than the IHC, the situation where IHCs are damaged but OHC remain intact is rare. However, the situation where severe damage was found in the IHCs but minimal damage in the OHCs was associated with a neural tuning curve almost normal in shape but shifted upward by about 40 dB (Figure 5.4-6D).

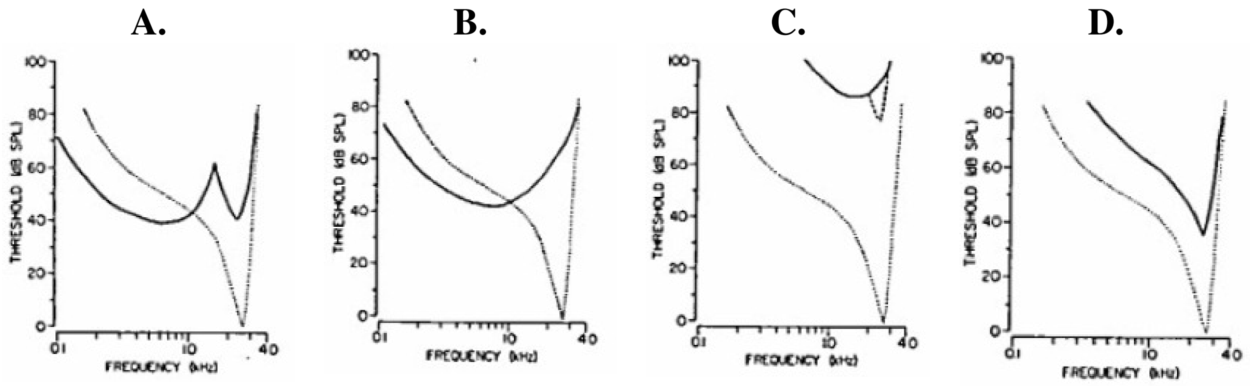


Figure 5.4-6. Schematic representation of neural tuning curves, taken from Liberman and Dodds (1984). The dotted grey line represents a normal tuning curve. The full line represents an impaired tuning curve. A. damaged OHC, intact IHC; B. Total loss of OHC, intact IHC; C. severe damage to both OHC and IHC; D. damaged IHC, minimal damage to OHC.

It has been suggested that damage to the OHC possibly accounts for approximately 50 dB hearing loss at low frequencies and 65 dB at high frequencies (Yates, 1990, Ruggero et al., 1997, Moore, 2007). For more severe losses it is likely that both OHCs and IHCs are affected.

Metabolic disturbances

Changes in the *endocochlear potential* are another possible cause of diminished frequency selectivity. Schmiedt et al. (2002) linked presbyacusis to degeneration of the stria vascularis which leads to loss of endocochlear potential. A change in endocochlear potential will affect the functioning of both the inner and outer hair cells.

Protein deficiencies

Pathologies have been described where the absence of a building brick (e.g. a protein) of the cochlear structure affects the functioning of the cochlea. One example is the genetic renal disease Alport's syndrome. This syndrome is characterized by a deficient formation of a protein called Type IV collagen. The exact effect of this defect in the cochlea is still not very clear but a histological study showed a separation between the basilar membrane and the organ of Corti and cellular in-filling of the tunnel and extracellular spaces of the organ of Corti (Zehnder et al., 2005). This might cause an inadequate adhesion between the cells of the organ of Corti and the basilar membrane resulting in inadequate tuning by the outer hair cells. The effect of this deficiency on the frequency selectivity is illustrated in the hearing profile of one of the impaired listeners (MFr) described in Chapter 8.

5.5 Compression: Temporal Masking Curves

5.5.1 What is compression?

Auditory compression is a result of the basilar membrane responding compressively to sounds (Rhode, 1971; Rhode and Robles, 1974; Ruggero et al., 1997; Rhode and Recio, 2000). This means that an increase in the magnitude of stimulation does not produce a proportional increase in the velocity or displacement of the basilar membrane (BM) vibration. The compression is frequency specific, meaning that the cochlear response of a specific BM site will be compressive only for signals with a frequency similar to the characteristic frequency (CF) of that BM-site, but linear to signals with a frequency an octave below CF. Compression is also level-specific: the

BM response to signals close to CF is only compressive for mid- and possibly high-level tones close to CF. For low-level tones the BM response is linear (Robles et al., 1986; Ruggero et al., 1997). The compression threshold is the lowest intensity at which the basilar membrane responds compressively and is typically found around 35 dB SPL in normal listeners (Plack and Drga, 2003; Stainsby and Moore, 2006). Intensities below this threshold will initiate a linear response.

Estimates of BM nonlinearity can be obtained using forward masking methods. Lopez-Poveda et al. (2003) found compression coefficients in normal listeners ranging between 0.2 and 0.3. A coefficient of 0.2 indicates that an input of 10 dB results in an output of only 2 dB.

5.5.2 Temporal Masking Curves: a measure of compression

The compressive properties of the human cochlear response can be inferred from threshold measurements of masked probe tones. The technique using Temporal Masking Curves (TMC) to estimate BM compression was introduced by Nelson et al. (2001) and subsequently adopted in a large number of studies estimating compression in normal and impaired listeners (Lopez-Poveda et al., 2003, 2005; Plack and Drga, 2003, Plack et al., 2004; Rosengard et al., 2005).

The TMC-method uses a forward masking paradigm and consists of measuring the level of a pure-tone forward masker required to just mask a pure-tone probe as a function of the masker-probe time interval. Nelson and Freyman (1987) showed that the masker threshold to just mask a probe increases as the interval (also referred to as gap) between the masker and the probe is increased. The increase in masker level required to forward mask that probe is caused by either of the following two mechanisms. The first mechanism is the recovery from forward masking that occurs at the probe frequency site of the BM. The second mechanism is the cochlear compression that exists at the probe frequency site of the BM (Oxenham and Moore, 1995, 1997; Plack and Oxenham, 1998). Either or both of the mechanisms can be measured using TMCs.

In the on-frequency condition of the TMC measurement both variables are at play. This condition entails that both the masker and the probe are presented at the same frequency and assumes that the masker and the probe are detected at the same BM site. A compressive response to the masker can occur when listening for the probe. The response to the probe is assumed to be linear because the probe is fixed at a low level. When the masker is presented at low stimulus levels the BM will respond linearly to both the masker and the probe. However as the masker level is increased, the BM will respond compressively to the masker. This non-linear response to the masker will yield a large increase in masker threshold since higher masker levels are required to just mask the probe. The steep increase in masker threshold with gap duration reflects the combined effect of recovery of forward masking and compression. Thresholds in the off-frequency condition, where the masker frequency is different from the probe frequency, are assumed to depend only on the recovery of forward masking. This is because an off-frequency masker an octave below the probe frequency produces a linear response at the probe frequency–site of the BM (Yates, 1990; Yates et al., 1990; Ruggero, 1992; Nelson and Schroder, 1997; Oxenham and Plack, 1997; Ruggero et al., 1997; Moore and Oxenham, 1998; Rhode and Recio, 2000). This difference in slope of the on-frequency masker and the off-frequency TMC is therefore a reflection the amount of compression.

5.5.3 Testing procedure

TMCs are measured for *probe frequencies* (f_p) of 250, 500, 1000, 2000, 4000 and 6000 Hz. In the on-frequency condition, f_p = *masker frequency* (f_m). Off-frequency thresholds are measured for a probe frequency of 6000 Hz and masker frequency (f_m) = 2400 Hz (0.4 * f_p) in normal listeners.

In impaired listeners it is unlikely that masker thresholds can be obtained for this probe-masker combination. The masker levels required to estimate the threshold usually exceed the maximum output of the software, resulting in clipping of the signal and no thresholds can be estimated. Two types of adjustments have been implemented to accommodate this problem. First, masker frequencies in impaired listeners are set closer to the probe frequency, at 0.55 * f_p. Secondly, a lower probe frequency is used in the off-frequency condition. If none of the three masker thresholds can be found at a probe frequency of 6000 Hz and f_m = 0.55 * f_p, a lower probe frequency is used. For some impaired listeners this will imply different probe frequencies have to be explored before a probe frequency is found that will allow the estimation of at least three masker thresholds.

Masker thresholds are measured for nine different *gap durations* (0.01, 0.02, 0.03, 0.04, 0.05, 0.06, 0.07, 0.08 and 0.09 s). The gap duration is defined as the duration of the silence period between the masker offset and the probe onset. *Masker* and *probe durations* are 0.108 s and 0.008 s respectively. In the 250 Hz condition, masker and probe duration is 0.108 s and 0.016 s respectively. The probe duration is increased to allow at least four cycles in the stimulus. The stimuli were ramped with raised cosine onset and offset times of 0.004 s.

For all conditions, the *probe level* is fixed at 10 dB SL. This means the level is set 10 dB above the threshold of the probe. It is important to present the probe at a low level to avoid spectral splatter (Moore, 1981).

The off-frequency masker thresholds are fitted with a straight line using a least squares method. New masker levels are read off from the fit for each gap condition, to obtain a smooth version of the off-frequency data. These new, smooth masker levels will be referred to as the *linear reference*.

The decision to use a linear reference is based on the method suggested by Lopez-Poveda and colleagues (2003). They introduced the idea of only measuring off-frequency data for a high-frequency probe, as opposed to obtaining off-frequency masker thresholds for each probe frequency (Nelson et al., 2001). The reason for this decision is that off-frequency slopes for low probe frequencies were found to be steeper than the off-frequency slopes for high probe frequencies. It was suggested that the cochlear response for low probe frequencies is still compressive for maskers with a frequency well below the probe frequencies. Therefore, the off-frequency condition for low probe frequencies is not a reliable representation of linearity. Since off-frequency maskers for high probe frequencies depend solely on the recovery of forward masking and since it is assumed that the recovery of forward masking is frequency independent, these masker levels can thus serve as a linear reference for all probe frequencies.

5.5.4 The shape of the TMC in normal and impaired listeners

Normal on-frequency TMCs often consist of two or three distinct segments. The segments originate from the non-linear response of the basilar membrane to mid and high level tones close to BF (Robles et al., 1986; Ruggero et al., 1997). At low levels the response is linear. At short gap durations (and therefore low masker levels), a shallow (linear) slope can be observed which progresses into a steep slope at mid-range gaps for which the masker levels are within the range where BM response is compressive (35-80 dB SPL). In some cases the steep slope goes back to being shallow at the longest gap durations (very high masker levels), suggesting a return to linearity.

Off-frequency TMCs represent linear responses and should consist of a single-segment, shallow slope. This single slope reflects the recovery of forward masking as described above. Figure 5.5-1A shows a schematic representation of an on- and off-frequency TMC in a normal listener.

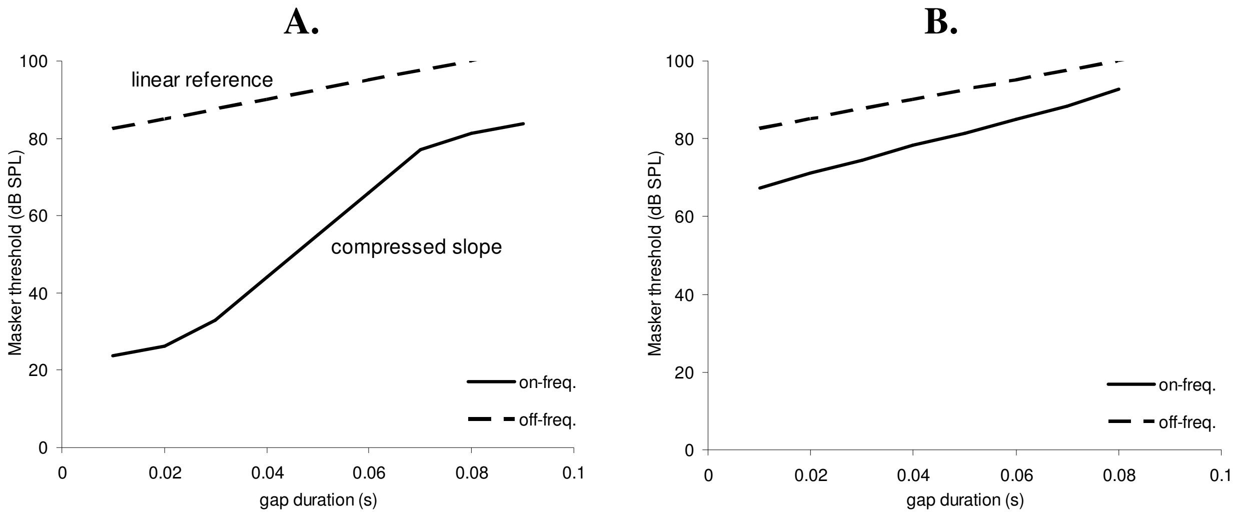


Figure 5.5-1. Schematic representation of on-frequency (full line) and off-frequency TMCs (dotted line) for A. a normal listener, B. impaired listener.

TMCs in impaired listeners have been shown to be shallower than normal TMCs (Nelson et al., 2001, Plack et al., 2004; Lopez-Poveda et al, 2005, Stainsby and Moore, 2006). The most straightforward explanation for this is that the BM responds in a less compressive, and therefore more linear, way in impaired listeners. Figure 5.5-1B shows an example of the on- and off-frequency TMC in an impaired listener

However, Plack et al. (2004) and Lopez-Poveda et al. (2005) have suggested that in a large number of impaired listeners the compression (expressed in a compression coefficient) is unchanged but rather the range over which compression is applied is reduced. In that situation the compression threshold is increased and the normal compression range of 35 to 80 dB SPL is reduced to for example 65 to 80 dB SPL. These findings have been recently confirmed by Jepsen and Dau (2008). Nevertheless, Stainsby and Moore (2006) recently presented results which support the reduced compression coefficient hypothesis. Both theories will be taken into account when quantifying the TMCs found in our listeners. Both the compression coefficient and the compression range will be estimated in evaluating the compression in the listeners.

5.5.5 Quantification of TMCs

In order to compare normal and impaired TMCs it is necessary to quantify the compression in some way. A generally accepted method of quantifying the compression is to derive the input-output functions from the on- and off-frequency masker thresholds. Analysis of the slope of the input-output function subsequently yields a compression ratio for each frequency.

In our experience, it is difficult to obtain useful input-output functions especially in impaired listeners. The hearing loss of the impaired listeners means that masker thresholds are high and the range of levels represented in the I-O-functions is limited. This makes it difficult to analyse and acquire reliable compression parameters. We therefore chose not to use derived input-output functions to quantify the compression in our listeners but to use the slopes of the on- and off-frequency TMCs to estimate the compression coefficient and the compression range.

It needs to be noted that the aim of our TMC measurements reported in the next chapters is not to estimate the characteristics of ‘the’ human cochlea per se. Rather, the TMCs are used in the context of obtaining a general insight into the compression in the system of a particular listener. The compression estimates reported in our normal listeners will be the basis of comparison when evaluating the compression found in our impaired listeners.

Simple analysis techniques were used to acquire a general estimate of compression in our listeners and establish standard compression parameters on the basis of which our impaired data was evaluated. Again, we need to emphasize that at no point was it our aim to find 'the' compression coefficient as is frequently described in the psychophysical literature. The *off-frequency* masker thresholds are fitted with a straight line using a least squares method. New masker levels for each gap are read off from the fit to form a smooth *linear reference* (as described above). The slope of the linear reference will be used in the on-frequency analysis (Figure 5.5-2). The analysis of the *on-frequency* TMC is based on the assumption that a normal on-frequency TMC consists of two shallow sections respectively at low and very high masker levels, and a steep section at mid-level. Therefore, three straight lines are fitted to the on-frequency TMCs. Figure 5.5-2 shows a schematic representation of these three straight lines. The slope of the shallow sections is set equal to the slope of the linear reference (linear slope). Subsequently, the three lines are fitted using a least squares method with the intersects of the three lines with the x-axis and the slope of the steep line (compressed slope) as free parameters. The fits are made using Excel Solver.

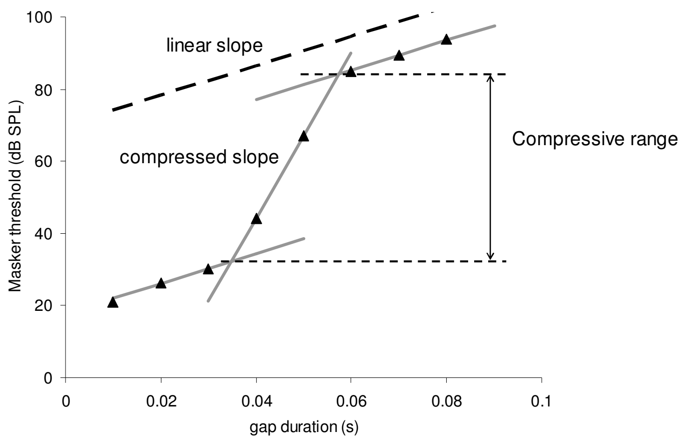


Figure 5.5-2. Schematic representation of the 3-segment fit to the on-frequency TMC. The triangles represent masker thresholds in the on-frequency condition, the grey lines represent the 3 straight lines fitted to the data points. The dotted line is the linear reference fitted to the off-frequency masker thresholds.

The *compression coefficient* is determined by dividing the linear slope by the compressed slope. Steep on-frequency TMCs will generate a small compression coefficient, shallow on-frequency TMCs will result in a large coefficient close to unity.

The *compression range* is determined by the intersection of each shallow slope with the compressed slope. In other words, the lowest compressed value is determined by the masker level associated with the transition from low-level linear slope to compressed slope and the highest value is determined by the masker level related to the transition from compressed slope to high-level linear slope. In some TMCs only one linear slope can be observed, either low or high level. In those cases, the compressed threshold adjacent to the side missing a linear slope will be regarded as the limit of the compressed range.

The analysis of the TMCs measured in 2 normal listeners will be used as a basis to evaluate the compression in impaired listeners and will be discussed in Chapter 6.

5.5.6 Causes of reduced compression

It has been suggested that the process responsible for sharp tuning and high sensitivity on the BM is also responsible for nonlinearity in the auditory system. As described in the previous section (5.4 Frequency selectivity) this process is situated in the OHCs. Measurements of BM vibration in nonhuman mammals have confirmed that interfering with the function of the OHCs, for example by furosemide injections, results in a more linear response to a tone at CF (Ruggero and Rich, 1991).

5.6 Conclusion

This chapter is concerned with describing the measures which will be used when developing the hearing profiles of the normal and impaired listeners. The testing procedure consists of 2 parts: a short clinical screening and an elaborate set of psychophysical tests.

The clinical screening aims to be familiarized with the participant's hearing and to formulate a clinical diagnosis at the start of the participant's testing series. It is as a baseline measure against which the psychoacoustic measures will evaluated.

The psychoacoustic measures look into the sensitivity, frequency selectivity and compression of the listener. Sensitivity measures are absolute thresholds and how these thresholds are a function of the tone duration. It is hypothesized that the difference between the threshold for a long tone and the threshold for a short tone can provide useful information. This idea will be discussed in detail in the next chapter.

The IFMC aims to explore how one sound interferes with another and how sounds are separated from each other in real life situation. Not only does it identify situations where the 'separability' is reduced, it will also identify the situations where tones are detected at a site of the BM whose BF is different than the frequency of the signal. In other words, it identifies the most favourable filter for a specific signal due to off-frequency listening. Allowing for off-frequency listening is the main reason why IFMCs are not directly comparable with the traditional PTC measurements reported in the literature. As mentioned before, the aim of an IFMC is not to measure BM tuning curves but to measure the interference between sounds, a much experienced problem in impaired listeners. We feel that the information on which filters are most sensitive, and therefore doing most of the work, is valuable when modelling the impaired hearing of a listener.

The TMC-measure focuses on identifying compression at different frequencies. In the literature, TMC measurements have been focused on estimating a degree of compression in normal and impaired listeners. We do not aim specifically to measure the compression in our listeners but rather explore whether and where compression is still present. Two compression characteristics are being derived from the TMCs. The compression coefficient is an indicator of the amount of compression at each frequency whereas the compression range represents the intensity levels at which the BM responds compressively. These two parameters are derived in a some what crude fashion and are therefore not a reliable representation of ‘the’ compression in the system. Nevertheless, they do offer a general insight into the compression at each frequency in a listener.

The assessment protocol will initially be tested on normal hearing listeners to evaluate the feasibility of the measures (see Chapter 7). Subsequently, a number of impaired listeners will be tested and their hearing profiles presented (Chapter 8).

CHAPTER 6

Threshold as a function of duration

6.1 Introduction

The threshold for a pure tone stimulus depends on the duration of the stimulus. Thresholds are lower for longer tones than for shorter tones. This dependency of threshold on duration is present for tone durations typically less than 500 ms. In normal listeners, a threshold difference of 2.4-3 dB per octave of duration is typically reported (e.g., Hughes, 1946; Plomp and Bouman, 1959; Sheeley and Bilger, 1964; Olsen and Carhart, 1966; Hall and Fernandes, 1983; Florentine et al., 1988). Hearing-impaired listeners are reported to have shallower threshold/duration functions than normal listeners. A threshold shift of around 0.6-0.9 dB per octave has been described for this group (Wright, 1968, Watson and Gengel, 1969, Gengel and Watson, 1971; Hall and Fernandes, 1983, Florentine et al., 1988, Gerken et al., 1990).

The threshold/duration trade off is commonly known as ‘temporal integration’. It has been argued that this term is rather unfortunate (Meddis, 2006, Viemeister and Wakefield, 1991) since the psychophysical phenomenon (threshold decreases as the duration of the stimulus increases) is being described using the name of the supposed underlying mechanism (a central temporal integrator). Reports in the literature however do not agree on what is being integrated and where this integration takes place (Plomp and Bouman, 1959, Zwislocki, 1960, Viemeister and Wakefield, 1991; Heil and Neubauer, 2003, Neubauer and Heil, 2004, 2008, Meddis, 2006). We endorse the argument that the term ‘temporal integration’ is not the best means of describing the dependence of threshold on stimulus duration and will use the term 'threshold/duration effect' when referring to this psychophysical phenomenon.

The difference in the threshold/duration effect between normal and impaired listeners is potentially important from a diagnostic point of view. This clinical application of the threshold/duration effect, called 'brief-tone audiometry', has been explored by a number of researchers prior to 1990 but has never been accepted as a routine audiological tool. One explanation is that the effect is so variable that the various measures could not be used to discriminate reliably between normal and impaired listeners. It is also possible that only some types of impairment produce a change in the threshold/duration function. Reduced threshold/duration effects were found specifically in listeners with presbycusis, noise-induced hearing loss, cochlear hearing loss and Meniere's disease, while listeners with a conductive hearing loss or an eighth nerve lesion appear to have normal threshold-duration functions (Harris et al., 1958; Pedersen and Elberling, 1973; Young and Kanofsky, 1973; Olsen et al., 1974; Pedersen and Salomon, 1977; Chung and Smith, 1980; Chung, 1982). Unfortunately the variability of the threshold/duration effect among listeners for both normal and impaired listeners and the overlap found between the 2 groups have created considerable uncertainty in the clinical value of brief tone audiometry.

Various methods for quantifying the threshold-duration effect have been described in the literature. A number of researchers looked at the difference in threshold of a short and a long tone (Young and Kanofsky, 1973; Barry and Larson, 1974; Richards and Dunn, 1974; Spence and Feth, 1974, Chung, 1981, 1982). Also, different types of functions have been utilized to fit threshold/duration data. Green et al. (1957) tried fitting a combination of 3 straight lines each with a different slope, while others found that one straight line fitted (most of) the threshold-duration data (Harris et al., 1958; Elliot, 1963; Pedersen and Elberling, 1972; Florentine et al., 1988). Wright (1968) and Watson and Gengel (1969) adopted Zwislocki's method (1960) of fitting an exponential function.

This report investigates the threshold/duration function in normal and impaired hearing and explores different ways of quantifying the threshold/duration effect. The methods described below are quantifying the difference in threshold between the long and the short tone, fitting a broken-fit function, an exponential function and finally a two-parameter reciprocal function. It will be concluded that this last function has potential to be developed into a rapid diagnostic indicator. The validity of the different fitting functions will be established in terms of its ability to fit data for both normal and impaired listeners and its ability to separate normal from impaired listeners.

The threshold/duration functions described in this chapter are based on thresholds for 7 different tone durations. However, it is will be shown that only the shortest and the longest durations are necessary to generate reliable parameter estimates using the reciprocal function. This will lead to the recommendation that clinical threshold measurements should be measured for both long and short duration tones. The difference between these two measurements can subsequently be used for diagnostic purposes.

This report will proceed as follows:

1. Measure threshold/duration functions for normal and hearing impaired hearers
2. Explore four different measures to quantify the threshold/duration effect
3. Show that a reciprocal formula fits all the data well

4. Show that the parameters derived from the reciprocal function distinguish between normal, conduction and sensorineural losses
5. Show that these parameters can be reliably assessed using only two threshold measurements.

6.2 Threshold/duration functions in normal and impaired listeners

The experiment reported below measures the threshold-duration functions in normal and impaired listeners over a range of frequencies. The results confirm findings in previous studies reporting reduced threshold/duration effects in most impaired listeners.

6.2.1 Method

Procedure. Absolute thresholds were measured using the Maximum Likelihood Method described in section 2.3. Participants were asked to indicate whether they heard a target tone or not. Start level was usually set at a level where the participant can hear the target tone, which is a tone intensity of 30 or 40 dB SPL for normal listeners and 70 dB SPL for impaired listeners. The level of the stimulus was adjusted in step sizes of 10 dB SPL until the first reversal (Initial phase). This first reversal is considered as trial 1 and start of the Threshold phase. From that point onwards a psychometric function is fitted after each response and the intensity of the next stimulus is set at the 50 % point of the psychometric function. The first psychometric function is fitted using the last response of the Initial phase and the first response of the Threshold phase.

A psychometric function is fitted after each response in the Threshold phase using Green's maximum likelihood paradigm (1993). A threshold estimate is subsequently produced after each trial corresponding to the 50 % point of the psychometric function. Each threshold run consists of

10 trials (not including the initial trials and the catch trials) and the final threshold (in dB SPL) is set at the 50% point of the last psychometric function.

Twenty percent of the trials were catch trials with no test stimulus presented. If a 'YES' response was given during a catch trial, the run was aborted, the participant informed and the run was restarted. In a block of threshold runs, the catch trial rate is reduced gradually after each flawless run to no less than 10 %. Catch trials are not included in the total trial count. Participants are encouraged to be conservative in their judgments and not indicate the presence of a sound if they are uncertain.

Data collection was divided in three identical blocks. Each block included all four test frequencies, presented at a random order within the block. Within a frequency, durations were also randomized. Data points to be displayed are the mean of three measurements.

Stimulus. Threshold measurements as a function of duration were measured for tones at 250, 1000, 4000, and 8000 Hz. Tone durations ranged from 0.008 to 0.512 s in steps of a factor of 2, with raised cosine onset and offset times of 4 ms. The stimulus was preceded by a cue consisting of a tone identical to the test tone but 10 dB SPL more intense than the test tone.

Listeners. Participants were 8 normal hearing listeners between the age of 20 and 35 and 8 hearing impaired listeners between the age of 22 and 67. None of the normal listeners had a medical history of hearing problems. All but one impaired listener had been fitted with hearing aids at some time in the past.

The left ear was tested in most cases. The right ear was tested for four impaired listeners because their left ear had either normal or very high thresholds. All listeners received adequate training prior to the actual data collection. Absolute thresholds for 500-ms tones at a range of frequencies are shown in Figure 6.2-1 for the normal and impaired listeners.

Apparatus. Stimuli were generated using Matlab at a sampling rate of 96000 Hz, with 24-bit resolution. They were played monaurally via the workstation headphone connection through a pair of circumaural Sennheiser HD600 to the participants seated in a double-walled sound attenuating booth.

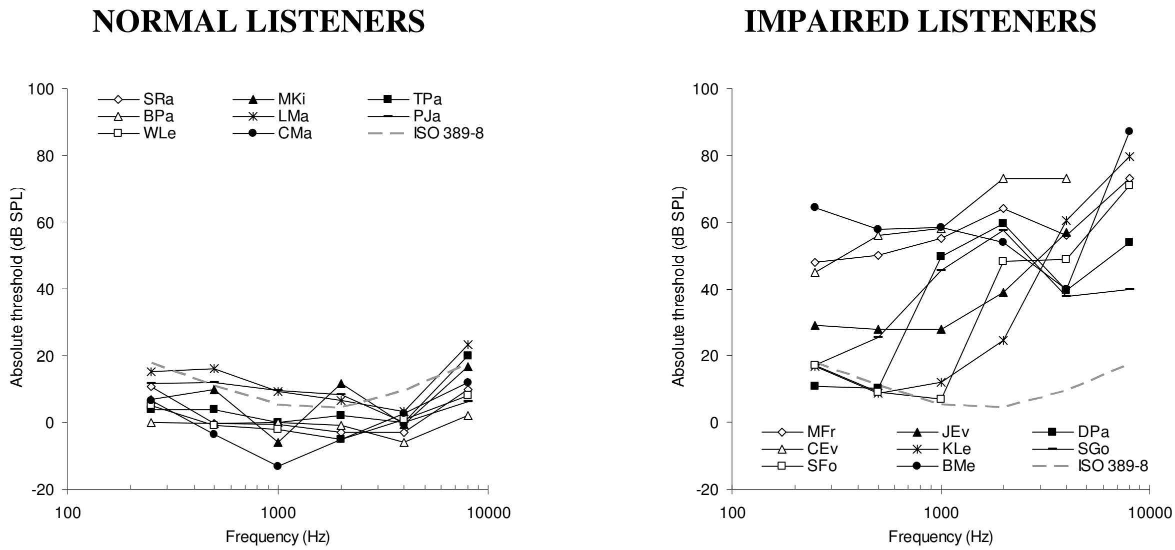


Figure 6.2-1. Absolute thresholds for normal and impaired listeners for a 500-ms tone at frequencies 250, 500, 1000, 2000, 4000 and 8000 Hz. The grey dotted line represents standards thresholds as specified in ISO 389-8.

6.2.2 Results

Figure 6.2-2 shows the threshold/duration data for the normal and impaired listeners. In three impaired listeners (JEv, CE and BMe) no threshold/duration function could be obtained for the 8000 Hz condition due to stimulus levels exceeding the clipping threshold of the soundcard.

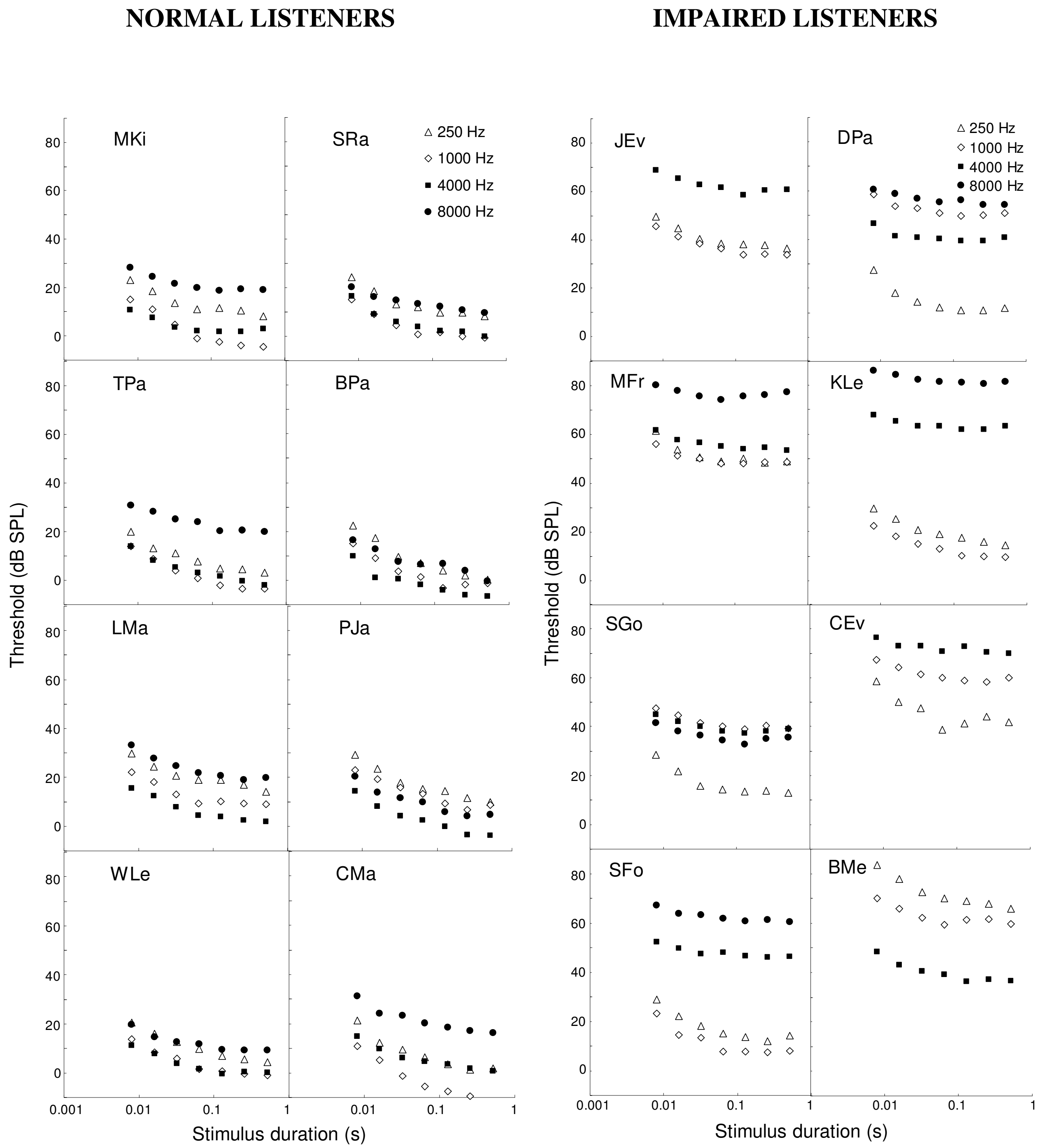


Figure 6.2-2. Thresholds as function of duration for 8 normal listeners (left hand panel) and 8 impaired listeners (right hand panel). Data points represent mean thresholds at 7 durations (0.008-0.512 s in octave steps) at 4 different frequencies (250, 1000, 4000, and 8000 Hz).

The functions of the impaired listeners are raised compared to the functions of the normal listeners. Furthermore most of the impaired listeners show functions which are shallower than those of the normal listeners. The overall mean threshold decrease between a 0.008 s tone and a 0.512 s tone is 15.3 dB SPL (SD = 2.9 dB SPL) in normal listeners and 8.9 dB (SD = 2.5 dB SPL) for the impaired listeners. This equates to a decrease in threshold of respectively 2.6 and 1.5 dB/octave. This threshold shifts for normal and impaired listeners correspond with data reported in the literature (Wright, 1968; Pedersen and Elberling, 1972; Spence and Feth, 1974, Florentine et al., 1988). The fact that the threshold/duration effect is smaller for impaired listeners than for normal listeners implies that impaired listeners experience less benefit when a tone is made longer.

6.3 Using Threshold/duration functions as a diagnostic tool

Using the data reported in the previous section, we explored different measures to quantify the threshold/duration effect in normal and impaired listeners. Our aim is to find a measure which will allow us to differentiate between normal and impaired listeners. The threshold/duration effect will be quantified by looking at the difference between the threshold of a long tone and of a short tone, and by fitting an exponential function, a broken-stick function and a two-parameter reciprocal function.

6.3.1 Threshold difference between a short and long tone

An analysis of the threshold/duration functions reported above has shown that the threshold difference between a short and a long tone (threshold/duration effect) is smaller in impaired listeners than in normal listeners across frequencies. The difference in threshold between a short and a long tone has been used by a number of researchers in the past to quantify the threshold/duration effect (Young and Kanofsky, 1973; Barry and Larson, 1974; Richards and Dunn, 1974; Spence and Feth, 1974, Chung, 1981, 1982). All report a difference between the normal and impaired group but most also mention a large variation between individuals. Only Barry and Larson (1974) reported that hardly any overlap between normal and impaired listeners was found. The threshold/duration effects reported in the previous section will be analysed in more detail to explore the possibility that the threshold difference between a short and a long tone is a useful measure to differentiate between normal and impaired listeners.

Table 6.3-1 shows the mean threshold difference (in dB/octave) across listeners for the 4 test frequencies (250, 1000, 4000, and 8000 Hz). Across all frequencies a smaller threshold difference is found in the impaired group compared to the normal group. At 250 Hz, the difference is rather small which can be explained by the fact that some listeners had normal thresholds at this frequency and therefore are expected to yield normal threshold differences. These normal estimates will subsequently affect the overall mean at that particular frequency.

Table 6.3-1. Mean threshold difference (in dB/oct) for normal (NH) and impaired listeners (IH) at frequencies 250, 1000, 4000, and 8000 Hz, based on the threshold difference between the shortest and the longest tone.

	Frequency			
	250 Hz	**1000 Hz**	**4000 Hz**	**8000 Hz**
NH	-2.9	-2.8	-2.4	-2.1
IH	-2.5	-1.7	-1.2	-0.9

Figure 6.3-1 shows the individual threshold differences (dB/octave) of the normal (filled triangles) and impaired (open circles) participants at 250, 1000, 4000, and 8000 Hz. The full and dotted lines are the mean threshold differences for the normal and the impaired group respectively, as already presented in Table 6.3-2. At all frequencies there is a large variation within the groups. Only at 8000 Hz are the values more grouped per condition. At 250, 1000 and 4000 Hz there is considerable overlap of threshold difference values between the groups. At 8000 Hz however there is a distinct separation between the normal and impaired listeners.

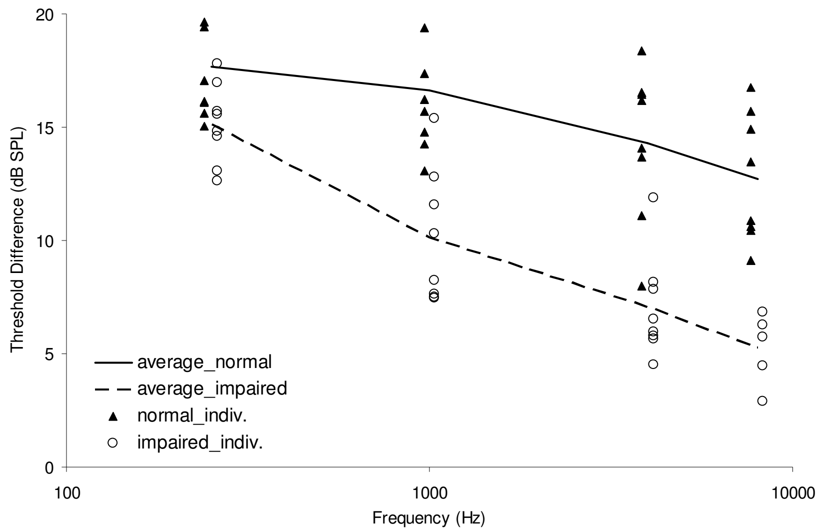


Figure 6.3-1. Threshold difference (in dB/oct) for normal and impaired listeners at 250, 1000, 4000, and 8000 Hz. The threshold difference is based on the difference in threshold between the shortest and the longest tone. Filled triangles correspond to individual threshold differences at the 4 frequencies for normal listeners. Open circles correspond to individual threshold differences of impaired listeners. The full line and the dotted line represent mean threshold differences for respectively the normal and impaired group.

It can be concluded that the mean threshold differences between a short and a long tone indicates a difference between the normal and the impaired group. Nevertheless, the individual differences within the normal and impaired group and the overlap between the 2 groups casts doubts on the usefulness of this measure as a diagnostic tool.

6.3.2 Fitting an exponential function

Zwislocki (1960) suggested fitting an exponential function to the threshold/duration data to subsequently estimate the time-constant involved in the decrease of threshold as a function of duration. This method was adopted by Wright (1968) and Watson and Gengel (1969).

An exponential function will be fitted to the threshold/duration functions reported in section 6.2 and estimates of the time-constant *tau* will be produced.

Method

An exponential function of the form

$$y = -10 * \log(1-e^{-d/tau}) \qquad (1)$$

was fitted to the threshold/duration data with *d* as the duration of the stimulus and *tau* as a free parameter, using a least squares best fit procedure executed in Excel Solver. The exponential function was found to provide a reasonable fit to the data (mean RMS error = 1.3) in most conditions. The function did not provide a satisfactory fit in all occasions, in particular steeper functions were not predicted accurately by the exponential function.

Results

Figure 6.3-2 shows the individual *tau*-estimates of the normal (filled triangles) and impaired (open circles) participants at 250, 1000, 4000, and 8000 Hz. The full line and the dotted line are the mean threshold differences for the normal and the impaired group respectively. The mean *tau*-estimate per frequency is lower for the impaired listeners than for the normal listeners. This indicates that the decrease of the thresholds as a function of stimulus duration is slower in the impaired group. Unfortunately, a large variation within the groups is seen at all frequencies and considerable overlap in the *tau*-estimates is present at almost all frequencies.

It is concluded that fitting a exponential function to the threshold/duration data is not a satisfactory method to quantify the threshold/duration effect due to the large variation within the groups and the overlap between the normal and the impaired group. Moreover, the exponential function did not provide a satisfactory fit for all functions.

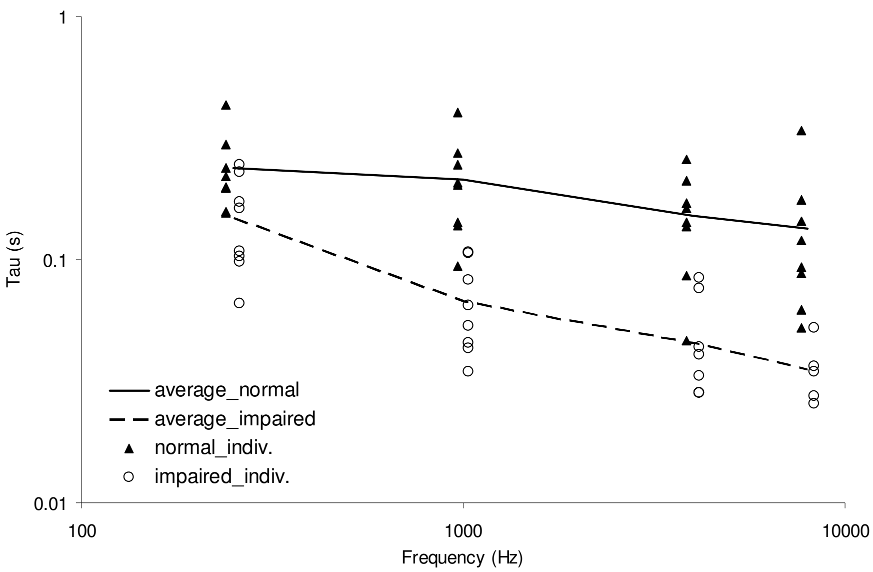


Figure 6.3-2. Time constant *tau*-estimates (in s) for normal and impaired listeners at 250, 1000, 4000, and 8000 Hz. Filled triangles correspond to individual values for normal listeners at the 4 test frequencies. Open circles correspond to individual values of impaired listeners. The full line and the dotted line represent mean *tau*-estimates for respectively the normal and impaired group at 250, 1000, 4000 and 8000 Hz.

6.3.3 Fitting a broken-stick function

Visual exploration of the data revealed that almost all threshold/duration functions consist of a sloping segment followed by a horizontal segment equivalent to the threshold for long durations. Therefore, a broken-stick function was fitted to the mean thresholds as a function of duration at each frequency for each participant. The sloping segment is subsequently used to quantify the threshold/duration effect based on the assumption that the horizontal segment does not contribute to the threshold/duration effect.

Method

The broken-stick function is a combination of a sloping line characterized by equation (3) and a horizontal line, characterized by equation (4). A schematic representation can be found in Figure 6.3-3. The broken stick function was fitted to the threshold/duration data with s, c and h as free parameters, using a least squares best fit procedure executed in Excel Solver.

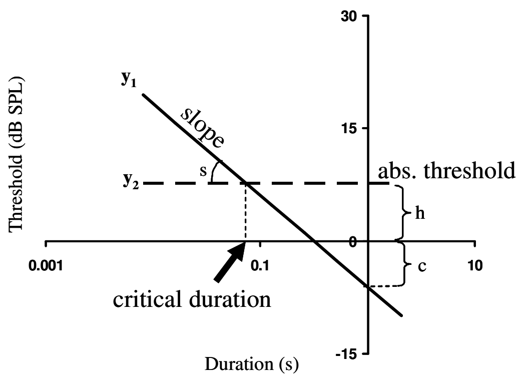


Figure 6.3-3. Schematic representation of the broken-stick function, combining the functions y_1 and y_2 .

The broken-stick function (Fig. 5.3-3) can be described as:

$$y = Max\,(y_1, y_2) \qquad (2)$$

where

$$y_1 = s*log(d)+c \qquad (3)$$

with s =slope
d =duration of stimulus
c = constant

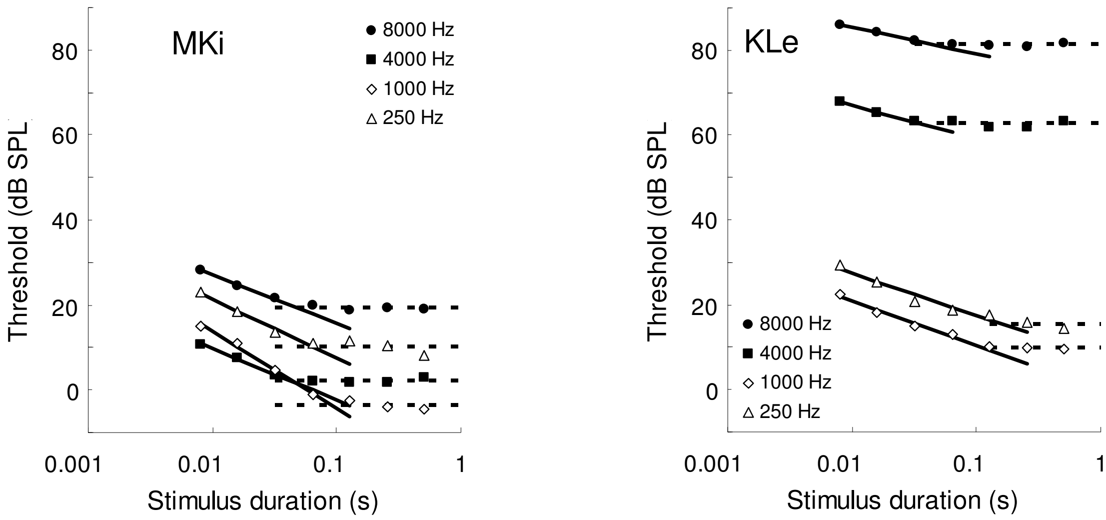
$$y_2 = h \qquad (4)$$

with h = absolute threshold for long duration

The broken-stick function was found to provide a good fit to the data (mean RMS error = 0.8). Figure 6.3-4 shows an example of the function for the threshold/duration data in a normal and an impaired listener and the parameters related to these fits. The slope in dB/oct is derived from the parameter s (in dB/decade) and the absolute threshold is directly taken from parameter h.

Table 6.3-2 shows the mean slopes per octave for normal and impaired listeners. Note that the slopes are negative since threshold decreases with increasing signal duration. For all frequencies but one the normal hearing listeners were found to have a steeper (i.e. more negative) slope than the impaired listeners. The values found in both groups are higher than reported in the literature. The exclusion of the horizontal section in calculating a slope is probably the cause of this observation.

At 250 Hz, the mean slope is *steeper* for the impaired group compared to the normal group. This can be partly explained by the normal threshold at this frequency for a number of the impaired listeners, resulting in a normal threshold/duration effect at this frequency.



	Frequency (Hz) 250	1000	4000	8000
Slope (dB/oct)	-4.2	-5.5	-3.6	-3.4
Abs.thres (dB SPL)	10	-3.6	2.1	19.2

	Frequency (Hz) 250	1000	4000	8000
Slope (dB/oct)	-3.0	-3.2	-2.3	-1.9
Abs.thres (dB SPL)	15.2	9.9	62.6	81.2

Figure 6.3-4. Example of broken-stick functions fitted to the threshold/duration data of a normal and an impaired listener. The straight line represents the sloping section formalized in equation (2), the dotted line corresponds to the horizontal segment of the function, associated with the absolute threshold (in dB SPL) for long durations. The tables show the estimates for the slope (in dB/oct) and the absolute threshold for each frequency generated from the data above.

Figure 6.3-5 shows the individual slopes of the normal (filled triangles) and impaired (open circles) participants at 250, 1000, 4000, and 8000 Hz. The full and dotted lines are the mean slopes for the normal and the impaired group respectively, as already presented in Table 6.3-1. Considerable variation can be seen in both groups with an overlap of slope values between the groups. It is concluded that the slope of the threshold/duration function as measured using the broken stick function does not provide us with a reliable measure to differentiate between normal and impaired listeners.

Table 6.3-2. Mean slope (in dB/oct) for normal (NH) and impaired (IH) listeners at 250, 1000, 4000 and 8000 Hz.

	Slope (in dB/octave) 250 Hz	1000 Hz	4000 Hz	8000 Hz

IH	-5.1	-3.8	-2.9	-2.1

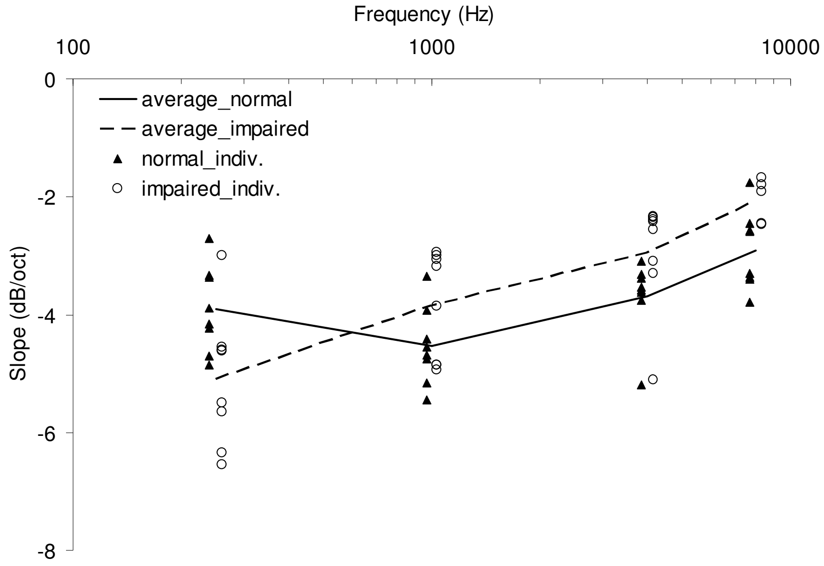


Figure 6.3-5. Slopes (in dB/octave) for normal and impaired listeners at 250, 1000, 4000, and 8000 Hz. Filled triangles correspond to individual slopes at the 4 aforementioned frequencies for normal listeners. Open circles correspond to individual slopes of impaired listeners. The full and dotted line represent mean slopes for respectively the normal and impaired group at 250, 1000, 4000, and 8000 Hz.

6.4 A two-parameter reciprocal fit to threshold-duration functions in normal and impaired listeners.

In the previous sections, three different measures were evaluated on their potential to differentiate between normal and impaired listeners. It was shown that neither of the fitting procedures separated normal from impaired listeners satisfactorily. In the next section a two-parameter reciprocal function will be presented. The two parameters *G* and *A* will be shown to have a diagnostic value and to provide useful data when developing a computer model of an individual's impaired hearing.

6.4.1 Predicting the threshold/duration effect in normal and impaired listeners

The threshold/duration data can be fitted with a function based on the formula

$$P_{thr} = (0.69/ d + A)/ G \qquad (5)$$

Where P_{thr} is the threshold (in μPa), d is the duration of the stimulus (in s), A (in s^{-1}) and G (in $\mu Pa^{-1} * s^{-1}$) are free parameters. The equation has been derived as follows: the reciprocal of the duration represents the decrease of threshold with increasing signal duration. The A parameter is added to accommodate for the vertical position of the function and finally G serves as a scalar to assure both sides of the equation have the same unit (i.e. μPa). The value 0.69 (natural logarithm of 0.5) is an arbitrary constant which is not critical to the work presented in this chapter, but is related to on-going theoretical research not presented in this thesis. This value of 0.69 is used consistently across different research projects but a value of 1 would have been equally suitable.

The formula offers a two-dimensional analysis of the threshold/duration functions. Different combinations of the parameters or dimensions give rise to different shapes of a threshold/duration function. Table 6.4-1 (also illustrated in Fig. 6.4-1) shows the use of this formula for three different combinations of the parameters G and A. The threshold/duration functions yielded by each parameter combination are typical for threshold/duration functions observed in normal and impaired listeners. The thresholds have been converted from μPa to dB SPL using a peak amplitude of 28 μPa at 0 dB SPL as a reference.

The first column in Table 6.4-1 shows the stimulus duration, the second column shows the thresholds predicted when G = 0.6 and A = 15. The threshold falls from 15.6 dB SPL at a duration of 0.008 s to -0.2 dB SPL at 0.512 s. The threshold is approaching an asymptote of 0 dB

SPL (filled diamonds in Fig. 6.4-1). This situation is considered typical for a normal threshold/duration function.

	♦	○	▲
G	**0.6**	**0.02**	**0.02**
A	**15**	**15**	**60**
Duration	**Threshold (dB SPL)**		
0.008 s	15.6	45.1	48.3
0.016 s	10.8	40.3	45.3
0.032 s	6.8	36.3	43.3
0.064 s	3.7	33.3	42.0
0.128 s	1.7	31.2	41.3
0.256 s	0.5	30.0	41.0
0.512 s	-0.2	29.3	40.8

Table 6.4-1. Threshold (dB SPL) as a function of duration for 3 combinations of parameters *G* and *A*.

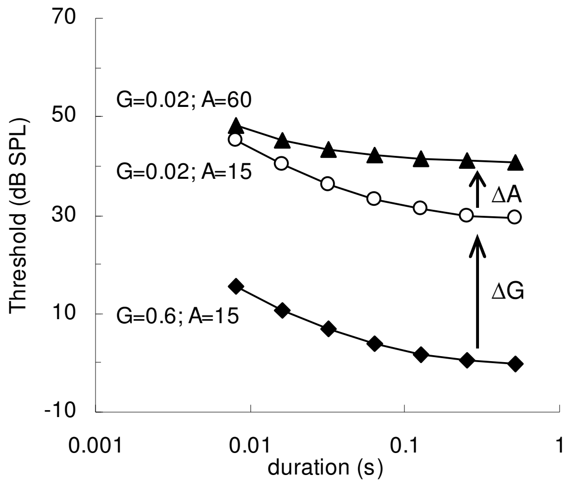


Figure 6.4-1. Thresholds as a function of duration for 3 combinations of parameters *G* and *A*, as presented in Table 6.4-1.

Column 3 in Table 6.4-1 shows how a decrease in the *G parameter alone* leads to an overall increase in thresholds. This can be seen in Fig. 6.4-1 as a simple upward shift in the function (open circles) relative to the normal threshold/duration function. The difference between the highest and the lowest threshold (16 dB) is, however, unchanged from the normal function. Column 4 in Table 6.4-1 shows how a *combined* change in parameters *G* and *A* affects the thresholds (filled triangles in Fig. 6.4-1). The function is shifted upwards as expected from a reduction in parameter *G*. However, the difference between the highest and the lowest threshold (7.5 dB) is reduced compared to the other two functions. This reduction in the size of the threshold shift is the result of the change in the parameter *A*. These two latter patterns are observed in 'impaired' threshold/duration functions. In principle, a fourth function could be generated showing the consequences of changing only parameter *A* and leaving *G* unchanged but this pattern is not expected to occur in practice.

6.4.2 Evaluation of the fitting formula

The threshold/duration data for normal and impaired listeners presented in section 6.2 was used to illustrate the suitability of the reciprocal function as a predictor of the threshold/duration effect.

The reciprocal function was fitted to the threshold/duration data of normal and impaired listeners presented in section 6.2 with *A* and *G* as free parameters using a least squares best fit procedure executed in Excel Solver.
Figure 6.4-2 shows the threshold/duration data for 8 normal and 8 impaired listeners. The continuous black lines show the fitted functions for each of the 16 listeners at the 4 test frequencies. The functions provide a good fit to the data in all cases yielding a mean RMS error of 0.9 dB SPL.

The parameter estimates obtained with each fit show a distinctive difference between normal and impaired listeners. Figure 6.4-3 shows the mean *G* and *A* parameters for the normal and the impaired group.

The mean *G* parameter is reduced in impaired listeners and is seen as the main indicator of the *raised* threshold/duration functions in the impaired listeners. There is some overlap in the *G* parameter between the normal and impaired group at 250 and 1000 Hz but at 4000 and 8000 Hz the normal and the impaired group are clearly separated.

The mean *A* values are clearly raised in the impaired group and reflect the *shallower* threshold/duration function seen in most of the impaired listeners. There is more overlap in the *A*-estimates between the normal and the impaired group of listeners. This is caused by a wide spread of *A* values in the impaired group. A more extensive discussion of this observation will follow later in this chapter.

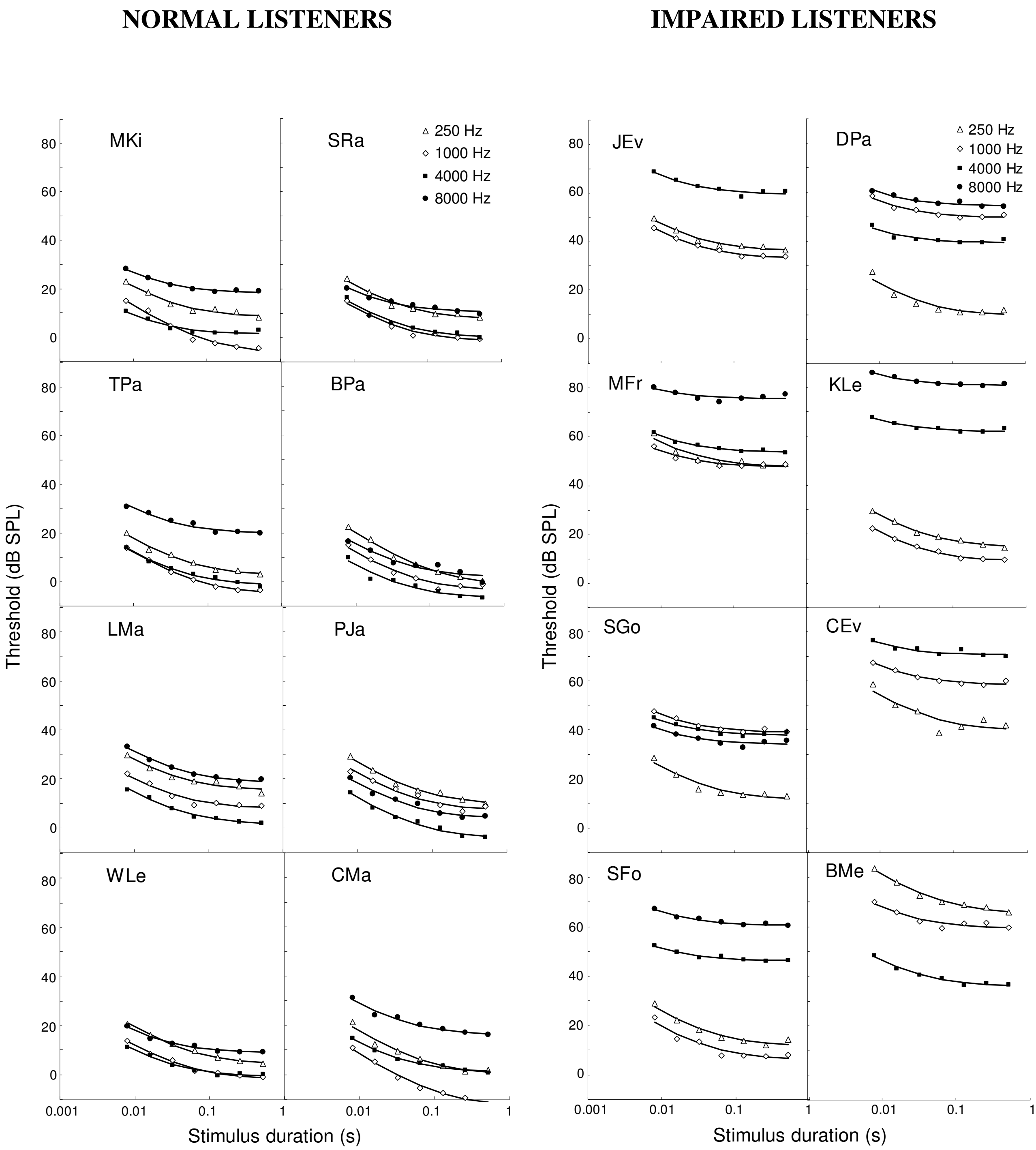
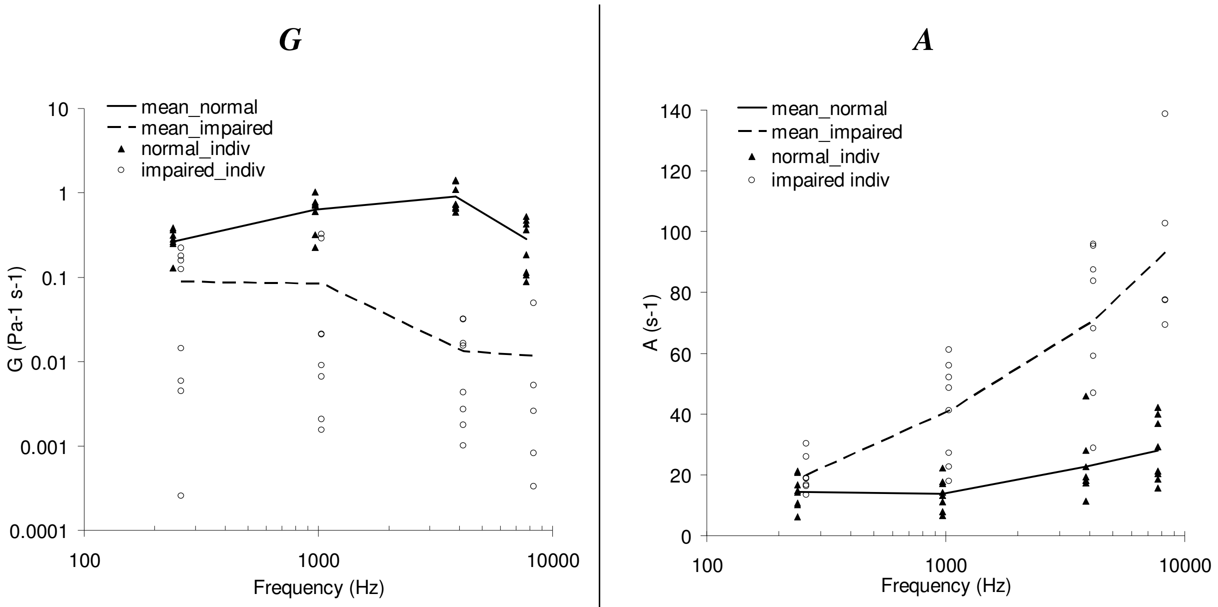


Figure 6.4-2. Threshold/duration functions for 8 normal listeners (left hand panel) and 8 impaired listeners (right hand panel). Data points represent thresholds at 7 durations (0.008-0.512 ms in octave steps) at 4 different frequencies (250, 1000, 4000, and 8000 Hz). The full lines represent predictions made using equation (5). Each fitted function provides an estimate of parameters *A* and *G*.



G

	Frequency (Hz)			
	250	1000	4000	8000
Normal				
Mean	0.27	0.64	0.92	0.29
SD	0.10	0.26	0.34	0.18
Impaired				
Mean	0.09	0.08	0.01	0.01
SD	0.09	0.14	0.01	0.02

A

	Frequency (Hz)			
	250	1000	4000	8000
Normal				
Mean	14.3	13.8	22.6	28.0
SD	5.2	5.3	10.5	10.5
Impaired				
Mean	19.8	40.9	70.7	93.2
SD	5.5	16.3	24.4	28.4

Figure 6.4-3. Individual and mean *G* and *A* estimates for 8 normal (filled triangles) and 8 impaired listeners (open circles). Full lines represent the mean estimate for normal listeners, dotted lines represent the mean estimate for impaired listeners. Left hand panel: individual and mean *G* estimates + associated standard deviations, Right hand panel: individual and mean *A* estimates + associated standard deviations.

In general, it is concluded that low *G* values and high *A* values are associated with impaired threshold/duration data. However, high *A* values observed in the impaired group will also be shown to be of diagnostic value.

6.4.3 Diagnostic value of *G* and *A*

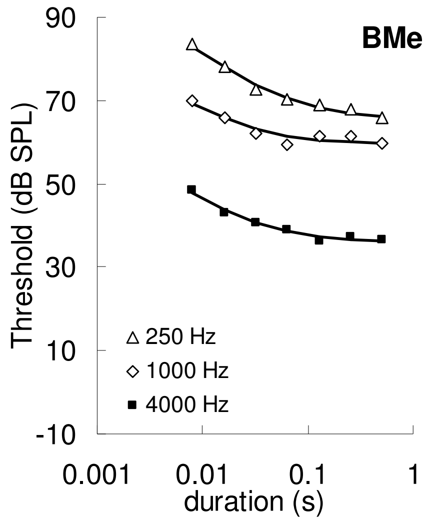
The mean parameter estimates in the impaired group conceal individual differences which might illuminate the origin of the impairment. It is hypothesized that the threshold/duration function not only offers a simple method to differentiate between normal and impaired hearing but will also allow for distinguishing two different groups within the impaired group. The two impaired patterns found in the threshold/duration functions both show raised thresholds. However, in one case the fall in threshold is the same as for normal listeners. In the other case, thresholds are raised combined with a reduced decline in threshold between the shortest and the longest tone duration relative to the normal pattern.

A raised function with a normal threshold/duration effect is characterized by a raised *G* parameter but a low *A* value. In the literature these raised and steep threshold/duration functions have been linked to conductive hearing loss or auditory neuropathy (Harris et al., 1958; Pedersen and Elberling, 1973; Young and Kanofsky, 1973; Olsen et al., 1974; Pedersen and Salomon, 1977; Chung and Smith, 1980; Chung, 1982).

A clear illustration of this pattern can be seen in the data of one of the impaired participants (BMe). This participant has been diagnosed with a conductive hearing loss. Figure 6.4-4 shows the threshold/duration functions of this person, in combination with the *G* and *A* estimates obtained by fitting the reciprocal function.

The functions are clearly raised relative to normal functions but show a steep decrease of threshold with increasing duration. The raised functions are reflected in very low *G* values. The threshold decrease ranges from 10 to 16 dB SPL, explaining the low *A* values.

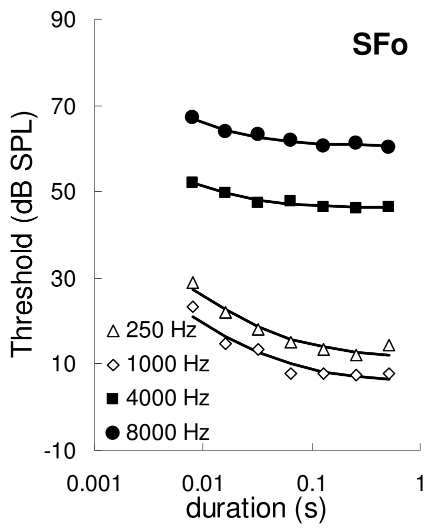
The *A* estimate at 1000 Hz is higher than the estimates at 250 and 4000 Hz. This is possibly related to a minor sensorineural problem, which will be discussed in Chapter 8 (Psychoacoustic profile of impaired listeners, BMe). Nevertheless this value might just represent normal variability within the sample.



	Frequency (Hz)		
	250 Hz	**1000 Hz**	**4000 Hz**
G	0.0003	0.0016	0.0165
A	13	41	29

Figure 6.4-4. Illustration of threshold/duration function and parameter estimates *G* and *A* associated with a conductive hearing loss (participant BMe).

The second and frequently observed pattern is a raised and shallow threshold/duration function. In this case both *G* and *A* parameters are very divergent from the values observed in normal listeners. An example is shown in figure 6.4-5.



	Frequency (Hz)			
	250 Hz	**1000 Hz**	**4000 Hz**	**8000 Hz**
G	0.157	0.325	0.015	0.003
A	17	18	87	77

Figure 6.4-5. Illustration of threshold/duration function and parameter estimates *G* and *A* associated with a senosorineural hearing loss (participant SFo).

Participant SFo has been diagnosed with a high-frequency sensorineural hearing loss. Figure 6.4-5 demonstrates normal threshold/duration functions at 250 and 1000 Hz and significantly raised and shallow functions at 4000 and 8000 Hz. The parameters for the test frequencies 250 and 1000 Hz are similar to the values associated with the theoretical 'normal' threshold/duration function shown in Figure 6.4-1. The two impaired, high frequency-functions generate very low *G* estimates and high *A* estimates.

The two examples described above illustrate the possibility that the parameters might differentiate between of hearing losses within the impaired group. Furthermore it could also identify specific types of losses at different frequencies in a single patient. This could be useful in the case where a hearing loss consists of a conductive and/or a sensorineural component at different frequencies. Identifying parameters per frequency will highlight frequencies where only the conductive component is present.

In the next section a further exploration of the parameters will be presented and suggestions will be made as to what the cut off points are to distinguish between normal and impaired values.

6.4.4 Standardization of G and A parameters

If *G* and *A* are to be useful clinical measures, a standardization of the values is necessary. Figure 6.4-6 shows *G* and *A* values as a function of absolute threshold (threshold at 512 ms, in dB SPL). In the left hand panel, the *G* values obtained for each test frequency in each listener (normal and impaired) are plotted against the threshold for the 512 ms tone at that specific frequency. The right hand panel shows a similar plot for the *A* parameter. For the group of normal listeners this results in 32 *G* and *A* estimates (8 listeners * 4 frequencies), for the impaired group 29 estimates of both parameters are available (three listeners had no data at 8000 Hz).

It is apparent that *G* and *A* are independent parameters. The *G* estimate is highly predictable on the basis of the absolute thresholds and therefore offers no new information over and above these thresholds. A raised threshold is always associated with a reduced *G*-value. The *A* estimate however is highly variable particularly in the impaired group where it is not related to absolute threshold. This *A* parameter can therefore be a source of new information that is additional to absolute threshold enabling a further differentiation within the impaired group.

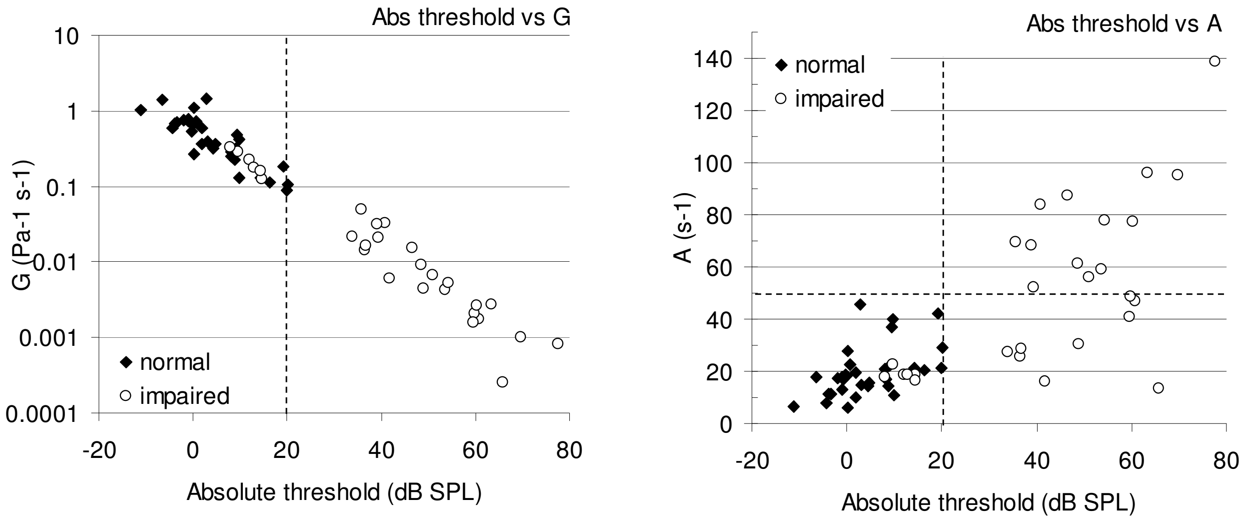


Figure 6.4-6. *G* values (left hand panel) and *A* values (right hand panel) as a function of absolute threshold (threshold for 0.512 s tone). Filled diamonds represent observations associated with normal listeners, open circles represent observations associated with impaired listeners.

Based on the assumption that thresholds below 20 dB SPL can be considered normal at all 4 test frequencies, it is concluded that *G* parameters above 0.1 Pa^{-1} $s^{-1}$and *A* parameters below 50 $s^{-}1$ are associated with normal thresholds. Low *G* values (below 0.1 Pa^{-1} s^{-1}) and high *A* values (above 50 s^{-1}) will therefore be considered as abnormal in the remainder of this thesis. The choice of the normal ranges is arbitrary and might require adjusting in the future.

The criteria set above allow for the exploration of 4 *G*-*A* combinations, i.e.

- normal *G*/normal *A* (Quadrant I, see Fig. 6.4-7)

- low *G*/normal *A* (Quadrant II)
- low *G*/high *A* (Quadrant III)
- normal *G*/high *A* (Quadrant IV)

Figure 6.4-7 shows the 4 quadrants associated with these 4 combinations when plotting *G* against *A*. Each data point represents the parameter *G* as a function of parameter *A* for one frequency. There are 32 observations for normal listeners and 29 observations for impaired listeners (listeners * frequencies).

Observations associated with normal hearing are almost exclusively contained to Q*uadrant I*. This quadrant also includes some observations of impaired listeners. These data points are associated with frequencies where normal thresholds are found for these impaired listeners. Quadrant II and III constitute only of observations associated with impaired listener. The data points included in *Quadrant II* are related to raised threshold/duration functions with a normal threshold shift between the shortest and the longest tone. Raised functions with a reduced threshold/duration effect will have parameters situated in *Quadrant III*.

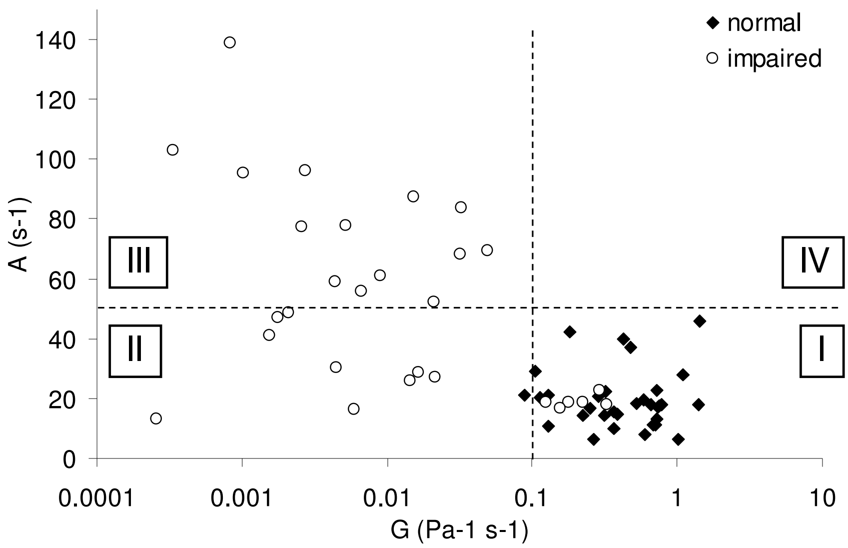


Figure 6.4-7. *A* as a function of *G*. Four quadrants can be distinguished relating to: (I) normal *G*/normal *A* (normal), (II) low *G*/normal *A* (conductive HL), (III) low *G*/high *A*, (IV) normal *G*/high *A*. Filled diamonds represent normal listeners, open circles represent impaired listeners.

The combination where a normal *G* is associated with a high *A* (Quadrant IV) did not show up in the data set described here. However on-going research (not reported here) has revealed a listener with threshold/duration parameters in Quadrant IV. For this specific listener, most observations were situated in Quadrant I. However at one frequency the *A* parameter was distinctly raised while the *G* parameter is normal, situating this observation in Quadrant IV. Further inspection of other psychoacoustic measures at this specific frequency revealed an anomaly in the frequency-selectivity-measure. This suggests that, although this listener can undoubtedly be classified as a normal listener, the *G* and *A* parameters revealed an anomalous observation which was confirmed in subsequent measures.

6.4.5 Parameters *G* and *A* in a simulated conductive hearing loss

To demonstrate the change in *G* and *A* parameters in a more clinical context, a conductive hearing loss was simulated in a normal listener

Method

Thresholds as a function of duration were measured in a normal listener with and without an earplug. First, the baseline/normal threshold/duration functions were established for 4 different frequencies (250, 1000, 4000, and 8000 Hz) in the left ear. The test conditions were similar to those reported in section 6.2. Secondly, the participant placed an earplug in the left ear and the threshold measurements were repeated.

Results

The mean threshold shift due to the presence of the earplug is 20 dB across all frequencies. The threshold/duration functions in the simulated condition (CL) are shifted upwards but retain a threshold/duration effect similar to the normal functions (Figure 6.4-8A).

The *G* estimates for the 4 test frequencies are distinctly reduced compared to the normal *G* estimates. The *A* estimates however shows no change compared to the normal *A* estimates (Figure 6.4-8B). The estimates would place this 'impaired' listener in Quadrant II for all 4 test frequencies. The parameter estimates also agree with the parameters presented in Figure 6.4-4 associated with a conductive hearing loss.

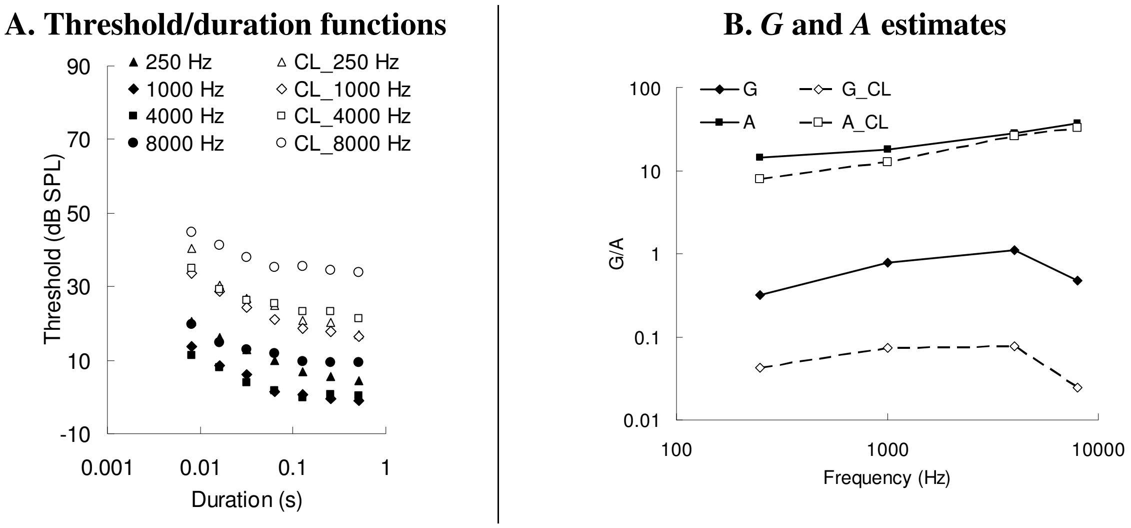


Figure 6.4-8. A. Threshold/duration functions in a single listener with normal hearing (filled symbols) and with a simulated conductive hearing loss (open symbols). The test frequencies are 250, 1000, 4000 and 8000 Hz. B. *G* and *A* parameters associated with the threshold/duration functions representing the normal condition (filled symbols, full line) and the conductive loss-condition (open symbols, dotted lines) at the 4 test frequencies.

6.4.6 Does the formula fit when only 2 thresholds are available?

The formula fits and the associated *G/A* parameters described in this chapter are all based on threshold/duration functions comprising of 7 data points, i.e. thresholds for 7 different tone durations at one specific frequency. From a clinical perspective, an important question emerges as whether it is possible to obtain similar parameters estimates with fewer measurements. An additional analysis on the threshold/duration data of 4 normal and 4 impaired listeners shows that equivalent results can be obtained when using only the thresholds for the shortest (0.008 s) and the longest (0.512 s) tones to fit the reciprocal function and estimate *G* and *A*.

Method

The *G/A* parameters acquired by fitting the reciprocal function using all 7 available thresholds are compared to *G/A* parameters generated by fitting the function only to the thresholds for the long tone (0.512 s) and the short tone (0.008 s). The intermediate thresholds are not included in this fit. The error arising by excluding these five intermediate thresholds from the fit is expressed in a percentage of the initial *G/A* parameter estimates.

Results

Table 6.4-2 shows the mean *G* and *A* estimates using seven or two thresholds to fit the function for normal and impaired listeners. The parameter estimates obtained by using only two thresholds are very similar to the initial parameters using all seven thresholds.

Table 6.4-2. Left hand panel: mean *G* estimates obtained by using 7 or 2 data-points of the threshold/duration functions. Means are based on estimates of 4 normal and 4 impaired listeners. Right hand panel: same as left hand panel but for *A* estimates.

G

	Frequency (Hz)			
	250	1000	4000	8000
Normal				
*G*_7points	0.30	0.78	0.98	0.30
*G*_2points	0.28	0.74	0.95	0.28
Impaired				
*G*_7points	0.044	0.089	0.009	0.002
*G*_2points	0.038	0.071	0.009	0.002

A

	Frequency (Hz)			
	250	1000	4000	8000
Normal				
*A*_7points	15.5	12.3	28.6	34.9
*A*_2points	13.6	11.9	29.9	32.7
Impaired				
*A*_7points	21.5	36.9	55.5	108.1
*A*_2points	20.6	35.6	58.6	104.0

The mean error is 8.5 % across all listeners. Impaired listeners show a lower error (6.8%) than the normal listeners (10.1%). Although an error of 10% can be a reason for concern in other contexts, it is not considered to be problematic in this particular situation. On inspection of the specific values it is concluded that the G/A estimates do not change substantially by only using the thresholds for the long and the short tone and it is not expected to affect the interpretation of the estimates adversely. In other words, the observations are still situated in the same quadrant.

It is concluded that the thresholds for a long and a short tone are sufficient to obtain reliable *G* and *A* parameters. The threshold/duration effect can be quantified by measuring only two thresholds in stead of the seven thresholds if the reciprocal function is used to fit the data.

6.5 Conclusion

Thresholds as a function of duration were measured in normal and impaired listeners and confirmed the commonly accepted effect that thresholds decrease as the tone duration is increased. Most threshold/duration functions found in the impaired listeners are shallower than the normal listeners' functions.

To quantify the threshold/duration effect and subsequently differentiate between normal and impaired listeners, several fitting procedures were explored using the threshold/duration data. First it was found that the difference in threshold between a long and a short tone does not provide a reliable measure to differentiate between the normal and the impaired listeners. This is caused by a large variability within each group and overlap between the groups. Secondly, an exponential function was fitted to the data but did not provide a very good fit. The time constant *tau* estimated by fitting this function was not able to differentiate between the two groups adequately. Fitting a broken stick-function to the threshold/duration data also did not provide a satisfactory measure of the effect. The slope showed to be an overestimate of the threshold/duration effect compared to slope values reported in the literature. This overestimation is thought to be caused by omitting the thresholds for the longer tones in the fit. As with the other functions, the slope measure did not differentiate reliably between normal and impaired listeners.

Finally, a two-parameter reciprocal function was used to fit the threshold/duration data. The function fitted the data well and two parameters G and A were estimated. The parameters *G* and *A* were shown to be independent parameters. Either or both of the 2 parameters may be affected in impaired listeners. Particularly *A* may prove to be useful for discriminating between patients.

Four possible *G*/*A* combinations were represented by 4 quadrants shown in Figure 6.4-7. Quadrant I is associated with normal hearing (normal *G*/normal *A*). Quadrant II and III are associated with impaired hearing. *G* is always low but *A* is either normal or raised. No observations were found in Quadrant IV (normal G/high *A*).

It can be hypothesized that observations situated in Quadrant II are associated with a conductive hearing loss. This is supported by the parameter estimates found in listener BMe, whose hearing loss is of a conductive nature and by the ear-plug experiment where a simulated conductive hearing loss resulted in Quadrant II parameters. Quadrant III is assumed to be associated with a sensorineural hearing loss. This assumption is based on the widely reported observations that most sensorineural hearing losses are associated with raised and shallow threshold/duration functions, typical for the functions found in Quadrant III.

Still, only 3 of the 9 data points situated in Quadrant II presented in this study are clearly associated with a conductive hearing loss. The 6 remaining points are associated with listeners whose loss is assumed to be of a sensorineural nature. This suggests that the origin of the sensorineural impairment is not the same as the sensorineural loss of the listeners in Quadrant III and emphasizes the potential diagnostic value of the *A* parameter. It is not clear at the moment which sensorineural pathology would generate Quadrant II-parameters. This topic will be revisited when discussing the hearing profiles of the impaired listeners, presented in chapter 8.

Ultimately, it was shown that reliable parameter estimates can also be generated with only 2 threshold measurements. Fitting the formula to the thresholds for a long tone (0.512 s) and a short tone (0.008 s) yields G and A values which are similar to the values obtained when fitting the function to thresholds for 7 different durations. This finding means a considerate reduction in the testing time (less than 2 minutes) which again makes the measure viable in a clinical setting.

CHAPTER 7

Hearing profiles of normal listeners

7.1 Introduction

The first few chapters of this thesis were aimed at describing the development of the threshold estimation procedures and explaining the assessment measures that will be applied when developing hearing profiles of individuals.

Although the target group of this exercise is the impaired listener, it is important that the entire protocol is tested on a number of normal listeners first. Acquiring hearing profiles of normal listeners is aimed at evaluating the suitability of the threshold procedure in terms of speed and reliability. Secondly, the profiles of the normal listeners allow us to test the psychoacoustic measures and evaluate the results against those reported in the literature. Finally, it is important to create a standard profile on which the comparison of impaired data can be based

As described in the previous chapters the hearing profile will provide information on the sensitivity, the frequency selectivity and the compression of the listener. It will be shown that the hearing profiles provide a detailed assessment of an individual's hearing and is able to highlight sub-clinical features undiagnosed in clinical screenings.

This report will proceed as follows:

1. Summarize the methodology used to measure the hearing profiles

2. Present the hearing profiles of 3 normal listeners
3. Evaluate the feasibility of developing hearing profiles

7.2 Method

A detailed description and motivation of the testing protocol has been presented in Chapters 5 and 6. A short description of the testing procedure will be repeated below.

The data sets/profiles discussed in this chapter were collected over several testing sessions. The initial session was aimed at screening and familiarizing the participant with the testing environment and tasks. Consecutive sessions were usually 90 minutes long with 2 to 3 5-minute brakes. When possible each session focused on one specific paradigm or task.

7.2.1 Tests

At the intake, a basic clinical assessment was performed to confirm normal hearing in the participants. The tests included in this assessment were otoscopy, pure-tone audiometry, tympanometry, DP OAE measurements and Rinne tuning fork test.

The psychoacoustic tests performed were Absolute Thresholds measurements for 500 ms and 8-ms tones, Threshold/Duration functions, Temporal Masking Curves (TMC) and Iso Forward Masking curves (IFMC). The mechanisms behind the paradigms and their value for the hearing profile were discussed in Chapter 5 and 6. The specific parameters used in each test are discussed below. All listeners were tested according to this set of tests and conditions unless stated otherwise.

Absolute threshold

Detection thresholds were measured for 500-ms and 8-ms pure tones (raised cosine onset and offset times of 0.004 s) at frequencies 250, 500, 1000, 2000, 4000 and 8000 Hz. Thresholds are usually based on a single measurement.

Threshold as a function of tone duration

Detection thresholds were measured for pure tones for seven different durations (0.008, 0.016, 0.032. 0.064, 0.128, 0.256 and 0.512 s) at frequencies 250, 1000, 4000, and 8000 Hz. Ramps were raised cosine onset and offset times of 0.004 s. Each threshold is the mean of three measurements.

The predictive reciprocal function described in Chapter 6.4 was fitted to the threshold/duration functions and estimates for parameters *A and G* were calculated for each test frequency.

Iso Forward Masking Curves (IFMC)

Masker thresholds were measured for probe frequencies (f_p) 250, 500, 1000, 2000, 4000, 6000 and 8000 Hz. The masker frequencies (f_m) are 0.5, 0.7, 0.9, 1, 1.1, 1.3, and 1.6 * f_p. For all conditions, the probe level was set at 10 dB SPL above the 8-ms threshold of the probe. Masker and probe durations were 0.108 s and 0.008 s respectively. In the 250 Hz condition, masker and probe duration were 0.108 s and 0.016 s respectively. The probe duration was increased to 0.016 s at 250 Hz to allow at least 4 cycles in the stimulus. The stimuli were ramped with raised cosine onset and offset times of 0.004 s. The gap duration was 0.01 s and was defined as the duration of the silence period between the masker offset and the probe onset. Thresholds were generally the mean of three threshold measurements except for conditions where clipping occurs and no threshold could be obtained.

Temporal Masking Curves (TMC)

Masker thresholds were measured for nine different gap durations (0.01, 0.02, 0.03, 0.04, 0.05, 0.06, 0.07, 0.08 and 0.09 s) for probe frequencies (f_p) 250, 500, 1000, 2000, 4000 and 6000 Hz. The gap duration was defined as the duration of the silence period between the masker offset and the probe onset. In the on-frequency condition, the masker frequency was set equal to the probe frequency. For all conditions, the probe level was fixed at 10 dB SPL above the 8-ms threshold of the probe. The linear reference (dotted line in charts) was found by fitting a straight line to the off-frequency data for a probe frequency of 6000 Hz and masker frequency (f_m) = 2400 Hz (0.4 * f_p). The rational behind this choice of linear reference has been discussed in Chapter 5.5. Masker and probe durations were 0.108 s and 0.008 s respectively. In the 250 Hz condition, masker and probe durations were 0.108 s and 0.016 s respectively. The probe duration was increased to 0.016 s at 250 Hz to allow at least 4 cycles in the stimulus. The stimuli were ramped with raised cosine onset and offset times of 0.004 s. Thresholds were the mean of three measurements except for conditions where no threshold could be obtained.

7.2.2 Threshold estimation procedure

The data presented in this chapter were collected over a period of 2 years. In this time our testing procedure was not yet finalized and therefore minor changes in the procedure occurred during this period. For most listeners one or two modifications were implemented during the data-collection of their profile. As discussed in Chapter 2, these changes are mainly related to the method of selecting the level of the successive stimuli in a threshold run and to the method used to fit the psychometric function. The improvements do not affect the threshold estimates but rather result in faster or more reliable threshold estimates.

The different procedures used were ‘Maximumlikelihood’, ‘Single Interval 50 %’ and ‘Single-Interval-Up-Down’. A detailed description of these procedures can be found in Chapter 2.3. The default number of Threshold trials used in all procedures is 10 trials (not including catch trials and the trials in the initial phase) unless stated otherwise. Table 7.2-1 provides an overview of the procedures used for each test and listener.

Table 7.2-1. Overview of procedures used for each test for 3 normal listeners.

	CMa	MKi	SRa
Absolute threshold	SingleInterval 50%	Max.likelihood	Max.likelihood (5 trials)
Threshold/duration	SingleInterval 50%	Max.likelihood	Max.likelihood
IFMCs	SingleInterval 50%	SingleInterval 50%	SingleInterval 50%
TMCs	SingleInterval 50%	SingleInterval 50%	Single-Interval-Up-Down

7.2.3 Listeners

Three normal listeners were tested. All listeners were students aged between 22 and 31 years. None of the listeners had a history of hearing impairment nor have any prior experience with psychoacoustic experiments. In all cases a full profile is obtained for the left ear, some tests are repeated in the right ear for comparison purposes. The tests are administered at different points in time over a one-year period. The listeners received payment for their participation in the project.

7.2.4 Apparatus

Pure Tone audiometry and tympanometry were performed using a Kamplex AA222 combination audiometer/impedance test station (PC Werth, London, UK). Distortion product OAEs were measured using a dual channel ILO292 Echoport USB-II (Otodynamics Ltd, Hatfield UK).

For all psychoacoustic tests, stimuli were generated using Matlab at a sampling rate of 96000 Hz, with 24-bit resolution. The stimuli are played monaurally via circumaural Sennheiser HD600 headphones to the participant seated in a double-walled sound attenuating booth.

7.3 Results

In the next section, hearing profiles of the three normal listeners will be presented separately.

7.3.1 Listener CMa

Participant CMa is a 22-year old male with normal hearing. There is no medical history of hearing impairment.

Audiological assessment

Otoscopy: Normal

Tympanometry: Normal in both ears

Rinne: Positive in both ears

A. Audiogram **B. DP OAE**

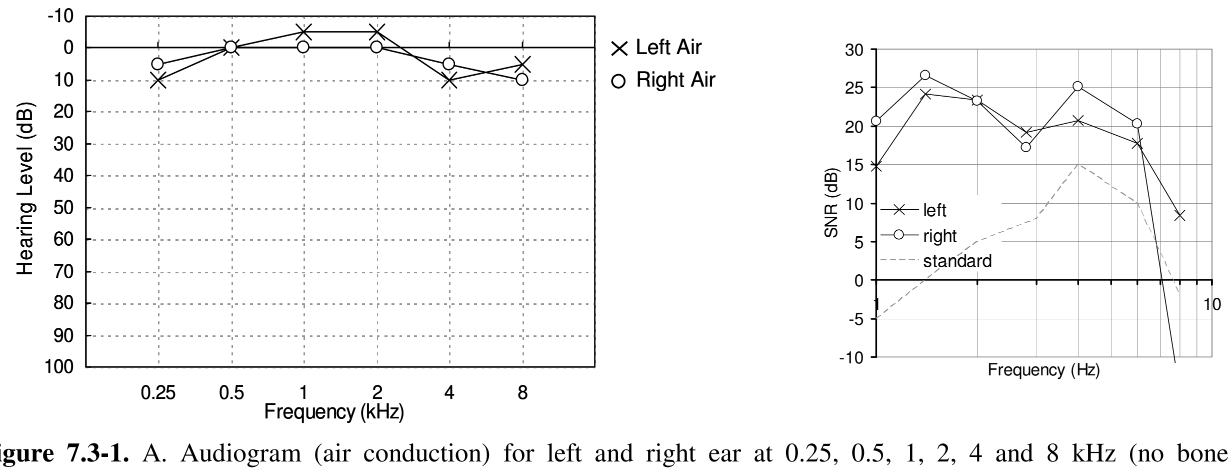


Figure 7.3-1. A. Audiogram (air conduction) for left and right ear at 0.25, 0.5, 1, 2, 4 and 8 kHz (no bone conduction measured). B. DP OAE responses for left and right ear for frequencies 1, 1.4, 2, 2.8, 4, 6 and 8 kHz. Responses are expressed in SNR (dB). The dotted grey line represent the 5th percentile of responses of normal listeners reported by Gorga et al. (1997).

The clinical screening confirms normal hearing. The audiogram is within normal range. There is no indication for an outer/middle ear problem. DP OAE responses are present except at 8 kHz in the right ear.

Psychoacoustic tests

Absolute thresholds: Thresholds were measured for pure tones with tone durations of 500 and 8 ms at frequencies 250, 500, 1000, 2000, 4000 and 8000 Hz both in the left and right ear (Figure 7.3-2A & F). Thresholds are based on the mean of 2 measurements in the 500-ms condition and on a single measurement in the 8-ms condition.

In both ears, all thresholds are well within normal limits. The difference between the thresholds for a 500-ms tone and a 8-ms tone is large at all frequencies (Figure 7.3-2A). The 500-ms and the 8-ms function run parallel. At 4000 Hz there is a small decrease in the difference but there is no additional evidence in the further test results that suggests a problem at this frequency.

Threshold/duration functions were measured in the left ear (Figure 7.3-2B) at frequencies 250, 1000, 2000, 4000 and 8000 Hz. Tone durations were 0.008, 0.016, 0.032, 0.064, 0.128, 0.256 and 0.512 s. Thresholds are the mean of three measurements.

The threshold/duration functions are normal at all frequencies. All functions are steep with an mean slope (based on the difference between thresholds for a 500-ms and 8-ms tone) of 3 dB/octave across all test frequencies. This value is in agreement with the values reported in the literature (see section 6.2).

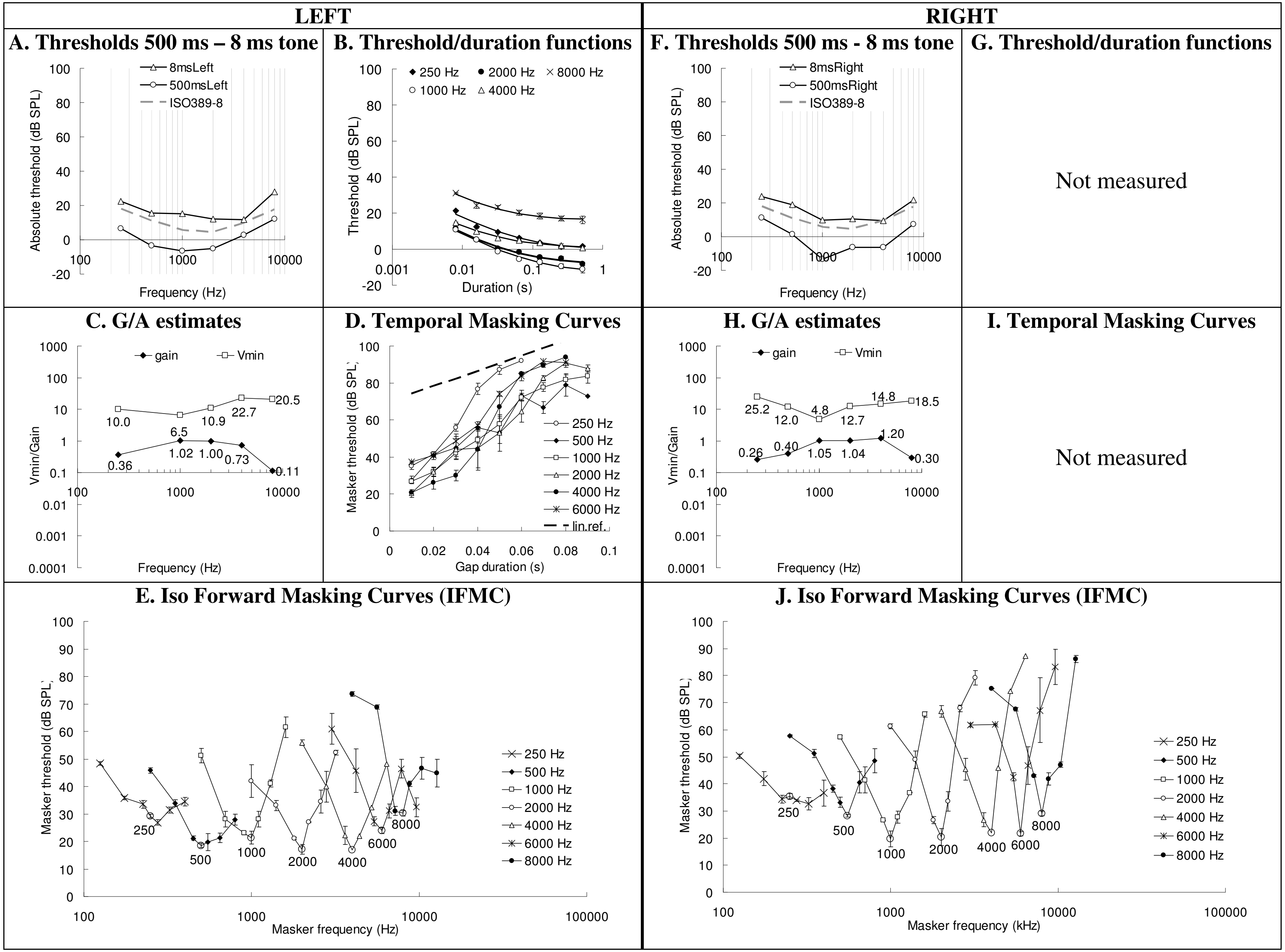


Figure 7.3-2: Normal Listener CMa
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Figure 7.3-2. Psychoacoustic profile of normal listener CMa.

A. Left ear: Absolute thresholds as a function of frequency for 500-ms tones (open circles) and 8-ms tones (open triangles) at frequencies 250, 500, 1000, 2000, 4000 and 8000 Hz. Thresholds are based on a single measurement for the 8-ms thresholds and the mean of 2 measurements for the 500-ms thresholds.

B. Left ear: Thresholds as a function of tone duration at frequencies 250, 1000, 2000, 4000, and 8000 Hz. Thresholds are the mean of 3 measurements. Error bars represent 1 standard error. The full lines represent predictions made using the reciprocal function described in section 6.4. Each fitted function provides an estimate of parameters *A* and *G* presented in Graph C.

C. Left ear: Estimates of parameters *G* (filled diamonds) and *A* (open squares), associated with the predictive functions (full lines in graph B) to the threshold/duration functions reported in graph B.

D. Left ear: Temporal Masking Curves (TMC) representing masker thresholds as a function of gap duration for probe frequencies (f_p) 250, 500, 1000, 2000, 4000 and 6000 Hz. All TMCs represent the condition where the masker frequency $f_m = f_p$ (on-frequency condition). The linear reference (dotted line) represents the off-frequency condition for f_p = 6000 Hz and f_m = 2400 Hz (0.4 * f_p). Gap durations vary from 0.01 s to 0.09 s in steps of 0.01. Thresholds are generally the mean of 3 measurements. Error bars represent 1 standard error.

E. Left ear: Iso Forward Masking Curves (IFMC) for probe frequencies (f_p) 250, 500, 1000, 2000, 4000, 6000 and 8000 Hz. The masker frequencies (f_m) are 0.5, 0.7, 0.9, 1, 1.1, 1.3, and 1.6 * f_p. The masker threshold at probe frequency is represented by the large open circle. The accompanying value indicates the probe frequency. Thresholds are generally the mean of 3 measurements. Error bars represent 1 standard error.

F. Same as A but for right ear.

H. Right ear: Estimates of parameters *G* (filled diamonds) and *A* (open squares) based on the difference in absolute threshold between the 8-ms and 500-ms tones reported in graph F.

J. Same as E. but for right ear.

The *G* and *A* parameters in the left ear (Figure 7.3-2C) were estimated by fitting the reciprocal function to the threshold/duration functions as described in section 6.4. The parameter estimates at all frequencies are well within the normal range determined in section 6.4.

For the right ear, *G* and *A* parameters were estimated by fitting the same reciprocal function (Figure 7.3-2H) . This time however the reciprocal function was fitted to the absolute thresholds of the 500-ms and 8-ms tone presented in Figure 7.3-2F. The estimates are well within normal limits and are very similar to the parameters found for the left ear. This is supportive evidence for our contention that two threshold measurements are a satisfactory basis for estimating *A* and *G*.

Iso Forward Masking Curves: Both in the left and the right ear IFMCs were measured for probe frequencies (f_p) 250, 500, 1000, 2000, 4000, 6000 and 8000 Hz. The masker frequencies (f_m) are 0.5, 0.7, 0.9, 1, 1.1, 1.3, and 1.6 * f_p. For all conditions, the probe level was set at 10 dB SPL above the 8-ms threshold. Thresholds are generally the mean of three measurements.

Normal IFMCs were found at all frequencies in both ears (Figure 7.3-2E & J). All IFMCs are V-shaped and sharp. This implies that maskers with a frequency close to the probe frequency are very potent at masking the probe and as the masker frequency moves away from the probe frequency, the intensity of this masker needs to be considerably increased to just mask the probe. This is a strong indication that CMa's auditory system is competent at filtering out interfering sounds or noises.

However, some peculiarities were seen in the IFMCs of the left ear. At the highest frequencies (6000 and 8000 Hz) the high frequency tails point downwards indicating that the masker with a frequency furthest away from the probe frequency is more potent at masking the probe than a masker with a frequency closer to the probe frequency. It is not clear what the origin of this anomaly could be.

Also at 250 Hz, the frequency of the most effective masker is greater than the probe frequency. This probably reflects the outer middle ear (OME) response which attenuates low frequencies.

Temporal Masking Curves: TMCs were measured in the left ear (Figure 7.3-2D) for probe frequencies 250, 500, 1000, 2000, 4000 and 6000 Hz. All functions represent the on-frequency condition where the masker frequency (f_m) = probe frequency (f_p). The linear reference (dotted line) was found by fitting a straight line to the off-frequency thresholds with a probe frequency of 6000 Hz and a masker frequency (f_m) = 2400 Hz (0.4 * f_p). For all conditions, the probe level was set at 10 dB SPL above the 8-ms threshold. Gap durations varied from 0.01s to 0.09 s in steps of 0.01. Thresholds are generally the mean of 3 measurements.

There is a clear indication of compression at all frequencies. To allow for closer inspection, the TMCs at each frequency are shown separately in Figure 7.3-3. All TMCs show a pattern which agrees with TMCs reported in the literature. In general, the function consists of shallow section followed by a steep section. In most TMCs a further shallow section is seen at high masker levels.

Conclusion

Listener CMa shows normal performance at all the measures presented in this report. A standard clinical screening confirms normal hearing with hearing levels within the normal range, no indication of an outer/middle ear problem and normal DP OAE response. The psychoacoustic tests returned normal results in all measures. Absolute thresholds are within normal limits, threshold/duration functions are steep, TMCs show clear signs of compression and the IFMCs are sharp. The origin of the downward pointing tails in the 6000 and 8000-Hz IFMCs in the left ear are not explained since there are no indications of possible anomalies in any of the other measures nor does there seem to be a methodological explanation for this observation given that they do not occur in the right ear.

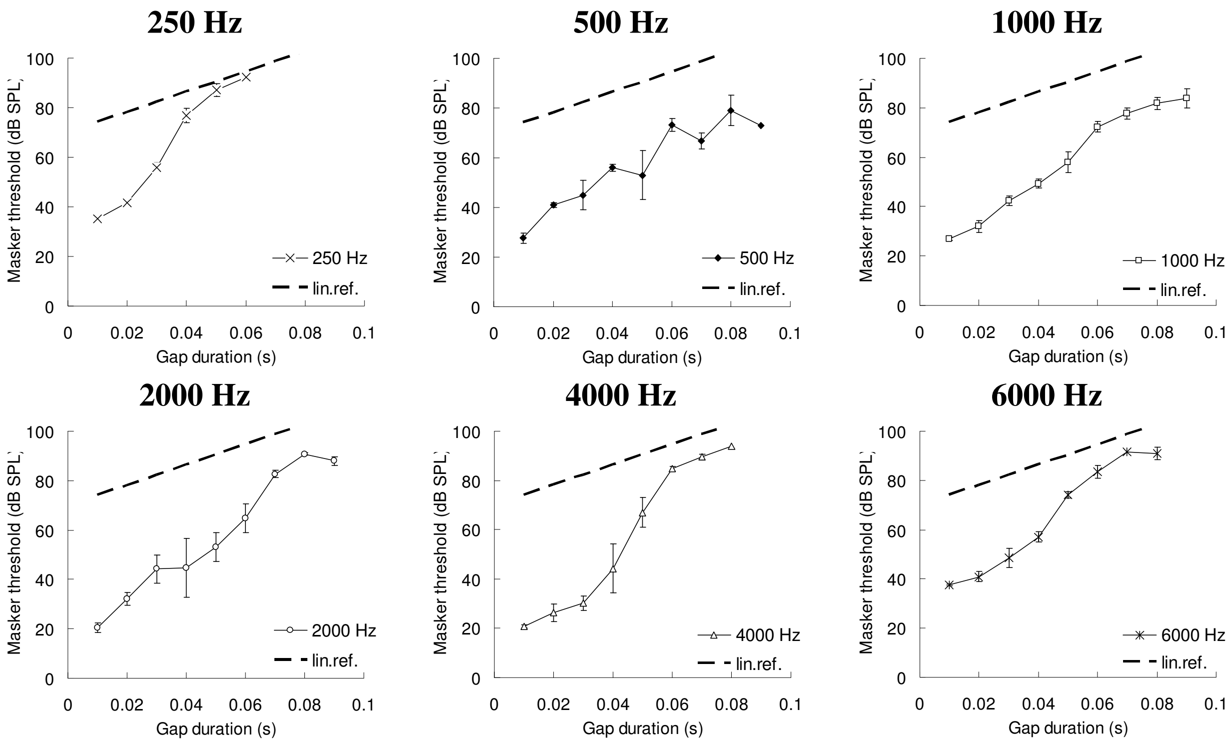

Figure 7.3-3. TMCs for listener CMa for probe frequencies (f_p) 250, 500, 1000, 2000, 4000 and 6000 Hz. Data points represent masker thresholds for gap durations from 0.01 to 0.09 (in steps of 0.1 s). Masker frequency (f_m) is set equal to the probe frequency (f_p) (on-frequency condition). The linear reference (dotted line) was found by fitting a straight line to the off-frequency data for a probe frequency of 6000 Hz and a masker frequency (f_m) = 2400 Hz (0.4 * f_p). The data points are generally the mean of 3 threshold measurements. Error bars represent 1 standard error.

7.3.2 Listener MKi

Participant MKi is a 26-year old female with normal hearing. There is no medical history of hearing impairment.

Audiological assessment

Otoscopy: Normal in both ears

Tympanometry: Normal in both ears

Rinne: Positive in both ears

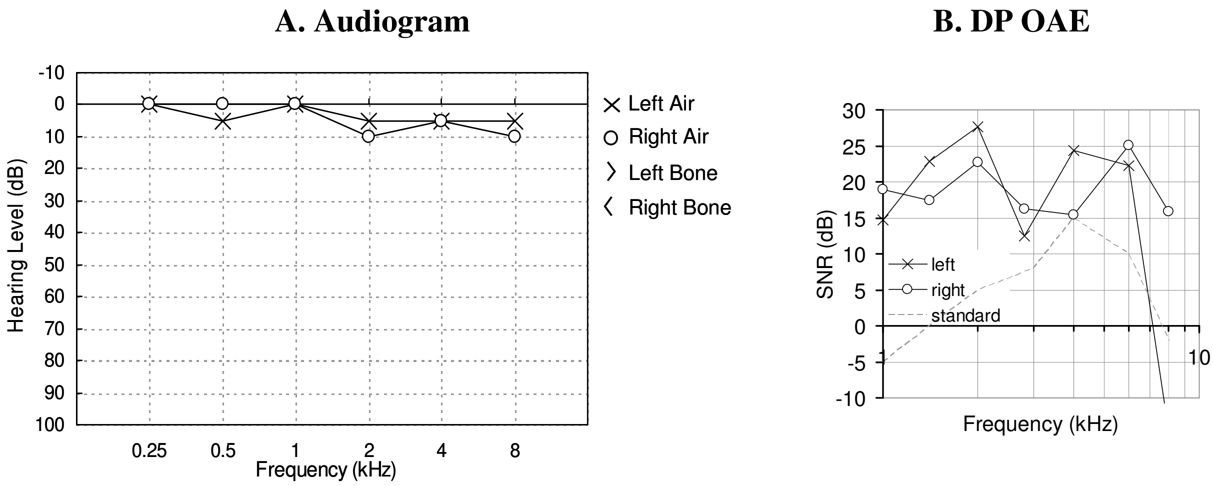

Figure 7.3-4. A. Audiogram (air conduction) for left and right ear at 0.25, 0.5, 1, 2, 4 and 8 kHz (no bone conduction measured). B. DP OAE responses for left and right ear for frequencies 1, 1.4, 2, 2.8, 4, 6 and 8 kHz. Responses are expressed in SNR (dB). The dotted grey line represent the 5^{th} percentile of responses of normal listeners reported by Gorga et al. (1997).

The results of the pure-tone audiometry show normal hearing sensitivity in both ears. Otoscopy and tympanometry are normal. The Rinne tuning fork test is positive in both ears. Distortion product OAEs (DPOAEs) were present in both ears except at 8 kHz in the left ear. In the left ear a sharp dip in response can be seen at 2.8 kHz.

Psychoacoustic tests

Absolute thresholds: In the left ear (Figure 7.3-5A), thresholds were measured for pure tones at frequencies 250, 500, 1000, 2000, 4000 and 8000 Hz for tone durations of 500 ms (open circles) and at frequencies 250, 500, 1000, 1400, 2000, 2600, 4000 and 8000 Hz for tone durations of 8 ms (open triangles). The extra thresholds at frequencies 1400 and 2600 Hz were measured to explore the region on both sides of the 2000 Hz (see below). For the right ear (Figure 7.3-5F), thresholds were measured for pure tones at frequencies 250, 500, 1000, 1500, 2000, 2500, 3000, 3500, 4000, 6000 and 8000 Hz for tone durations of 500 ms (open circles) and 8 ms (open triangles). Thresholds are based on a single measurement.

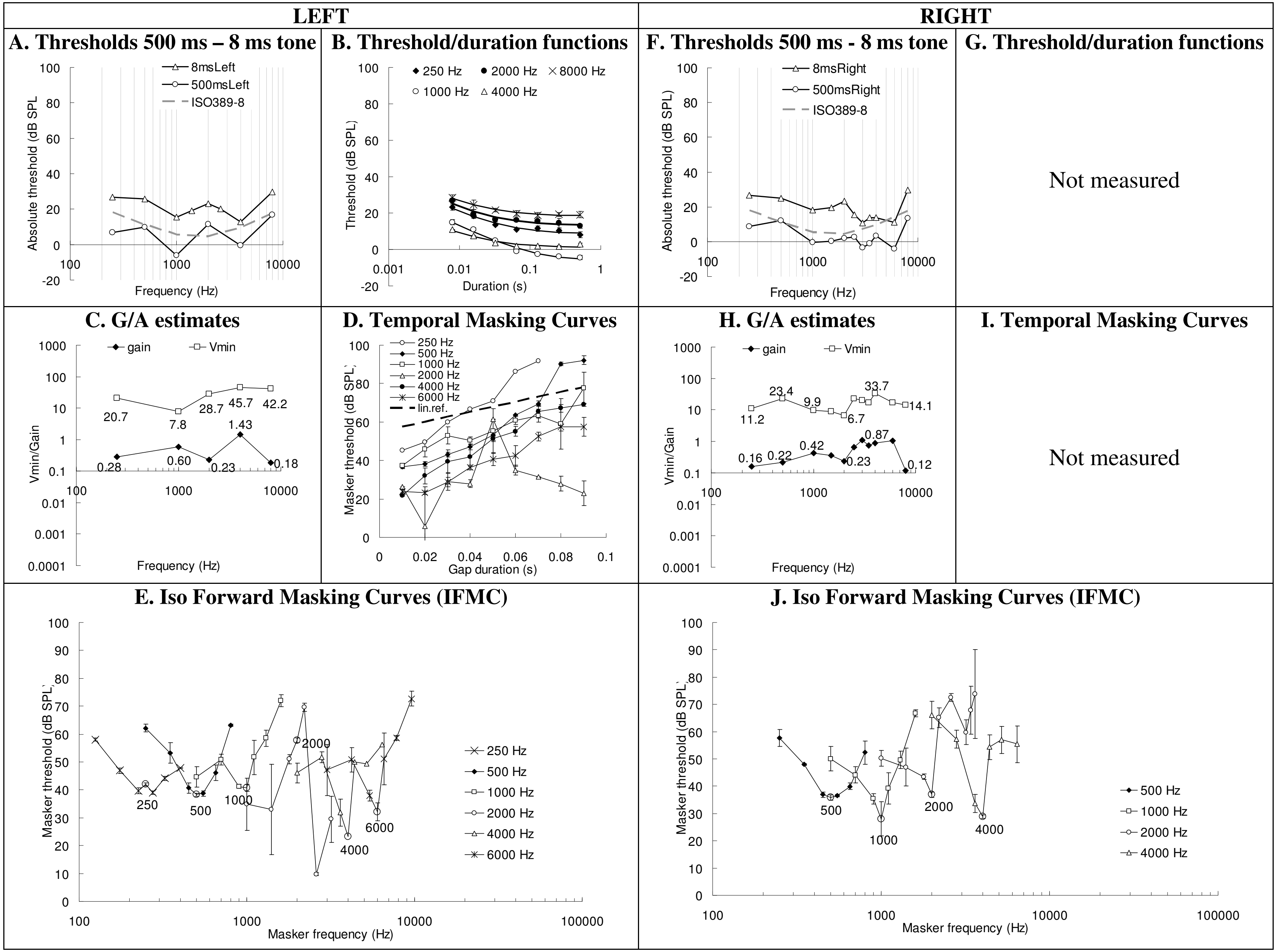

Figure 7.3-5: Normal Listener MKi

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Figure 7.3-5. Psychoacoustic profile of normal listener MKi.

A. Left ear: Absolute thresholds as a function of frequency for 500-ms tones (open circles) and 8-ms tones (open triangles). Thresholds are based on a single measurement.

B. Left ear: Thresholds as a function of tone duration at frequencies 250, 1000, 2000, 4000, and 8000 Hz. Thresholds are the mean of 3 measurements. Error bars represent 1 standard error. The full lines represent predictions made using the reciprocal function described in section 6.4. Each fitted function provides an estimate of parameter *A* and *G* presented in Graph C.

C. Left ear: Estimates of parameters *G* (filled diamonds) and *A* (open squares) at frequencies 250, 1000, 2000, 4000 and 8000 Hz, associated with the predictive functions (full lines in graph B) to the threshold/duration data reported in graph B.

D. Left ear: Temporal Masking Curves (TMC) representing masker thresholds as a function of gap duration for probe frequencies (f_p) 250, 500, 1000, 2000, 4000 and 6000 Hz. All TMCs represent the condition where the masker frequency $f_m = f_p$ (on-frequency condition). The linear reference (dotted line) represents the off-frequency condition for f_p = 6000 Hz and f_m = 2400 Hz (0.4 * f_p). Gap durations vary from 0.01 s to 0.09 s in steps of 0.01. Thresholds are generally the mean of 3 threshold measurements. Error bars represent 1 standard error.

E. Left ear: Iso Forward Masking Curves (IFMC) for probe frequencies (f_p) 250, 500, 1000, 2000, 4000 and 6000 Hz. The masker frequencies (f_m) are 0.5, 0.7, 0.9, 1, 1.1, 1.3, and 1.6 * f_p. The masker threshold at probe frequency is represented by the large open circle. The accompanying value indicates the probe frequency. Thresholds are generally the mean of 3 measurements. Error bars represent 1 standard error.

F. Same as A but for the right ear.

H. Right ear: Estimates of parameters *G* (filled diamonds) and *A* (open squares) based only on the difference in absolute threshold between the 8-ms and 500-ms tones reported in graph F.

J. Same as E. but for right ear.

In both ears, all thresholds are well within normal limits but in the left ear the threshold at 2000 Hz is distinctly raised compared to the thresholds at the surrounding frequencies.

The 500 ms-function runs parallel with the 8 ms-function in both ears. Peaks and dips appear at the same frequencies in both functions. The most prominent feature is a peak at 2000 Hz in the left ear, both in the long and the short tone function. In the right ear a corresponding peak may be present in the short tone function but is less distinct in the long tone function.

In the left ear, the difference between the thresholds for a 500-ms tone and a 8-ms tone is large at all frequencies, but slightly reduced from 2000 Hz onwards. In the right ear, this reduction seems to appear from 2500 Hz onwards. This reduced threshold difference could possibly point to a abnormality at these frequencies.

Threshold/Duration functions: were measured in the left ear (Figure 7.3-5B) at frequencies 250, 1000, 2000, 4000, and 8000 Hz. Tone durations were 0.008, 0.016, 0.032, 0.064, 0.128, 0.256 and 0.512 s. Thresholds are the mean of three measurements.

The threshold/duration functions are normal at all frequencies. All functions are steep with an mean slope (based on the difference between thresholds for a 500 and 8-ms tone) of 2.2 dB per octave across all test frequencies. This value is in agreement with the values reported in the literature (see section 6.2).

The *G* and *A* parameters in the left ear (Figure 7.3-5C) were estimated by fitting the reciprocal function to the threshold/duration functions as described in section 6.4. All parameter estimates are generally within the normal range at all test frequencies as established in Chapter 6.4. At 4000 and 8000 Hz the A estimates are high but still within the normal range. However when we estimate *A* values for these frequencies by fitting the reciprocal function based only on the 8-ms and the 500-ms thresholds presented in Figure 7.3-5A, normal *A* values are found. The comparison of these 2 sets of values will be presented further in this chapter (Figure 7.3-13).

For the right ear, *G* and *A* parameters were estimated by fitting the reciprocal function to the absolute thresholds of the 500-ms and 8-ms tone presented in Figure 7.3-2F. The estimates are well within normal limits and are similar to the parameters found for the left ear.

Iso Forward Masking Curves: In the left ear, IFMCs were measured for probe frequencies 250, 500, 1000, 2000, 4000 and 6000 Hz (Figure 7.3-5E). The masker frequencies (f_m) are 0.5, 0.7, 0.9, 1, 1.1, 1.3, and 1.6 * f_p. For all conditions, the probe level was set at 10 dB SPL above the 8-ms threshold. The data points are generally the mean of three threshold measurements.

In the left ear, normal IFMCs were found at all frequencies except at 2000 Hz. The IFMCs at 250, 500, 1000, 4000 and 6000 Hz have a normal V-shape. The 2000-Hz IFMC however demonstrates a completely different pattern. Instead of the normal V-shape, a W-shaped IFMC is found, meaning that the masker thresholds are highest for masker frequencies close to the probe frequency and lowest for the most remote masker frequencies. In other words, a 2000-Hz probe is most successfully masked by a masker with a frequency substantially higher or lower than the probe frequency. In this specific case, maskers with the frequencies 1400 Hz and 2600 Hz are most effective at masking the 2000-Hz probe. This suggests that the 2000-Hz probe is being registered at BM sites away from the BM site with a BF of 2000 Hz. Additional measures discussed below will explore this hypothesis in more detail.

A smaller peculiarity can be seen in the IFMCs at 1000, 4000 and 6000 Hz. In these three IFMCs the low-frequency tails of the IFMCs bend downwards at the most remote masker frequency. Currently there seems to be no obvious explanation for this observation.

In the right ear, IFMCs were measured for frequencies 500, 1000, 2000 and 4000 Hz (Figure 7.3-5J). For the 2000-Hz IFMC some extra masker thresholds were measured for masker frequencies above 3000 Hz (1.7 and 1.8 * f_p). In general, no distinct abnormalities could be seen in the IFMCs. The 2000-Hz IFMC has a normal V-shape but is slightly raised. The extended high-frequency tail does show a drop at masker frequency 3200 Hz, adding an extra V-shape to the IFMC. This could point to an anomaly since in a normal situation one would expect to see a continuous rise in the high-frequency tail.

Additional IFMC measurements: In the left ear, IFMCs with probe frequency 1400 and 2600 Hz were measured (see Figure 7.3-6). The downward pointing skirts of the 2000-Hz IFMC coincide nicely with the high-frequency slope of the 1400-Hz IFMC and with the low-frequency slope of the 2600-Hz IFMC. This implies that a 2000-Hz probe is actually detected at a BM site with a BF of 1400 Hz or 2600 Hz, since maskers at these frequencies are most potent at masking the 2000-Hz probe. We can conclude from this data that there is an unresponsive or less responsive BM-region around 2000 Hz ranging from 1400 Hz and 2600 Hz.

Temporal Masking Curves: TMCs were measured in the left ear (Figure 7.3-5D) for probe frequencies 250, 500, 1000, 2000, 4000 and 6000 Hz. The masker frequency (f_m) is set equal to the probe frequency (on-frequency condition). The linear reference (dotted line) was found by fitting a straight line to the off-frequency thresholds for a probe frequency of 6000 Hz and a masker frequency (f_m) = 2400 Hz (0.4 * f_p). For all conditions, the probe level was set at 10 dB SPL above the 8-ms threshold. Gap durations varied from 0.01s to 0.09 s in steps of 0.01. Thresholds are generally the mean of three measurements.

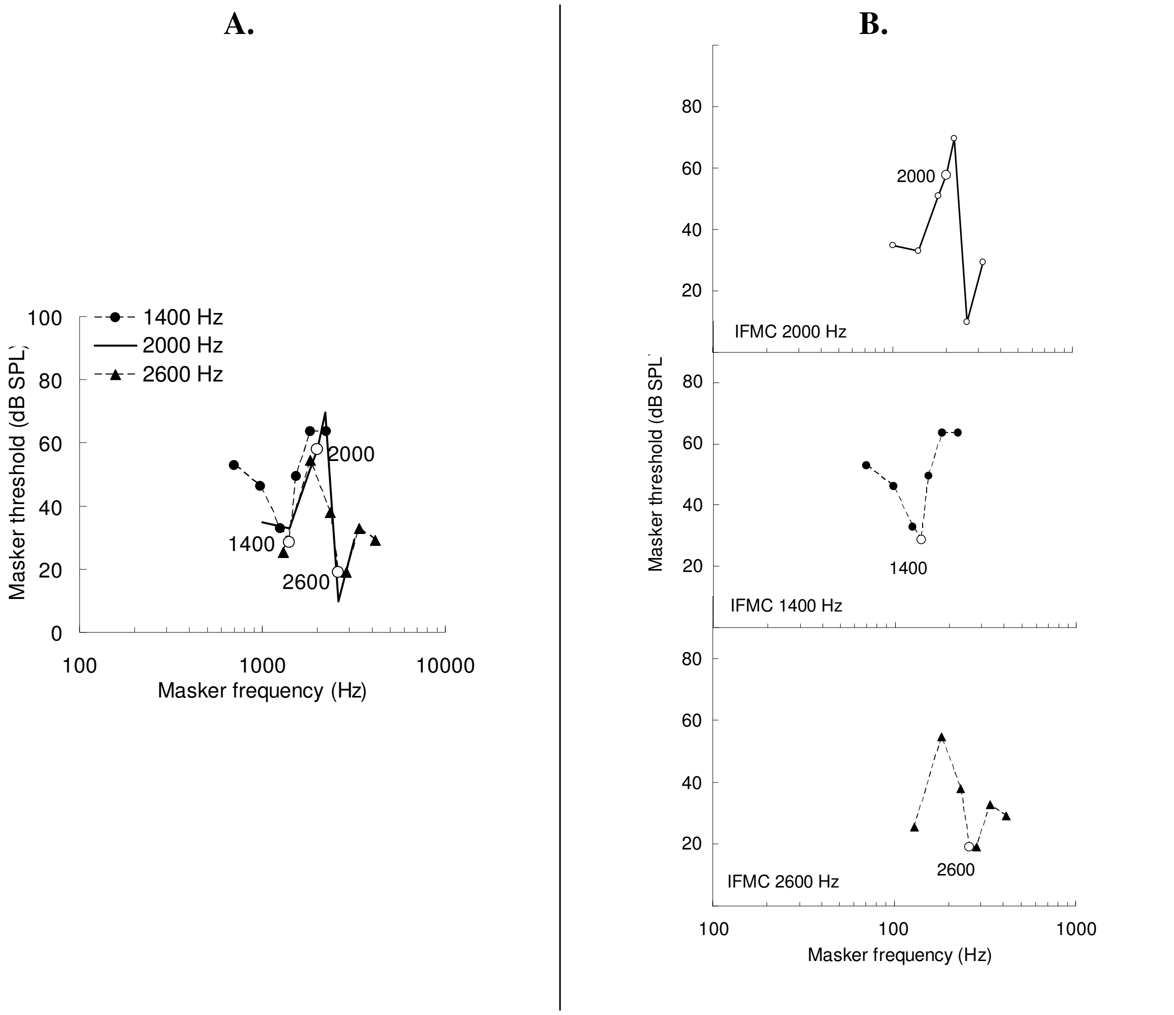

Figure 7.3-6. A. Left ear: IFMCs with probe frequency 2000 Hz (full line) and 1400 and 2600 Hz (dotted lines). The large open circle, represents the masker threshold for a masker frequency at probe frequency. The accompanying data labels represent the probe frequency. B. Same as A but IFMCs separated per frequency.

There are indications of some compression at all frequencies except at 2000 Hz. To allow for closer inspection, the TMCs at each frequency are shown separately in Figure 7.3-7. Most TMCs consist of a steep section with a slope considerably steeper than the linear reference. However, at 2000 Hz the TMC shows an erratic pattern. At this frequency the threshold estimates are variable and it was not possible to get three threshold measurements for all conditions. This may indicate OHC damage at the 2000 Hz site of the basilar membrane (BM).

The TMC at 1000 Hz also shows considerable variability but there is still indication of compression. In general, all frequencies where an increased variability was found were associated with the participant reporting difficulties in performing the task.

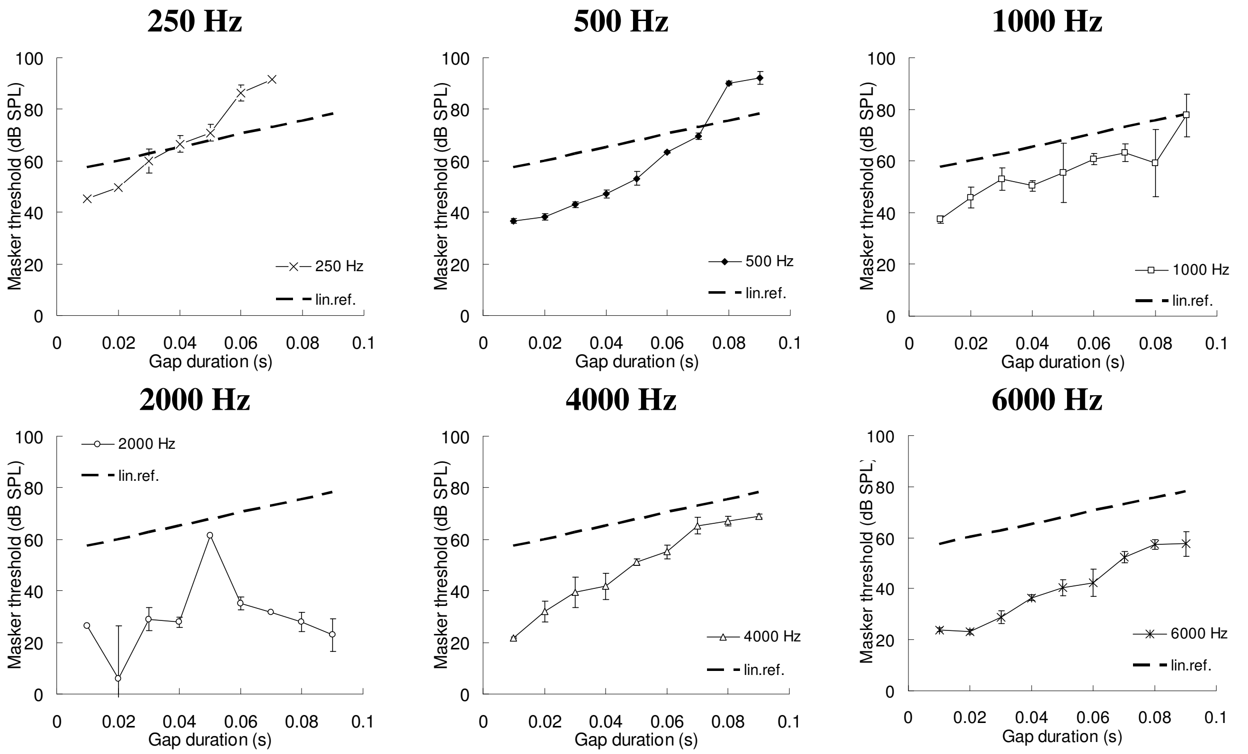

Figure 7.3-7. TMCs for listener MKi. Temporal Masking Curves (TMC) for probe frequencies 250, 500, 1000, 2000, 4000 and 6000 Hz. Data points represent masker thresholds for gap durations from 0.01 to 0.09 (in steps of 0.1 s). Masker frequency (f_m) is set equal to the probe frequency (f_p) (on-frequency condition). The linear reference (dotted line) was found by fitting a straight line to the off-frequency data for a probe frequency of 6000 Hz and a masker frequency (f_m) = 2400 Hz (0.4*f_p). Thresholds are generally the mean of 3 measurements. Error bars represent 1 standard error.

Interaural pitch matching experiment: The different IFMCs at 2000 Hz in the left and right ear raised the question that diplacusis might be present in this frequency area. *Diplacusis* is the phenomenon where the same tone presented alternately to the two ears is perceived as having different pitches in the two ears (Moore, 2007). The presence of diplacusis has been associated with cochlear damage.

Experiment 1: In a first test MKi was asked to compare the pitch of a pure tone in the ‘problematic’ left and the ‘normal’ right ear. The tone was presented alternately to the left and the right ear. MKi was asked to rate the pitch in the left ear compared to the right ear (higher, lower or same as in the right ear). Tones at a range of widely-spaced frequencies (1000, 1500, 2000, 2500, 3000 and 4000 Hz) were presented.

For tones from 2000 Hz to 3000 Hz, MKi rated the pitch in the ‘problem’ ear as having a higher pitch than the same tone in the normal ear. Below 2000 Hz the pitch was rated the same in both ears. The pitch of a 4000 Hz-tone was rated lower in the ‘problematic’ ear than in the normal ear.

It is concluded that tones at 2000, 2500 and 3000 Hz in the ‘problematic’ left ear are potentially picked up by a filter with a higher BF, hence the perception of a higher pitch in the defective ear.

Experiment 2: In a second experiment, the ‘normal’ right ear was always presented with a 2000 Hz-tone. The tones in the ‘problematic’ left ear were presented at different frequencies (1800, 1900, 1940, 1950, 1960, 1980, 1995, 2000 Hz). MKi was asked to rate the pitch in the left ear compared to the pitch of the 2000 Hz-tone in the right ear.

Tones presented in the left ear with frequencies *from 1800 Hz to 1950 Hz* were correctly perceived as having a lower pitch than the 2000 Hz tone in the normal ear. When both tones have a frequency of *2000 Hz* MKi replicates the response given in the first experiment and rates the pitch in the normal ear as lower than the pitch in the defective ear. However, *between 1960 Hz and 1995 Hz*, tones in the defective ear are rated as having the same pitch as a 2000 Hz tone in the normal ear. This is unexpected since normal listeners are extremely sensitive in perceiving frequency differences between 2 tones, so one would expect a normal listener to rate tones of these frequencies as having a different pitch[3].

A possible explanation of this result is that tones between 1960 and 1995 Hz are picked up in the defective ear by the same filter to the 2000-Hz filter in the normal ear. However this does not agree with a previous hypothesis (see Experiment 1) that a 2000-Hz tone in the defective, left ear

[3] At 2000 Hz the difference limen for frequency (= the smallest difference in frequency at which people can still distinguish 2 tones as being different) is less than 5 Hz (Moore, 2004).

is picked up by filter with higher BF. If there is a 2000-Hz filter in the left ear, responding to the 1960-1995 Hz tones, then surely this filter should also respond to a 2000 Hz tone in that ear. It is currently not clear what the origin of this effect is.

Conclusion

MKi has proven to be a very interesting case. A standard clinical screening confirms normal hearing with hearing levels within the normal range and no indication of an outer/middle ear problem. DP OAE responses are present. The psychoacoustic tests generally show normal results in agreement with similar experiments presented in the literature.

Although MKi has no complaints about her hearing and the clinical screening shows no signs of a problem at any frequency, an abnormality was found at the 2000 Hz-site in the left ear. It is concluded that the origin of the 2000-Hz problem could be an island of non-functioning outer hair cells at the 2000-Hz site of the basilar membrane with boundaries at 1400 and 2600 Hz.
The findings supporting this conclusion are:

- The 2000 Hz-IFMC is W-shaped instead of the normal V-shape. The lowest masker thresholds were found at 1400 and 2600 Hz.
- No reliable TMC can be obtained at 2000 Hz.
- The absolute thresholds for long and short tones show a distinct peak at 2000 Hz.

Not all measures show anomalies at this frequency. The threshold/duration function and subsequent *G* and *A* estimates did not show any abnormalities. In hindsight, when taking a closer look at the DP OAE responses in the left ear, there is a strong DP OAE response at 2 kHz but a very distinct dip at 2.8 kHz. However, it is not clear if the 2.8 kHz dip correlates with the results found in the psychoacoustic measures.

The anomalies in the left ear described above are not found in the limited data collected on the right ear. A minor peak is seen at 2000 Hz in the absolute threshold for a short tone but the *G* and *A* estimates are normal. The IFMCs are normal at all frequencies. At 2000 Hz the IFMC is V-shaped but appears to be slightly raised. A dip is found in the high-frequency slope of this IFMC but no further problems were identified. We can conclude that the anomaly found in the left ear is not present in the right ear although small abnormalities observed in the right ear could indicate that a similar but less severe problem is present in the right ear.

In conclusion, the profile of MKi unexpectedly proved to be very interesting. The fact that a sub-clinical problem is diagnosed in a number of psychoacoustic measures encourages us that the hearing profile is indeed a useful tool in constructing a detailed picture of a person's hearing. Furthermore the findings specific to MKi's profile do give rise to a number of interesting questions such as 'Does this 2000 Hz unresponsive region predict future problems with hearing?' and 'Is it only a symptom of a larger undiagnosed problem in MKi's hearing?', etc…

7.3.3 Listener SRa

Participant SRa is a 31-year old female with normal hearing. There is no medical history of hearing impairment.

Audiological assessment:

Otoscopy: Normal in both ears

Tympanometry: Normal in both ears

Rinne: Positive in both ears

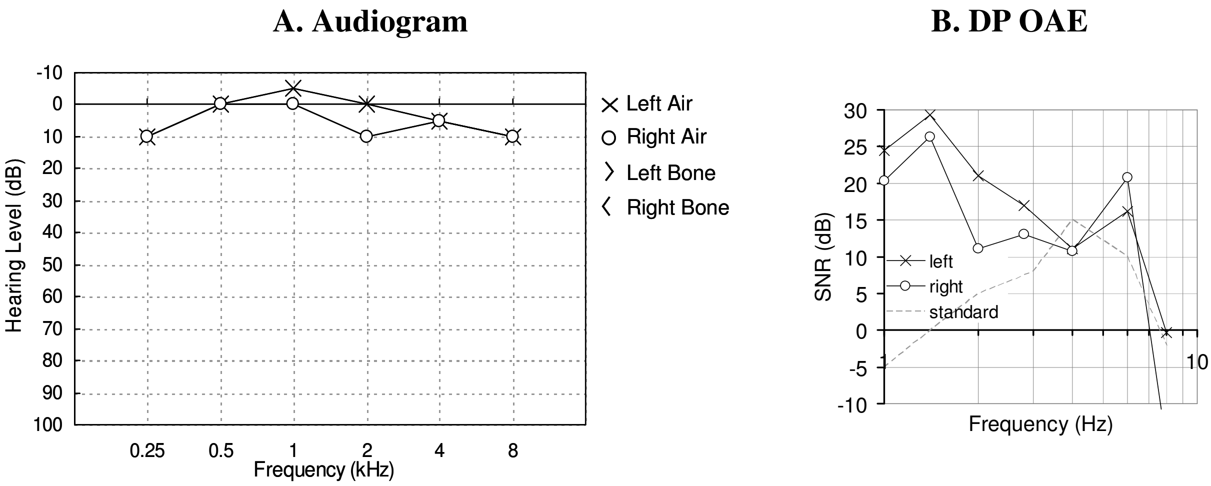

Figure 7.3-8. A. Audiogram (air conduction) for left and right ear at 0.25, 0.5, 1, 2, 4 and 8 kHz (no bone conduction measured). B. DP OAE responses for left and right ear for frequencies 1, 1.4, 2, 2.8, 4, 6 and 8 kHz. Responses are expressed in SNR (dB). The dotted grey line represent the 5th percentile of responses of normal listeners reported by Gorga et al. (1997).

The results of the pure tone audiometry show normal hearing sensitivity in both ears. Otoscopy and tympanometry are normal. The Rinne tuning fork test is positive in both ears. Distortion product OAEs (DPOAEs) are present in both ears except at 8 kHz in the right ear. DPOAE responses at 4 kHz are present in both ears but slightly weaker than the responses associated with the 5th percentile of normal listeners suggested by Gorga et al. (1997).

Psychoacoustic tests:

Absolute thresholds: Thresholds were measured for pure tones with tone durations of 500 and 8 ms at frequencies 250, 500, 1000, 2000, 4000 and 8000 Hz both in the left and right ear (Figure 7.3-10A & F). Thresholds are based on a single measurement in the 500-ms condition and on the mean of 2 measurements in the 8-ms condition.

Absolute thresholds are well within normal limits in both ears, but some small anomalies can be observed at 2000 Hz and 8000 Hz. Overall, the 500-ms function runs parallel with the 8-ms function. At 2000 Hz a small peak is observed in the long as well as the short tone–function in both left and right ear. This peak is a potential indicator of a sub-clinical problem at 2000 Hz.

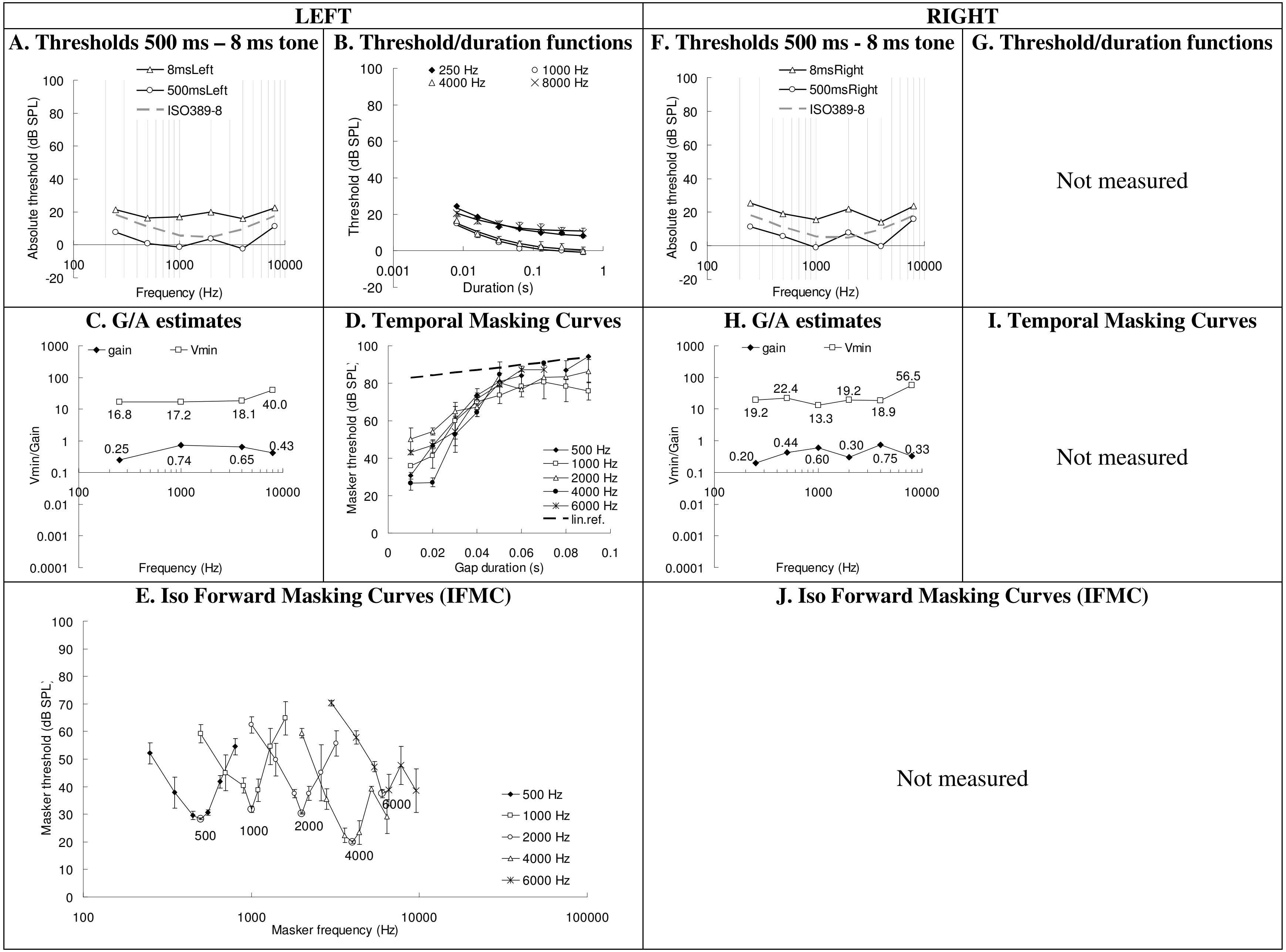

Figure 7.3-9: Normal Listener SRa

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Figure 7.3-9. Psychoacoustic profile of normal listener SRa.

A. Left ear: Absolute thresholds as a function of frequency for 500-ms tones (open circles) and 8-ms tones (open triangles). Thresholds are the result of a single measurement for the 500-ms thresholds and the mean of 2 measurements for the 8-ms thresholds.

B. Left ear: Thresholds as a function of tone duration at frequencies 250, 1000, 4000, and 8000 Hz. Thresholds are the mean of 3 measurements. Error bars represent 1 standard error. The full lines represent predictions made using the reciprocal function described in section 6.4. Each fitted function provides an estimate of parameter *A* and *G* presented in Graph C.

C. Left ear: Estimates of parameters *G* (filled diamonds) and *A* (open squares) at frequencies 250, 1000, 4000 and 8000 Hz, associated with the predictive functions (full lines in graph B) to the threshold/duration data reported in graph B.

D. Left ear: Temporal Masking Curves (TMC) representing masker thresholds as a function of gap duration for probe frequencies (f_p) 250, 500, 1000, 2000, 4000 and 6000 Hz. All TMCs represent the condition where the masker frequency $f_m = f_p$ (on-frequency condition). The linear reference (dotted line) represents the off-frequency condition for f_p = 6000 Hz and f_m = 2400 Hz (0.4*f_p). Gap durations vary from 0.01 s to 0.09 s in steps of 0.01. Thresholds are generally the generally the mean of 3 threshold measurements. Error bars represent 1 standard error.

E. Left ear: Left ear: Iso Forward Masking Curves (IFMC) for probe frequencies (f_p) 250, 500, 1000, 2000, 4000 and 6000 Hz. The masker frequencies (f_m) are 0.5, 0.7, 0.9, 1, 1.1, 1.3, and 1.6 * f_p. The masker threshold at probe frequency is represented by the large open circle. The accompanying value indicates the probe frequency. Thresholds are generally the mean of 3 measurements. Error bars represent 1 standard error.

F. Same as A but for right ear.

H. Right ear: Estimates of parameters *G* (filled diamonds) and *A* (open squares) based only on the difference in absolute threshold between the 8-ms and 500-ms tones reported in graph F.

The difference between the thresholds for a 500-ms tone and a 8-ms tone is large at most frequencies. At 8000 Hz however a reduction of the threshold difference is observed, which could point to an anomaly at this frequency.

Threshold/duration functions: were measured in the left ear (Figure 7.3-9B) at frequencies 250, 1000, 4000, and 8000 Hz. Tone durations were 0.008, 0.016, 0.032, 0.064, 0.128, 0.256 and 0.512 s. Thresholds are the mean of 3 measurements.

The threshold/duration functions measured in the left ear are normal at all frequencies. All functions are steep with an mean slope (based on the difference between thresholds for a 500 and 8-ms tone) of 2.5 dB/octave across all test frequencies. This value is in agreement with the values reported in the literature (see section 6.2).

The *G* and *A* estimates (Figure 7.3-9C) in the left ear were estimated by fitting the reciprocal function to the threshold/duration functions as described in section 6.4. The parameter estimates are well within the normal range at most test frequencies. At 8000 Hz the *A* value is high but still within normal limits. When we estimate *A* values for this frequency by fitting the reciprocal function based only on the 8-ms and the 500-ms thresholds presented in Figure 7.3-9A, a lower *A* value is found. The comparison of these 2 sets of values will be presented further in this chapter (Figure 7.3-13).

For the right ear, *G* and *A* parameters (Figure 7.3-9H) are estimated by fitting the reciprocal function only to the absolute thresholds of the 500-ms and 8-ms tone presented in Figure 7.3-9F. As in the left ear, the estimates are within normal limits except for the 8000 Hz which generates a high A value outside the normal range. It seems that in both ears a minor problem could be present at 8000 Hz.

Iso Forward Masking Curves: IFMCs are measured in the left ear only (Figure 7.3-9 E). Masker thresholds are measured for probe frequencies (f_p) 250, 500, 1000, 2000, 4000, 6000 and 8000 Hz. The masker frequencies (f_m) are 0.5, 0.7, 0.9, 1, 1.1, 1.3, and 1.6 * f_p. For all conditions, the probe level was set at 10 dB SPL above the 8-ms threshold. Thresholds are generally the mean of 3 measurements.

Normal IFMCs are found at all frequencies. All IFMCs are V-shaped and sharp, consistent with the results found in the other normal listeners However, as seen in listener CMa, the high frequency slopes show a downward pointing tail in the IFMCs at 4000 and 6000 Hz. This indicates that the masker with a frequency furthest away from the probe frequency is more potent at masking the probe than a masker with a frequency closer to the probe frequency. As mentioned in the report on listener CMa, an explanation for this effect is not available at the moment.

Temporal Masking Curves: TMCs were measured in the left ear (Figure 7.3-9 D) for probe frequencies 250, 500, 1000, 2000, 4000 and 6000 Hz. All functions represent the on-frequency condition with masker frequency (f_m) = probe frequency (f_p). The linear reference (dotted line) was found by fitting a straight line to the off-frequency thresholds with a probe frequency of 6000 Hz and a masker frequency (f_m) = 2400 Hz (0.4*f_p). For all conditions, the probe level was set at 10 dB SPL above the 8-ms threshold. Gap durations varied from 0.01s to 0.09 s in steps of 0.01. Thresholds are generally the mean of three measurements.

There is a clear indication of compression at all frequencies in the left ear. To allow for closer inspection, the TMCs at each frequency are shown separately in Figure 7.3-10. All TMCs show a pattern which agrees with TMCs reported in the literature. In general, the functions consist of shallow section followed by a steep section. In most TMCs a second shallow section is seen at high masker levels. At 2000 Hz the TMC is raised relative to the other probe frequencies, and the compressed section is less steep than in the TMCs at the other frequencies. This pattern cannot be categorized as abnormal but is merely less straightforward.

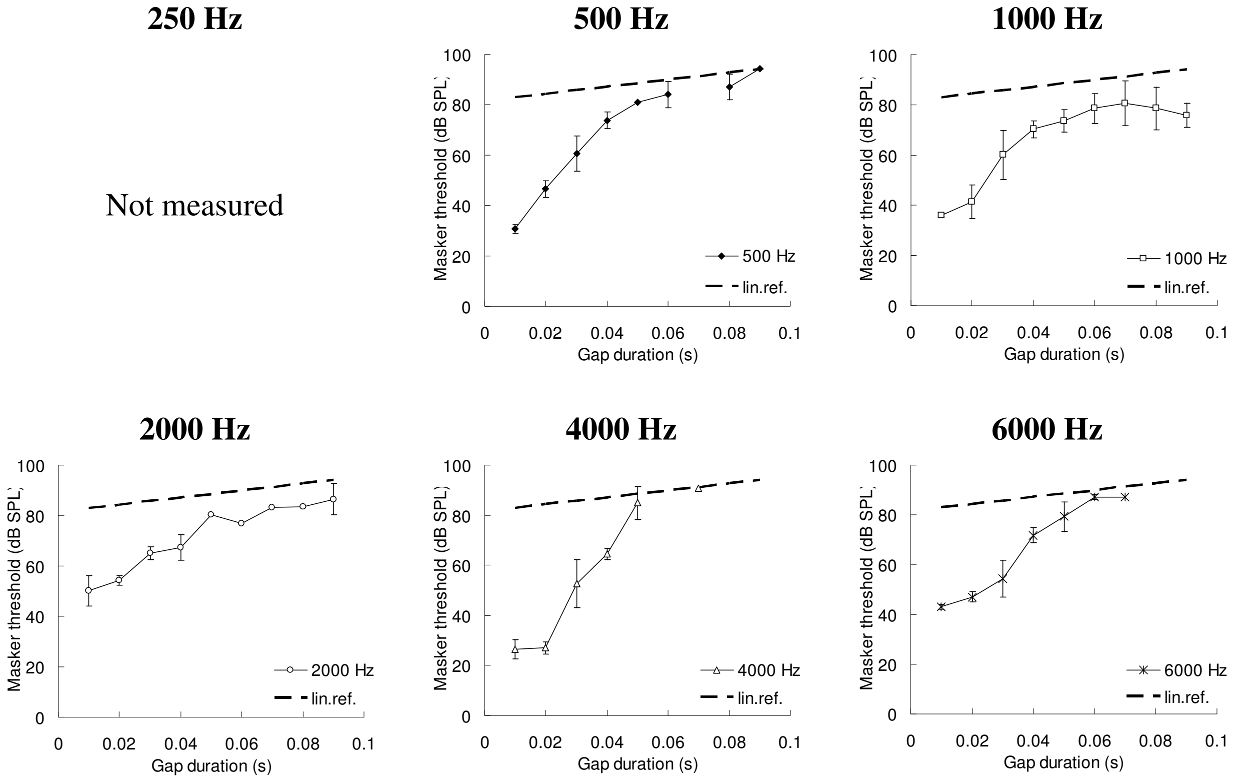


Figure 7.3-10. TMCs for listener SRa for probe frequencies (f_p) 250, 500, 1000, 2000, 4000 and 6000 Hz. Data points represent masker thresholds for gap durations from 0.01 to 0.09 (in steps of 0.1 s). Masker frequency (f_m) is set equal to the probe frequency (f_p) (on-frequency condition). The linear reference (dotted line) was found by fitting a straight line to the off-frequency data for a probe frequency of 6000 Hz and a masker frequency (f_m) = 2400 Hz ($0.4*f_p$). Thresholds are generally the mean of 3 measurements. Error bars represent 1 standard error.

Conclusion

The clinical screening confirms normal hearing for SRa. The hearing levels audiogram are within normal range. DP OAE responses are present and there is no indication of an outer/middle ear problem.

The psychoacoustic tests returned normal results in all measures. Absolute thresholds are within normal limits, threshold/duration functions are steep, TMCs show clear signs of compression and the IFMCs are sharp. At 2 different measures (absolute threshold and TMC) a cautious 'red flag' was raised for the 2000 Hz condition. However no measure showed a clear anomaly at this frequency. The origin of the downward pointing tails in the 4000 and 6000-Hz IFMC is not understood. There are no indications of possible anomalies in any of the other measures nor does there seem to be a methodological explanation for this observation.

7.3.4 Evaluation of the 'hearing profile'

In the previous section, three hearing profiles of normal listeners were presented and discussed. This section will evaluate and discuss the practicalities associated with developing psychoacoustic profiles of the listeners such as the time needed to collect the data presented in the profiles and the variability associated with the different measures. Additionally, we will evaluate the different measures and look at the possibilities of quantifying the effects associated with these measures. This will provide us with a useful basis when discussing and comparing similar profiles in impaired listeners.

Time

A major concern at the start of this project was the time it would take to collect the data needed for the profile. For the psychoacoustic profile to be a workable tool in a clinical setting, the data should be collected in a reasonable amount of time. The Single-Interval-Up-Down procedure proved to be a fast and reliable way to estimate thresholds and enabled us to collect a large data-set on each listener.

The total time needed to perform the tests was calculated for our three normal listeners and is shown in Table 7.3-1. These durations represent the total durations of the actual threshold runs.

Section A shows the total testing time per participant with the mean testing time across participants in the bottom row. No specific information was available on the duration of the clinical screenings so a fixed duration of 30 minutes was included in the total testing time of each participant to account for this part of the testing. Training time was not taken into consideration since no or minimal training was required. The general approach was to start testing after a verbal briefing and a single threshold run aimed at familiarizing the listener with the testing equipment. In the rare occasion where a listener was unsure about the task, a block of threshold runs would be presented as training. This did not exceed 5 minutes.

Section B shows the time needed per test for each participant. Again the mean testing time per measure is presented in the bottom row. The number of ears tested is shown below the test name.

Table 7.3-1. Overview of testing time. A. Total testing time (in hours and minutes) needed to collect data presented in the profiles of each participant + average testing time needed across participants. B. Testing time (in minutes) needed for each measure separately for each participant + the average time needed across participants for each measure. Between brackets is shown how many ears were tested during this time.

	A.	**B.**			
	Total testing time	Absolute Threshold (2 ears)	Threshold/ Duration (1 ear)	IFMC (1 ear)	TMC (1 ear)
CMa	9 h 9 min	76 min	104 min	114 min	138 min
MKi	9 h 48 min	42 min	74 min	189 min	186 min
SRA	7 h 3 min	34 min	62 min	160 min	137 min
Mean testing time	**8 h 40 min**	**51 min**	**80 min**	**154 min**	**154 min**

A mean testing time of 8 hours and 40 minutes was found across the three listeners. This is a very short testing time relative to the large amount of data collected and the number of repeats measured in most conditions. It is also a lot faster than testing times reported in the literature on single measure-experiments.

In absolute terms, 9 hours is still a large amount of time needed to collect the profile and is too long to be clinically viable. However, further experience with this testing protocol will undoubtedly enable us to fine tune the procedures. Certain measures might contain more useful information than others. Also, it might be decided that specific data-points could be omitted in a certain measure due to being less informative. This has already occurred for the Threshold/duration measure where it was decided that measuring thresholds for the longest and the shortest tone offered information which was equally useful to measuring the complete threshold/duration functions consisting of seven threshold measurements (see Chapter 6.4).

Moreover, it is anticipated that there could be situations where the testing time is shorter in hearing impaired listeners. The first reason for this is that some frequencies will not be testable due to difficulties in finding a threshold. Also, an audiologist might decide to omit certain frequencies from the profile. For example when a listener has normal hearing up to 1000 Hz, the data on 250 Hz might not add a major contribution to the profile and could be omitted.

In addition it must be noted that the testing procedures are fully automated. The tests do not necessarily need to be run by a clinician but could easily be handed over to a trained technician whose time will be less expensive than that of the audiologist.

Variability

A second issue we were faced with when developing the test procedures involving the psychoacoustic profile was that the threshold estimates need to be reliable. Two types of tasks are used in our testing protocol, detection in quiet and detection after a masker. In Chapter 3, predictions on the accuracy in a detection task in quiet were presented. Computer simulations showed that detection thresholds in quiet, based on a single 10 trials-run, are associated with a Standard Deviation (SD) of 1.6 dB SPL [4]. In the next section, the SDs found in our three normal listeners will be presented and compared with the predictions discussed in Chapter 3.

[4] Assuming a psychometric slope of 0.5 and a j-value of 1.4.

Standard deviations of the mean were calculated for thresholds reported in the threshold /duration measure, the IFMCs and the TMCs for each listener. The thresholds are generally the mean of three measurements. No SDs are reported for the absolute thresholds since only a single measurement was made in most cases. See Table 7.3-2.

Table 7.3-2. Standard deviation associated with the each measure. The middle rows show the SD per participant for each measure. The bottom row shows the mean SD across all 3 participants. SDs are based on the mean of 3 measurements, measured in a single ear.

	Threshold/ Duration	IFMC	TMC
CMa	1.6 dB	3.7 dB	5.4 dB
MKi	1.9 dB	5.6 dB	6.0 dB
SRA	1.8 dB	6.1 dB	7.9 dB
Mean SD	**1.8 dB**	**5.1 dB**	**6.4 dB**

The average SD of the mean found for the Threshold/duration data was 1.8 dB. This SD is based on three measurement runs. The computer simulations presented in Chapter 3 predict a SD of 0.9 dB for a mean based on three threshold runs. The difference in SD between the computer simulations and the human listeners is considerable but needs to be seen in perspective. Overall, a SD of 1.8 dB for our normal listeners is in line with standard deviations generally accepted for measurements of absolute thresholds (1-2 dB SPL). Moreover, one would expect more variability in a human listener compared to computer simulations, which represent a perfect listener. In real testing situations however, there are a number of factors which result in an increase in error such as fatigue and genuine mistakes.

The SD of the mean thresholds in the IFMCs is 5.1 dB when averaged across all listeners. The TMC thresholds returned a standard deviation of 6.4 dB. Predictions concerning the variability found for detection thresholds in the presence of a forward masker have not been presented in this thesis, as explained in Chapter 3. Procedures reported in the literature indicate that the variability for forward masking experiments is expected to be larger than in detection in quiet-experiments. Rosengard et al. (2005) reported that 80 % of their TMC thresholds, based on two runs, manifested a SD above 4 dB. Lopez-Poveda and Alves-Pinto (2008) allowed for a SD of 6 dB for thresholds based on three runs before collecting an extra run.

It has been proposed that high masker thresholds are associated with shallower slopes of the psychometric function due to the maskers being compressed at higher intensity levels. A shallow slope of the psychometric function would result in a high variability (Oxenham and Plack, 2000; Schairer, et al., 2003; Plack and Yasin, 2008). A compression coefficient of 0.2 would predict that the variability will increase by a factor of 5 compared to thresholds measured in the linear region.

Absolute threshold and the difference in threshold between a 500-ms tone and an 8-ms tone

The profiles of our listeners presented above have shown that comparing the detection thresholds for long and short tones at the same frequency can often hold useful information. The graphs A and F in the profiles show thresholds as a function of frequency for an 8-ms tone and a 500-ms tone. The function connecting the thresholds for the short tone is situated higher than and parallel to the long tone-function. Two important features should be evaluated when looking at these plots. The first is the difference between the thresholds at each frequency or the spacing between the 2 lines. The second is how similar the patterns of the lines are.

First, the difference between the 2 thresholds allows us to fit the reciprocal function discussed in Chapter 5 and estimates of the G and A parameters can be found. Practically, this means that when the difference between the thresholds is reduced, this could potentially point to an anomaly in a person’s hearing. This feature will be discussed in detail in the next section.

The second feature is the pattern of the long and the short tone functions. In general, we can assume that the functions run parallel, equally spaced from each other. Also, when measuring thresholds for frequencies in the range of 250 to 8000 Hz we expect to see a bowl-shaped function. Any deviations to this shape can hold information on anomalies. When this deviation is found in both the long as the short tone-function this strengthens the suspicion that a ‘real’ anomaly is present rather than the peak being the result of measurement error. For example, in the case of MKi, a peak in the thresholds at 2000 Hz is found both in the long and the short-tone function. The replication of this peak in both functions reinforces the idea that the peak is reliable and therefore means something. In the case of MKi this has proven to be reliable information since consecutive tests confirmed an anomaly at 2000 Hz. For listener SRa this was not the case.

G and *A* estimates

The threshold/duration functions measured in each of our normal listeners were fitted with the reciprocal function described in Chapter 6.4. In the same chapter normal values for *G* and *A* were established. *G* parameters are considered normal above 0.1 $Pa^{-1}s^{-1}$ and normal *A* parameters were found to be below 50 s^{-1}.

The *G* and *A* parameters of the three normal listeners are illustrated in Figure 7.3-11. Individual parameters are shown alongside the mean parameters across the three listeners. There are no parameter estimates at 2000 Hz for listener SRa. The mean *G* is represented by the full line, mean *A* values are represented by the dotted line. The grey shaded area illustrates the normal range of *G* values and the dashed area shows the normal range of the *A* values.

It is concluded that all *G* and *A* parameters are situated within the normal range. Some *A* estimates are at the high end of the normal scale e.g. *A* estimates at 4000 and 8000 Hz for listener MKi. In the discussion of the individual profiles this has been attributed to possible sub-clinical problems present in MKi’s hearing.

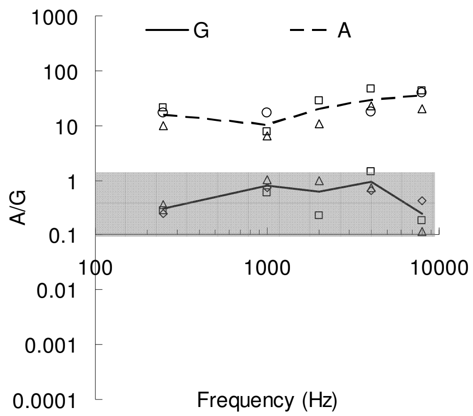


Figure 7.3-11. Estimates for parameters *G* and *A* for 3 normal listeners (CMa, MKi, SRa) at frequencies 250, 1000, 2000, 4000 and 8000 Hz. The full line represents the average *G* across the 3 listeners, the dotted line represents the average *A*-value. Individual parameter estimates are shown (CMa: open triangles, MKi: open squares, SRa: open circles). The grey area illustrates the normal range of *G* values, the dashed grey area illustrates the normal range of *A* values.

In the previous paragraph it was described how the *G* and *A* estimates of our three normal listeners are within the normal limits established in Chapter 6.4. As mentioned in Chapter 5.4 there is no need to collect all seven data-points presented here in the threshold/duration functions since selecting only the two extreme values also allows us to make a reliable estimate of the two parameters. It will now be demonstrated that the Absolute Threshold measure for a 500-ms and a 8-ms tone can provide us with similar information over a wider range of frequencies.

Figure 7.3-12 shows the *G* and *A* estimates based on two different threshold–sets. The full symbols represent the *G* and *A* parameters estimated on the basis of the threshold/duration functions (presented in graph B of the profiles) consisting of the thresholds associated with the 7 different durations. Moreover these seven thresholds are the mean of three measurements so the parameters extracted from this data can be assumed to be reliable.

The open symbols are the *G* and *A* parameters based only on the thresholds for the 500-ms and the 8-ms tone (presented in graph A of the profiles). These thresholds are generally based on a single measurement so we can expect some variability in the parameters extracted from the threshold difference. However we can conclude that the parameters are very similar regardless of the threshold-sets they are based on.

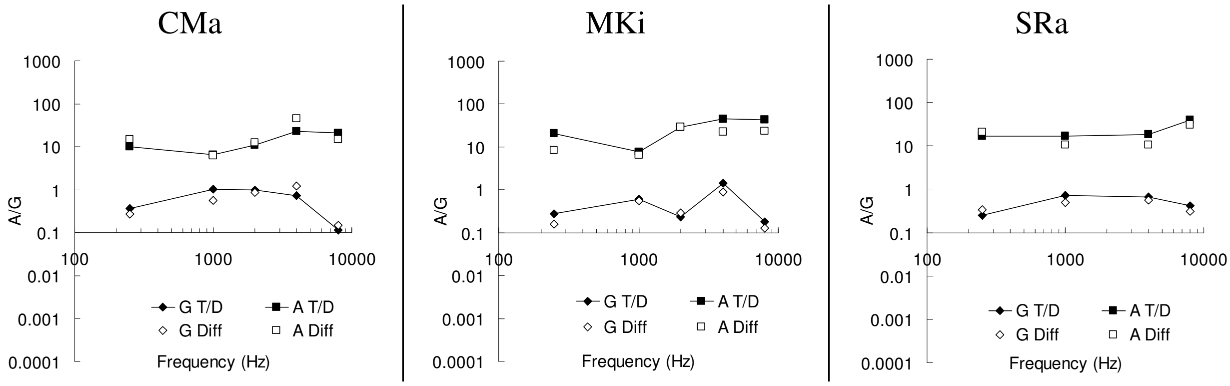


Figure 7.3-12. Comparison of *G* and *A* estimates for normal listeners (CMa, Mki, SRa) obtained using 2 different data-sets. Full symbols represent the *G* and *A* estimates based on the threshold/duration data (*G* T/D and *A* T/D). Open symbols represent *G* and *A* estimates based on the difference between the thresholds for a long and a short tone (*G* Diff and *A* Diff).

This data demonstrates that reliable *G* and *A* estimates can be obtained by a single measurement of a long and a short tone. Since more frequencies are usually tested in the Absolute Threshold measure, a wider range of parameter estimates can be obtained.

IFMCs

The IFMCs have shown to be a suitable measure to gain insight into the frequency selectivity of a listener. They give a clear picture of how one tone is affected by another tone. The IFMCs of the normal listeners will serve as a comparison to interpret the IFMCs found in impaired listeners. For this reason, it was decided that a quantification of the functions could be useful.

A common method to quantify the sharpness of tuning curves is to establish the Q10 of the curve. A high Q10 (Q for quality) is associated with a sharp curve. IFMCs are not directly comparable with tuning curves but the Q10 measure will nevertheless be applied to quantify the sharpness of the IFMCs. The procedure used to calculate the Q10s has been described in Chapter 4.4 so will not be repeated here.

Table 7.3-3 shows the Q10 values found for the IFMCs at each frequency in our 3 normal listeners. The Q10 values for MKi are not included in the mean Q10 values due to the anomaly found in the 2000 Hz region of this listener. The bottom row shows the mean Q10 for each frequency based on the Q10-values of CMa and SRa. There is a tendency for the Q10-values to increase with increasing frequency.

Table 7.3-3. Q10 estimates for 3 normal listeners for frequencies 250, 500, 100, 2000, 4000, 6000 and 8000 Hz. Row in bold shows the mean Q10 (based on CMa and SRa) as a function of frequency.

	Frequency (Hz)						
	250 Hz	500 Hz	1000 Hz	2000 Hz	4000 Hz	6000 Hz	8000 Hz
CMa	0.93	1.20	2.53	2.50	3.10	3.38	4.06
SRa		2.20	2.38	2.63	2.73	2.04	
Mean Q10 (based on CMa & SRa)	**0.93**	**1.70**	**2.46**	**2.56**	**2.92**	**2.71**	**4.06**
MKi	1.00	2.06	2.22		6.47	4.34	

The Q10 values reported in the literature are not suitable as comparative data when evaluating the sharpness of the IFMCs. Primarily because the Q10 values reported in the literature are meant to represent the bandwidth of basilar membrane tuning curves and precautions are taken to prevent off-frequency listening. As discussed earlier, the IFMCs are not a representation of a BM tuning curve but rather a representation of how sounds interfere with each other. It is therefore not appropriate to compare IFMC-Q10s with Q10 values found in the literature. Secondly, Q10 estimates are sensitive to changes in parameters and testing procedures (Stelmachowicz & Jestaedt, 1984; Vanden Abeele et al., 1992). There seems to be no clear consensus on what standard values are.

Consequently, it was no surprise to find that the estimates found in our normal listeners did not agree with various estimates found in the literature. The estimates obtained in this study are considerably lower. However, we do feel that the Q10 estimates reported in Table 6.3-3 can be useful as a comparison for the Q10 estimates that will be obtained when analyzing IFMCs of the impaired listeners. The motivation for this assumption is the fact that the same methodology will be used both in measuring the thresholds as in analyzing the IFMCs and therefore makes the results directly comparable. Interestingly, the Q10 values found in our analysis are similar to the values found in cats and guinea pigs (Evans, 1975).

TMCs

In general TMC measurements were more difficult to obtain for a number of reasons. First, in many conditions high intensity levels were needed to find the masker threshold. This carries the risk that participants might react aversively to these intense stimuli and respond in an avoiding manner. Furthermore software restrictions cause clipping of the signal at high intensity levels. In that particular situation, no threshold is obtained. This problem is expected to be more prominent in hearing impaired listeners but also occurred for our normal listeners. Second, participants did report having more difficulty with the TMC task. This could be due to the different gap durations used and the random order of the gap condition, making it less straightforward where/when to listen for the probe. Finally, the threshold estimates obtained in the TMC test were found to be more variable compared to the estimates in the IFMC paradigm.

Two measures will be used to evaluate the compression of a listener. The first is the ratio of the on-frequency TMC slope and the slope of the linear reference, the second is the range over which compression occurs. In other words the lowest level at which compression is first observed and the level at which the system returns to a linear mode. As mentioned before in section 5.5

quantification of compression is not a central aim of the project, but it does enable us to make more founded statements about the level of compression in a person's hearing.

Table 7.3-4 shows the compression coefficients at each frequency in our 3 normal listeners. The estimates for MKi are not included in the mean compression coefficient values due to the anomaly found in the 2000 Hz region of this listener.

Table 7.3-4. Compression coefficient as a function of frequency.

	Frequency (Hz)					
	250 Hz	500 Hz	1000 Hz	2000 Hz	4000 Hz	6000 Hz
CMa	0.20	0.63	0.43	0.43	0.18	0.34
SRa		0.10	0.10	0.17	0.07	0.11
Mean compression coefficient	**0.20**	**0.37**	**0.26**	**0.30**	**0.13**	**0.22**
MKi	0.31	0.19	0.19		0.41	0.47

The mean compression coefficients based on 2 listeners range between 0.13 and 0.237. This is in agreement with compression coefficients reported in the literature in the range of 0.2 – 0.3 (Lopez-Poveda et al., 2003; Stainsby and Moore, 2006). The coefficient estimates of MKi also fall within that range, especially for the low frequencies. No estimate at 2000 Hz is available due to an erratically shaped TMC at this frequency .

Table 7.3-5 shows the compression range at each frequency in our 3 normal listeners. Once more, the estimates for MKi are not included in the mean compression coefficient values.

Table 7.3-5. Compression range as a function of frequency.

	Frequency (Hz)					
	250 Hz	500 Hz	1000 Hz	2000 Hz	4000 Hz	6000 Hz
CMa	44-84 dB	35-70 dB	24-80 dB	27-87 dB	33-84 dB	42-89 dB
SRA		30-79 dB	33-76 dB	47-77 dB	29-86 dB	47-83 dB
Mean Compression range	**44-84 dB**	**32-74 dB**	**28-78 dB**	**37-82 dB**	**31-85 dB**	**45-86 dB**
MKi	50-91 dB	56-88 dB	64-77 dB		24-66 dB	21-57 dB

The mean compression range is similar at most frequencies ranging from 35 – 80 dB SPL. This is in agreement with the compression range reported by Nelson and colleagues (2001) ranging between 40-80 dB SPL. The compression ranges found in listener MKi are reduced compared to the compression ranges found in the two other listeners. At the low frequencies the lower compression threshold is raised and at the high frequencies the upper compression threshold is reduced.

It is concluded that the compression measures used yield values in agreement with the literature.

7.4 Conclusion

A comprehensive hearing profile of three normal listeners was developed. Information on absolute thresholds, threshold/duration effect, frequency selectivity and compression proves to be a rich source of information of a person's hearing. Although no abnormalities were expected to be found, the micro-analysis did highlight some sub-clinical features.

Overall, the profiles found in our normal listeners show a very similar pattern and correspond to data found in the literature for these various tests for normal listeners. All absolute threshold measures are within normal limits. The Threshold/duration analysis and the related G and A parameter estimates lie within the normal limits established in Chapter 5. All listeners show sharp IFMCs except for listener MKi whose IFMC at 2000 Hz is W-shaped in stead of V-shaped. The Q10- estimate proved to be a viable measure of evaluating the sharpness of the IFMCs. The TMCs generally consist of a shallow section followed by a steep section, indicating basilar membrane compression with compression coefficients and compression ranges in agreement with the literature.

Not all the data collected fitted the normal ideal. Listener MKi appears to have an anomalous 2000 Hz-BM site. The evidence first emerged when measuring the IFMC at this frequency. Furthermore anomalous TMC measurements emerged at the same frequency and the participant did indicate difficulties in performing the tasks when a stimulus of 2000 Hz was involved.

The precise origin of this 2000 Hz problem is yet to be determined but it can be hypothesized that an unresponsive region is situated at the 2000 Hz site of the basilar membrane between 1400 and 2600 Hz. This finding is interesting as the absolute thresholds were within normal limits. However, when taking a closer look an increase in the thresholds at 2 kHz was seen nevertheless still within normal range. On the one hand, this peak could be part of the normal detection threshold microstructure as discussed by Cohen (1982)[5]. However if it emerges that these out-of-character peaks are associated with sub-clinical problems in normal listeners, this could be of clinical importance.

A second anomaly concerns the downward pointing tails of some IFMCs. At least one of those is observed in each listener. No explanation is available for this effect for the moment.

The threshold estimation procedure developed during this thesis project (see Chapter 2, 3 and 4) proved to be a efficient way of acquiring the large amount of data needed to develop the profile of a person's hearing, both in terms of speed and accuracy. On average less than 9 hours were needed to complete the standard set of data collection. In general the variability was low with a mean varying from 1.8 dB for the absolute threshold measurements to 6.8 dB in the TMC measurement.

In conclusion we can state that the profiles described above provide us with detailed information on a person's hearing and will form a useful guide to analyzing the profiles found in the impaired listeners.

[5] Cohen (1982) reported that the detection thresholds can vary substantially with (very small) changes in the signal frequency in normal listeners, showing a detection threshold microstructure with distinct peaks and valleys.

CHAPTER 8

Hearing profiles of impaired listeners

8.1 Introduction

The psychoacoustic profiles of normal listeners described in the previous chapter confirm that both the threshold estimation procedure discussed in Chapter 2 and the test battery explained in Chapter 5 and 6, enable us to develop a detailed picture of a person's hearing.

In this chapter this exercise will be expanded to include profiles of impaired listeners. The structure of this chapter is very similar to the previous chapter. Similar procedures and paradigms were used as the ones described in the previous chapter. However certain conditions did force us to adjust parameters to the specific hearing losses of our participants, such as the frequencies which could be tested. The profiles of the impaired listeners will subsequently be used to make a detailed evaluation of the hearing of the listener.

8.2 Method

The data-sets/profiles discussed in this chapter were collected over several testing sessions. The initial session was aimed at screening and familiarizing the participant with the testing environment and tasks. Consecutive sessions were usually 90 minutes long with 2 to 3 5-minute breaks. When possible each session focused on one specific paradigm or task.

8.2.1 Tests

On the first testing session, an audiological assessment was performed to explore the hearing of the participant. The tests included in this assessment were otoscopy, pure-tone audiometry, tympanometry, DP OAE measurements and Rinne tuning fork test.

The psychoacoustic tests performed were Absolute Thresholds measurements for 500 ms and 8-ms tones, Threshold/Duration functions, Temporal Masking Curves (TMC) and Iso Forward Masking curves (IFMC). The mechanisms behind the paradigms and their value for the hearing profile were discussed in Chapter 4 and 5. The specific parameters used in each test are discussed below. All listeners were tested according to this set of tests and conditions unless stated otherwise.

Absolute threshold

Detection thresholds were measured for 500-ms and 8-ms pure tones (raised cosine onset and offset times of 4 ms) at frequencies 250, 500, 1000, 2000, 4000 and 8000 Hz. Thresholds are usually based on a single measurement.

Thresholds as a function of tone duration

Detection thresholds were measured for pure tones for seven different durations (0.008, 0.016, 0.032. 0.064, 0.128, 0.256 and 0.512 s) at frequencies 250, 1000, 4000 and 8000 Hz. Ramps were raised cosine onset and offset times of 0.004 s. Each threshold is the mean of three measurements.

The predictive reciprocal function described in section 6.4 was fitted to the threshold/duration functions and estimates for parameters *A* and *G* were calculated for each test frequency according to the procedure discussed in Chapter 6.

Iso Forward Masking Curves (IFMC)

Masker thresholds were measured for probe frequencies (f_p) 250, 500, 1000, 2000, 4000 and 6000 Hz. The masker frequencies (f_m) were 0.5, 0.7, 0.9, 1, 1.1, 1.3, and 1.6 * f_p. For all conditions, the probe level was set at 10 dB SPL above the 8-ms threshold of the probe. Masker and probe durations were 0.108 s and 0.008 s respectively. In the 250 Hz condition, masker and probe duration were 0.108 s and 0.016 s respectively. The probe duration was increased to 0.016 s at 250 Hz to allow at least 4 cycles in the stimulus. The stimuli were ramped with raised cosine onset and offset times of 0.004 s. The gap duration was 0.01 s.

Each threshold is the mean of three measurements except for conditions where no threshold could be obtained.

Temporal Masking Curves (TMC)

Masker thresholds were measured for nine different gap durations 0.01, 0.02, 0.03, 0.04, 0.05, 0.06, 0.07, 0.08 and 0.09 s for probe frequencies (f_p) 250, 500, 1000, 2000, 4000 and 6000 Hz. The gap duration was defined as the duration of the silence period between the masker offset and the probe onset. In the on-frequency condition, the masker frequency (f_m) = f_p. For all conditions, the probe level was set at 10 dB SPL above the 8-ms threshold of the probe. Each threshold is the mean of three measurements except for conditions where clipping occurred and no threshold could be obtained. In the off-frequency condition the masker frequency (f_m) was set at 0.55 * probe frequency (f_p). The use of a masker frequency = 0.4 * probe frequency (f_p), as used for the normal listeners, was not possible since he masker levels needed for these thresholds all exceeded clipping thresholds. The probe frequency in the off-frequency condition was set at the highest possible frequency which would return at least three off-frequency masker thresholds below the clipping threshold. The linear reference was subsequently found by fitting a straight line to the off-frequency thresholds. The rationale behind this choice of linear reference was discussed in Chapter 5.5. Masker and probe durations were 0.108 s and 0.008 s respectively. In the 250 Hz condition, masker and probe duration were 0.108 s and 0.016 s respectively. The probe duration was increased to 0.016 s at 250 Hz to allow at least 4 cycles in the stimulus. The stimuli were ramped with raised cosine onset and offset times of 0.004 s. The compression coefficient and the compression range were calculated using the method described in chapter 5.5.

8.2.2 Threshold estimation procedure

The data presented in this chapter were collected over a period of 2 years. In this time our testing procedure was not yet finalized and therefore minor changes in the procedure occurred during this period. For most listeners 1 or 2 modifications were implemented during the data-collection of their profile. As discussed in chapter 2, these changes are related to the method of selecting the level of the next stimulus and to the method used to fit the psychometric function. The improvements do not affect the threshold estimates but rather result in faster or more reliable threshold estimates.

The different procedures used were Maximumlikelihood (ML), Single interval Least Squares 50 % (SI LS 50%) and Single-Interval-Up-Down (SI UD). A detailed description of these procedures can be found in Chapter 2.3. The default number of Threshold trials used in all procedures was 10 trials, not including catch trials and trials in the Initial phase. Table 8.2-1 provides an overview of the procedures used in each test and each listener.

8.2.3 Listeners

Four impaired listeners were tested extensively. The participants were recruited within the university and received reimbursement for travel expenses only. When the performance in the left and the right ear was similar, as is the case for participants MFr and JEv, the left ear was tested. The right ear was tested for participants SFo and BMe, respectively due to tinnitus or normal hearing in the left ear. The tests were administered at different points in time over a one-year period.

Table 8.2-1. Overview of procedures used for each test for 3 normal listeners. ML=Maximum likelihood, SI 50%=Single Interval 50 %, SI-UD=Single-Interval-Up-Down.

	MFr	JEv	SFo	BMe
Absolute threshold	ML (5 trials)	ML (5 trials)	SI 50 %	SI 50 %
Threshold/duration	ML (10 trials)	ML (10 trials)	SI 50 %	SI 50 %
IFMCs	ML (5 trials)	SI 50 %	SI 50 %/UD	SI 50 %/UD
TMCs	SI 50 %	SI 50 %	SI 50 %	SI 50 %

8.2.4 Apparatus

Pure tone audiometry and tympanometry were performed using a Kamplex AA222 combination audiometer/impedance test station (PC Werth, London, UK). Distortion product OAEs were measured using a dual channel ILO292 Echoport USB-II (Otodynamics Ltd, Hatfield UK).

For all psychoacoustic tests, stimuli were generated using a sampling rate of 96000 Hz, with 24-bit resolution. They were played monaurally via circumaural Sennheiser HD600 headphones to the participant seated in a double-walled sound attenuating booth.

8.3 Results

8.3.1 Listener JEv

JEv is 69 year old male. He has been experiencing gradual hearing loss over the last 20 years. JEv worked as a mining engineer so industrial hearing loss could be a contributor to the hearing problem.

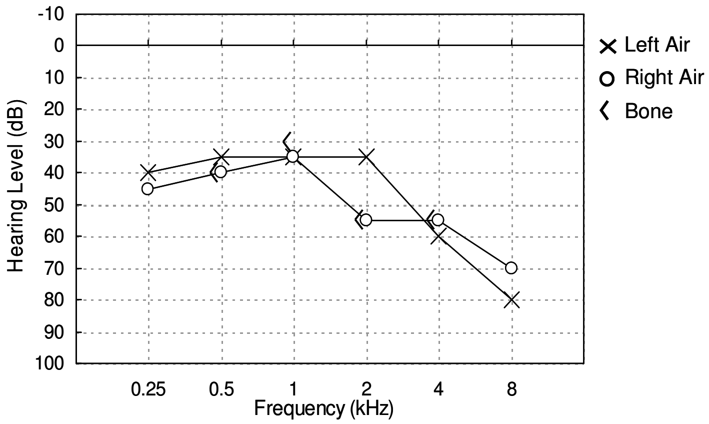
A. Audiological assessment:

Otoscopy: Normal

Tympanometry: Normal

Rinne: Positive

A. Audiogram



B. DP OAE

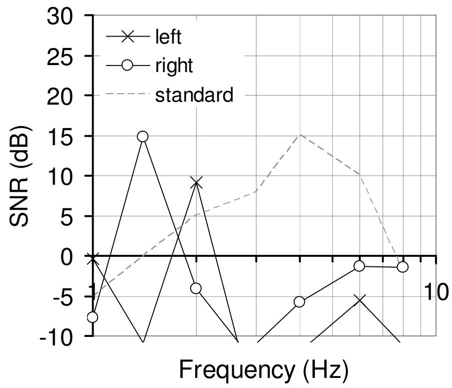


Figure 8.3-1. A. Audiogram: air conduction thresholds at 0.25, 0.5, 1, 2, 4 and 8 kHz for left and right ear. Bone conduction thresholds at 0.5, 1, 2 and 4 kHz. B. DP OAE responses for frequencies 1, 1.4, 2, 2.8, 4, 6 and 8 kHz for left and right ear. Responses are expressed in SNR (dB). The dotted grey line represent the 5th percentile of responses of normal listeners reported by Gorga et al. (1997).

JEv has a symmetrical bilateral sensorineural hearing loss (Figure 8.3-1). The low frequencies show a flat loss of about 35 dB SPL, gradually increasing at the higher frequencies. There is no indication for an outer/middle ear problem. DP OAE responses in both ears are absent at all frequencies except at 1.4 kHz in the right ear and 2 kHz in the left ear.

Psychoacoustic tests

Absolute thresholds: Thresholds were measured for pure tones with tone durations of 500 and 8 ms at frequencies 250, 500, 1000, 2000 and 4000 Hz both in the left and the right ear (Figure 8.3-2A & F). No thresholds could be found at 8000 Hz. Thresholds are based on a single measurement.

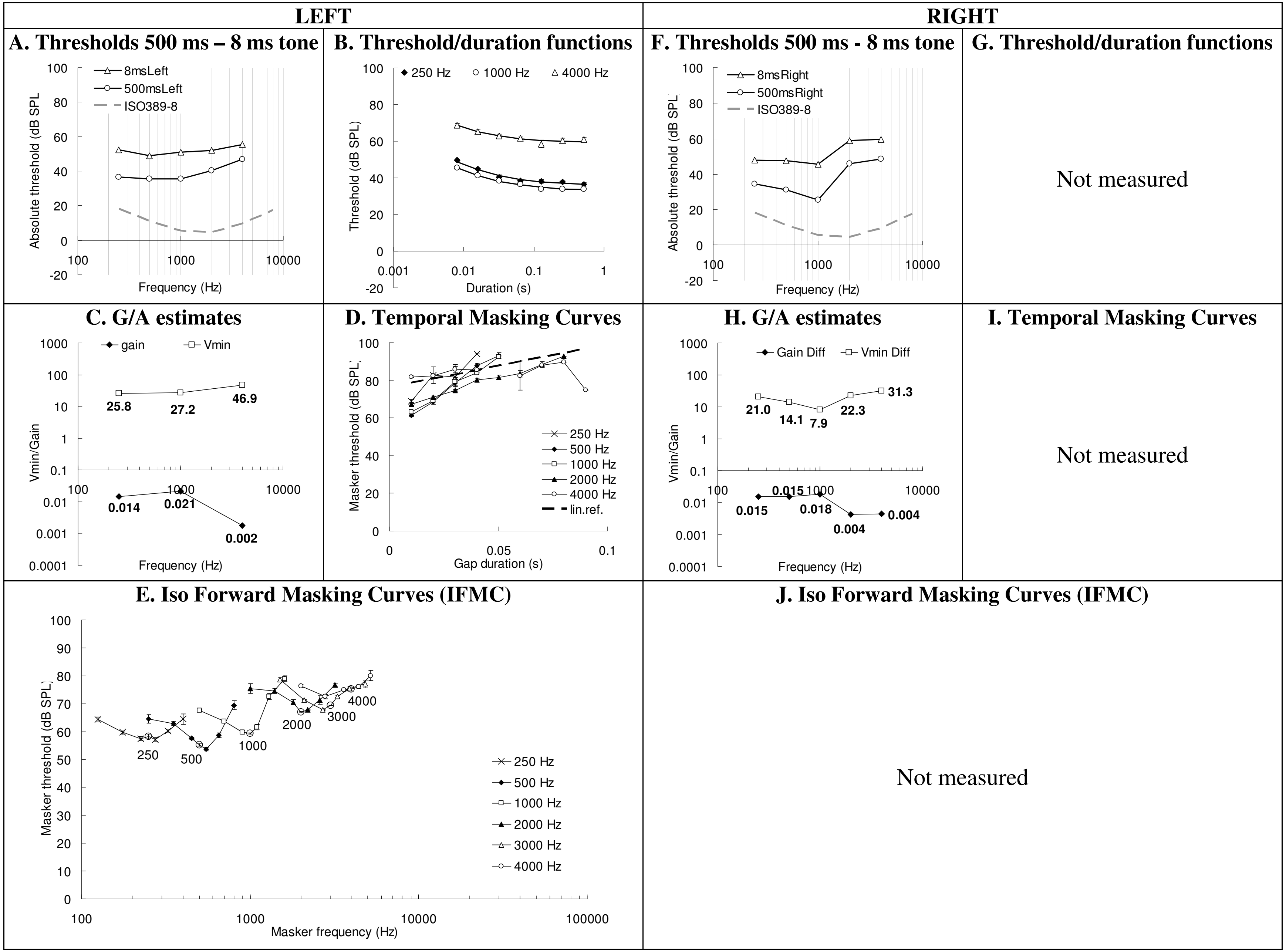


Figure 8.3-2: Impaired listener JEv
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Figure 8.3-2. Psychoacoustic profile of impaired listener JEv.

A. Left ear: Absolute thresholds as a function of frequency for 500-ms tones (open circles) and 8-ms tones (open triangles) at frequencies 250, 500, 1000, 2000 and 4000 Hz. Thresholds are based on a single measurement.

B. Left ear: Thresholds as a function of tone duration at frequencies 250, 1000 and 4000 Hz. Each threshold is the mean of 3 threshold measurements. Error bars represent 1 standard error. The full lines represent predictions made using the reciprocal function described in section 6.4. Each fitted function provides an estimate of parameter *A* and *G* presented in Graph C.

C. Left ear: *G* (filled diamonds) and *A* (open squares) estimates at frequencies 250, 1000 and 4000 Hz, associated with the predictive functions (full lines in graph B) to the threshold/duration functions reported in graph B.

D. Left ear: Temporal Masking Curves (TMC) representing masker thresholds as a function of gap duration for probe frequencies (f_p) 250, 500, 1000, 2000 and 4000 Hz. All TMCs represent the condition where the masker frequency $f_m = f_p$ (on-frequency condition). The linear reference (dotted line) represents the off-frequency condition for f_p = 4000 Hz and f_m = 2200 Hz (0.55 * f_p). Gap durations vary from 0.01 s to 0.09 s in steps of 0.01. Thresholds are generally the mean of 3 measurements. Error bars represent 1 standard error.

E. Left ear: Iso Forward Masking Curves (IFMC) for probe frequencies (f_p) 250, 500, 1000, 2000, 3000 and 4000 Hz. The masker frequencies (f_m) are 0.5, 0.7, 0.9, 1, 1.1, 1.3, and 1.6 * f_p. The masker threshold at probe frequency is represented by the large open circle. The accompanying value indicates the probe frequency. Thresholds are generally the mean of 3 measurements. Error bars represent 1 standard error.

F. Same as A but for right ear.

H. Right ear: estimates of parameters *G* (filled diamonds) and *A* (open squares) estimates based only on the difference in absolute threshold between the 8 ms and 500-ms tones reported in graph F.

In both ears absolute thresholds for long tones are raised with a small loss up to 1000 Hz and an increasing loss at high frequencies (Figure 8.3-2A). The difference between the 500-ms tones and the 8-ms tones is large up to 1000 Hz but reduced from 1000 Hz onwards suggesting a different or larger problem at these high frequencies compared to the low frequencies.

Threshold/duration functions: were measured in the left ear (Figure 8.3-2B) at frequencies 250, 1000 and 4000 Hz. No threshold/duration function was measured at 8000 Hz. Tone durations were 0.008, 0.016, 0.032, 0.064, 0.128, 0.256 and 0.512 s. Thresholds are the mean of three measurements.

All threshold/duration functions are raised. At 250 and 1000 Hz the slope is steep but a shallower slope is found at 4000 Hz. The mean slope (based on the difference between thresholds for a 500 and 8-ms tone) is 1.8 dB/octave across all test frequencies and 2.2, 1.9 and 1.3 respectively at 250, 1000 and 4000 Hz. The values at 1000 and 4000 Hz are lower than the values found in our normal listeners (Chapter 7). However the value found at 250 Hz is very similar to values found in the normal listeners.

The *G* and *A* parameters in the left ear (Figure 8.3-2C) were estimated by fitting the reciprocal function to the threshold/duration functions as described in section 6.4. All frequencies show an abnormal (reduced) *G*. These *G* values represent the raised thresholds at all three frequencies. The *A* values however are well within the normal range, representing the steep slope. All *A/G* combinations can be situated in Quadrant II (see Chapter 6.4).

For the right ear, *G* and *A* parameters (Figure 8.3-2H) were obtained by fitting the same reciprocal function to the absolute thresholds of the 500ms and the 8-ms tones at different frequencies presented in Figure 8.3-2F. Again abnormal *G* values are found at all frequencies but the *A* values are within normal range.

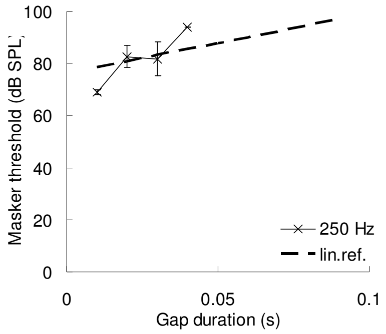
Iso Forward Masking Curves: IFMCs were measured in the left ear (Figure 8.3-2E) for probe frequencies (f_p) 250, 500, 1000, 2000, 3000 and 4000 Hz. The 3000-Hz IFMC was added to the basic set of measurements to explore the transition between 2000 Hz and 4000 Hz. The masker frequencies (f_m) are 0.5, 0.7, 0.9, 1, 1.1, 1.3 and 1.6 * f_p. For all conditions, the probe level was set at 10 dB SPL above the 8-ms threshold of the probe. Thresholds are generally the mean of 3 measurements.

IFMCs are raised and shallower than normal at all frequencies especially from 2000 Hz onwards. At most frequencies, the IFMCs are V-shaped and some tuning can be found. Q10 values are 0.63, 1.11, 1.26 for IFMCs at low probe frequencies (250, 500 and 1000 Hz respectively). This is lower than normal but the deviation from normal is less than for the higher frequencies. At 2000, 3000 and 4000 Hz, Q10s of respectively 0.84, 0.90 and 0.75 are found. These Q10s represent a substantial reduction in sharpness. The 4000-Hz IFMC shows little evidence of tuning and its high-frequency tail seems to coincide with the high-frequency tail of the 3000-Hz IFMC. The lowest masker threshold in the 3000-Hz IFMC is situated at 2700 Hz and at 2800 Hz in the 4000-Hz IFMC. This indicates that probes with frequencies above 2700/2800 Hz are always most effectively masked by a 2700 Hz masker. This raises the question if the 2700 Hz- site of the BM harbours the ‘last filter’. Is it possible that no tuning is present beyond this point and a 2700 Hz is the boundary of an unresponsive region?

Temporal Masking Curves: TMCs were measured in the left ear (Figure 8.3-2D) for probe frequencies 250, 500, 1000, 2000 and 4000 Hz. All functions represent the on-frequency condition with the masker frequency (f_m) = probe frequency (f_p). The linear reference (dashed line) was found by fitting a straight line to the off-frequency thresholds with a probe frequency of 4000 Hz and a masker frequency (f_m) = 2200 Hz (0.55 * f_p). For all conditions, the probe level was set at 10 dB SPL above the 8-ms threshold of the probe. Gap durations varied from 0.01 to 0.09 s in steps of 0.01. Thresholds are generally the mean of three measurements. Compression range and coefficient were determined using the method discussed in Chapter 5.5.

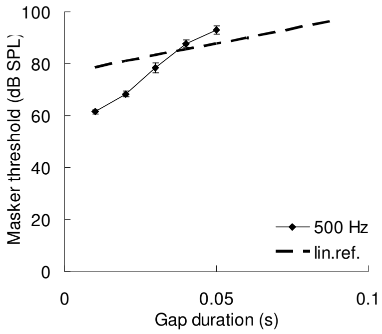
Based on visual inspection, it is concluded that TMCs for 250, 500, 1000 Hz and 2000 Hz show some indication of compression. There is no evidence for compression in the 4000-Hz TMC.
To allow for closer inspection, the TMCs at each frequency are shown separately in Figure 8.3-3. Below each TMC the compression ratio and compression range for that frequency are shown. The TMCs for probe frequencies 250, 500, 1000 and 2000 Hz have slopes which are steeper than the slope of the linear reference. This is a clear indication that compression is still present. For probe frequency 4000 Hz, the masker thresholds run parallel to the linear reference suggesting very limited or absent compression. The compression coefficients for 250, 500 and 1000 Hz agree with values found in normal listeners. The coefficient at 2000 Hz is clearly higher than normal and at 4000 Hz a compression coefficient of 1 represents total absence of compression. The compression range is clearly reduced compared to compression ranges found in normal listeners. Especially in the 2000-Hz TMC the range is limited. Interestingly but not surprisingly, masker thresholds could be measured for most gap durations for the frequencies with clearly reduced or absent compression. The lack of compression prevented the masker thresholds from exceeding the clipping threshold and the masker thresholds remained just below clipping-point.

250 Hz



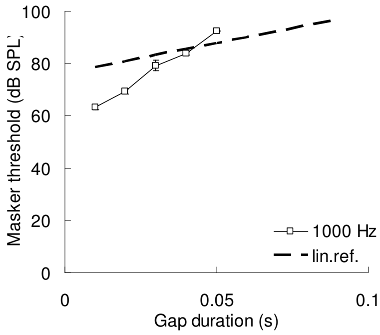
Coefficient	0.23
Range	68-85 dB SPL

500 Hz



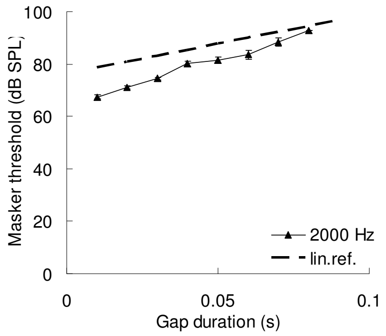
Coefficient	0.24
Range	62-92 dB SPL

1000 Hz



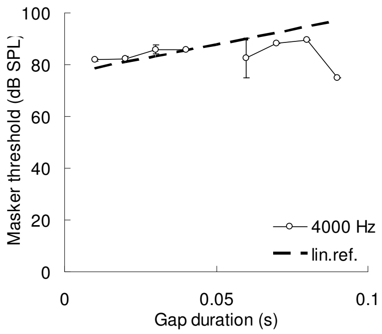
Coefficient	0.31
Range	62-93 dB SPL

2000 Hz



Coefficient	0.51
Range	63-82 dB SPL

4000 Hz



Coefficient	1
Range	/

6000 Hz

Not measured

Figure 8.3-3. TMCs for listener JEv for probe frequencies (f_p) 250, 500, 1000, 2000 and 4000 Hz. Data points represent masker thresholds for gap durations from 0.01 to 0.09 (in steps of 0.1 s). Masker frequency (f_m) was set equal to the probe frequency (f_p) (on-frequency condition). The linear reference (dotted line) was found by fitting a straight line to the off-frequency data for a probe frequency of 4000 Hz and a masker frequency (f_m) = 2200 Hz (0.55 * f_p). Thresholds are generally the mean of 3 measurements. Error bars represent 1 standard error. Compression coefficient and range for each test frequency are presented below the relevant TMC-graph and were calculated using the method described in chapter 5.5.

Speech recognition: As part of a pilot study looking into measures for the recognition of speech in a noisy environment, a speech recognition test was performed. The data obtained from this measure has not been implemented in the hearing profile but will be presented for informative purposes.

Method: For speech recognition-testing, the Boothroyd AB-wordlists were employed (Boothroyd, 1968). The listener was presented with lists of Consonant-Vowel-Consonant (CVC) words. Each list consists of 10 words. The listener was asked to verbally repeat each presented word. Each list is scored as phonemes correct out of 30. This score is then converted into a percentage of correct phonemes. One list was presented per condition. Speech level was always fixed at 70 dB SPL. The software restricted the testing of any other speech levels. The noise used was multitalker babble and the noise level was varied according to different SNR-conditions. Stimuli and noise were presented to both ears via circumaural headphones. The recordings are copyright © retained by the University of Southampton (1998). The software used to present the stimuli and the noise is copyright © retained by MRC-IHC (1998 and 2005).

Initially, three lists were presented with no noise present. Subsequently, speech performance in noise was tested at 5 different SNR-levels (20, 15, 10, 5, 0). The SNR-level is defined as the difference in intensity level between the stimulus and the noise. A positive SNR represents a stimulus which is louder than the noise.

Results: When listening to speech in quiet an average performance of 55% was obtained (Q in figure 8.3-4) across the 3 lists. When introducing noise, performance drops with decreasing SNR. Speech recognition decreases dramatically from SNR=15 onwards. At SNR 0, hardly any phonemes can be identified. The overall performance could be affected by the speech level at 70 dB which probably was too low for the hearing loss found in this listener.

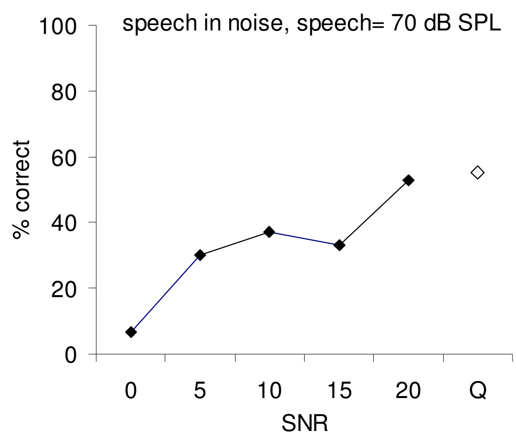


Figure 8.3-4. Speech in noise-test: Percentage correct identification of phonemes in CVC-words in quiet (Q, open diamond) and in noise (SNR = 20, 15, 10, 5, 0)

The speech recognition test shows that even in quiet, the speech recognition of JEv is significantly impaired. Subsequently, when introducing only a small amount of noise, performance is reduced considerably. This supports the findings discussed in the IFMC-measurement that tuning is reduced or absent at all frequencies. Any noise added to a signal will therefore be a effective masker reducing the speech recognition performance of the listener.

Interpretation

A standard clinical screening confirmed that JEv has a moderate hearing loss at low frequencies and a moderately severe to severe hearing loss at the high frequencies. There is no indication of an outer/middle ear problem. DPOAE responses were weak except at 1400 Hz in the right ear and at 2000 Hz in the left ear where a strong response was registered. Threshold/duration functions generate abnormal *G* estimates due to the raised thresholds but normal *A* estimates. Compression is present up to 2000 Hz but is reduced at 2000 Hz and absent at 4000 Hz. IFMCs show reduced sharpness at all frequencies with possibly an unresponsive region from 2700 Hz onwards.

We might speculate that the origin of the hearing loss described above is a combination of OHC and IHC damage with possibly no functioning hair cells from 2700 Hz onwards (unresponsive region). Below 2000 Hz, tuning is present but reduced. Also some compression was found in this frequency region. Furthermore strong DP OAE responses were recorded at 1.4 and 2 kHz in the right and left ear respectively. All these findings strengthen the idea that IHC damage is combined with some OHC damage. From 2000 Hz onwards tuning is reduced or absent and hardly any compression was found. No DPOAE responses were found in this frequency region. It is suggested that extensive IHC/OHC damage is present here with possibly a non-responsive region from 2700 Hz onwards. Interestingly, it can be pointed out that normal *A* values (combined with abnormal *G* values) are found at all frequencies. This could offer a clue to the hearing impairments associated with G/A combinations situated in Quadrant II (section 6.4) and could suggest that this quadrant is associated with IHC/OHC damage.

The profile described above might be representative for a noise-induced loss. However, a reasonable hypothesis worth exploring is that the endocochlear potential (EP) is reduced. This would affect both the IHC and the OHC. This finding is in agreement with Schmiedt and colleagues (2002) who described the effects of a reduction of the EP on the functioning of the IHC and OHC. They suggested that a reduction in EP will affect high frequencies more than low frequencies.

8.3.2 Listener SFo

SFo is a 58-year old female whose hearing complaints started about 5 years ago. She reports tinnitus in both ears with a more disturbing tinnitus (high pitched hissing sound) on the left ear. The tinnitus onset coincided with the onset of the hearing loss. There is a possibly a hereditary factor in her hearing problems since both parents had hearing loss. One sibling has a conductive hearing loss.

Audiological assessment:

Otoscopy: Normal

Tympanometry: Normal

Rinne: Positive

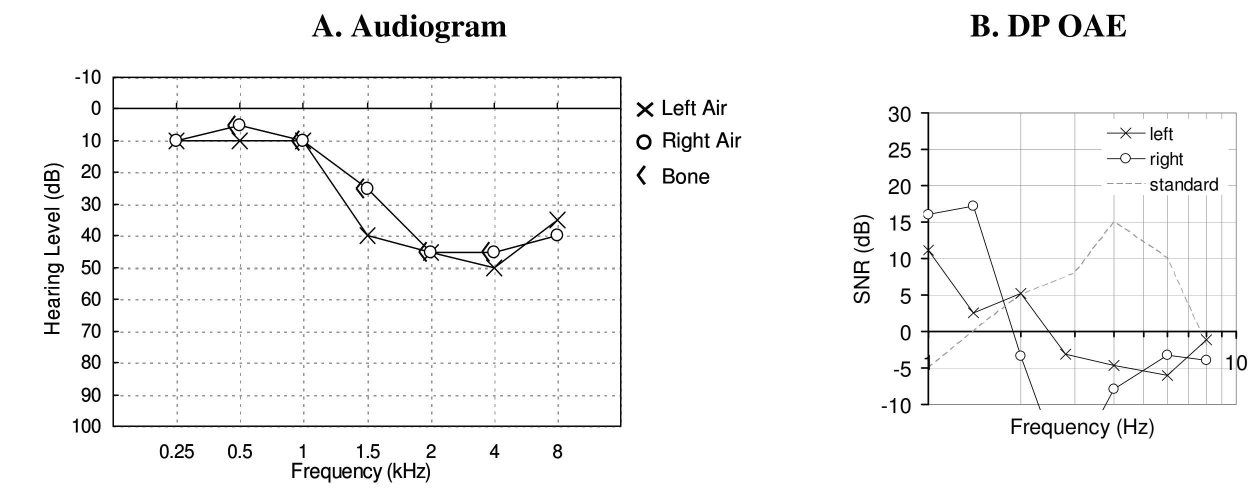


Figure 8.3-5. A. Audiogram: air conduction thresholds at 0.25, 0.5, 1, 2, 4 and 8 kHz for left and right ear. Bone conduction thresholds at 0.5, 1, 2 and 4 kHz. B. DP OAE responses for frequencies 1, 1.4, 2, 2.8, 4, 6 and 8 kHz for left and right ear. Responses are expressed in SNR (dB). The dotted grey line represent the 5th percentile of responses of normal listeners reported by Gorga et al. (1997).

SFo shows a symmetrical bilateral moderate hearing loss with normal audiometric thresholds up to 1 kHz (Figure 8.3-5). Thresholds increase sharply from 1 kHz onwards. Beyond 2 kHz there is no further decrease in thresholds but a flat hearing loss around 45 dB HL. Distortion product OAEs (DPOAEs) were present up to 2 kHz but absent for the higher frequencies. Otoscopy and tympanometry were normal. The Rinne tuning fork test was positive in both ears.

Psychoacoustic tests:

Absolute thresholds: Thresholds were measured for pure tones with tone durations of 500 ms and 8 ms at frequencies 250, 500, 1000, 1500, 2000, 2500, 3600, 4000, 5000, 6000, 6600, 7000 and 8000 Hz in both ears (Figure 8.3-6A & F). An extended range of frequencies was tested to explore the frequency region where raised thresholds were observed.

Both in the left and right ear, SFo shows normal thresholds for long tones up to 1000 Hz. There is a sharp threshold increase with a flat loss from 1500 Hz onwards.

The difference between the 8 ms and the 500-ms tone thresholds is strongly reduced from 1000 Hz onwards. The pattern of absolute threshold functions for the long and the short tone is similar.

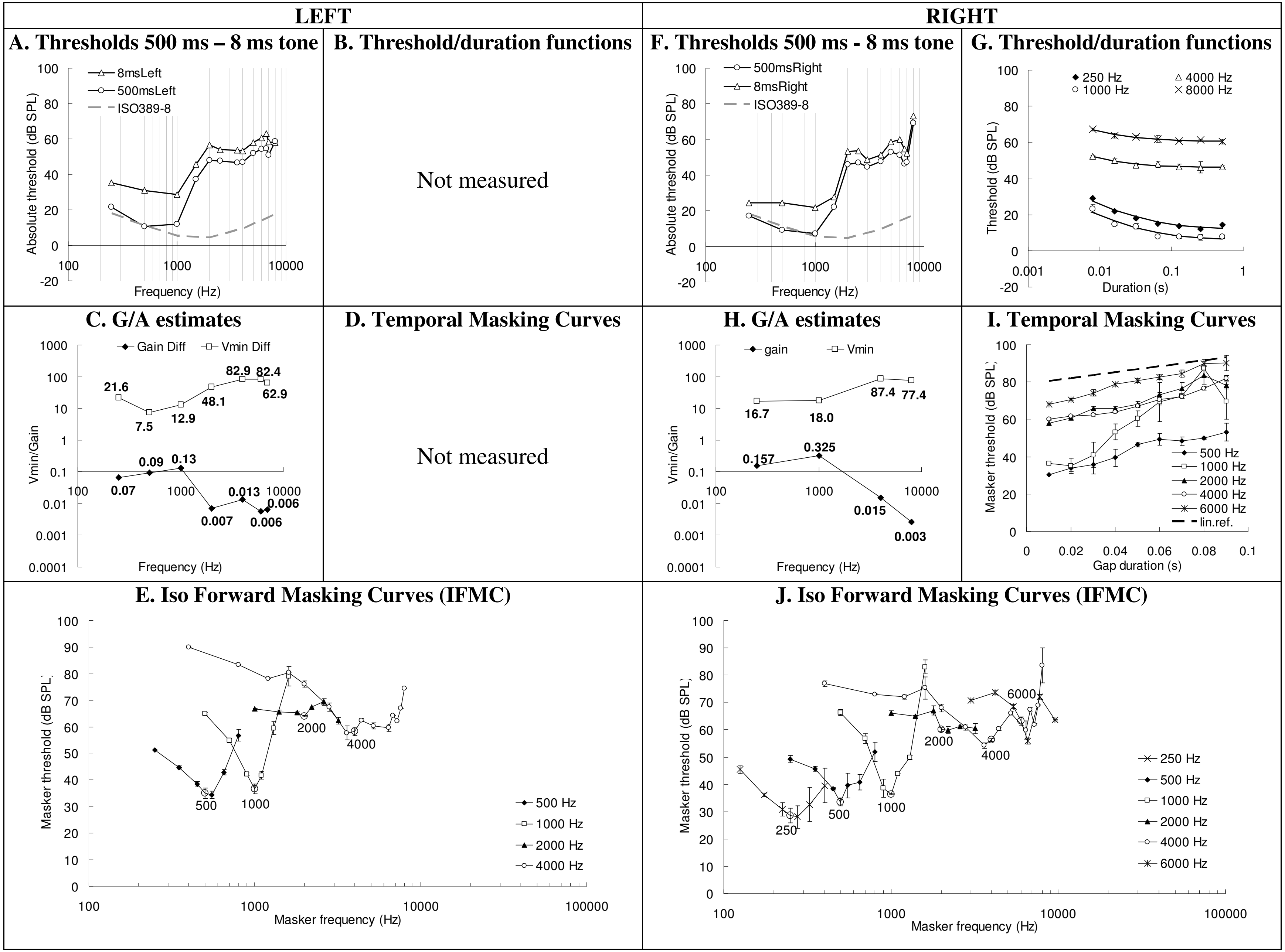


Figure 8.3-6: Impaired listener SFo.
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Figure 8.3-6. Psychoacoustic profile of impaired listener SFo.

A. Left ear: Absolute thresholds as a function of frequency for 500-ms tones (open circles) and 8-ms tones (open triangles) at frequencies 250, 500, 1000, 1500, 2000, 2500, 3600, 4000, 5000, 6000, 6600, 7000 and 8000 Hz. Thresholds are based on a single measurement.

C. Left ear: estimates of parameters *G* (filled diamonds) and *A* (open squares) estimates based only on the difference in absolute threshold between the 8 ms and 500-ms tones reported in graph A.

E. Left ear: Iso Forward Masking Curves (IFMC) for probe frequencies (f_p) 500, 1000, 2000 and 4000 Hz. The masker frequencies (f_m) are 0.5, 0.7, 0.9, 1, 1.1, 1.3, and 1.6 * f_p. IFMC with probe frequency 4000 Hz shows extra masker thresholds for masker frequencies 0.1, 0.2, 0.3, 0.4, 1.7, 1.8, 1.9 and 2 * f_p. The masker threshold at probe frequency is represented by the large open circle. The accompanying value indicates the probe frequency. Thresholds are generally the mean of 3 measurements. Error bars represent 1 standard error.

F. Right ear: Same as A.

G. Right ear: Thresholds as a function of tone duration at frequencies 250, 1000, 4000, and 8000 Hz. Each threshold is the mean of 3 measurements. Error bars represent 1 standard error. The full lines represent predictions made using the reciprocal function described in section 6.4. Each fitted function provides an estimate of parameter *A* and *G* presented in Graph C.

H. Right ear: *G* (filled diamonds) and *A* (open squares) estimates at frequencies 250, 1000, 4000 and 8000 Hz, associated with the predictive functions (full lines in graph B) to the threshold/duration functions reported in graph B.

I. Right ear: Temporal Masking Curves (TMC) representing masker thresholds as a function of gap duration for probe frequencies (f_p) 500, 1000, 2000, 4000 and 6000 Hz. All TMCs represent the condition where the masker frequency $f_m = f_p$ (on-frequency condition). The linear reference (dotted line) represents the off-frequency condition for f_p = 6000 Hz and f_m = 3300 Hz (0.55 * f_p). Gap durations vary from 0.01 s to 0.09 s in steps of 0.01. Thresholds are generally the mean of 3 measurements. Error bars represent 1 standard error.

J. Right ear: Same as E. for frequencies 250, 500, 1000, 2000, 4000, 6000 and 6600 Hz.

Threshold/duration functions: were measured in the right ear (Figure 8.3-6G) at frequencies 250, 1000, 4000 and 8000 Hz. Tone durations were 0.008, 0.016, 0.032, 0.064, 0.128, 0.256 and 0.512 s. Thresholds are the mean of 3 measurements.

The threshold/duration functions are normal at 250 and 1000 Hz. The functions at 4000 and 8000 Hz are raised and shallow. The mean slope (based on the difference between thresholds for a 500 and 8-ms tone) is 1.8 dB/octave across all test frequencies and 2.4, 2.6, 0.9 and 1.1 respectively at 250, 1000, 4000 and 8000 Hz. The values at 250 and 1000 Hz are very similar to values found in the normal listeners, where as the values for 4000 and 8000 Hz are lower than normal (see Chapter 7).

G and A parameters in the right ear (Figure 7.3-6H) were estimated by fitting the reciprocal function (as described in Chapter 6.4) to the threshold/duration functions pictured in Figure 8.3-6G. Normal *G* and *A* values are found at 250 and 1000 Hz. At 4000 and 8000 Hz, both *G* and *A* values are abnormal with very low *G* estimates and high *A* estimates.

For the left ear, *G* and A parameters were estimated by fitting the reciprocal function to the absolute thresholds for the 500 ms and 8-ms tones only. The parameters are presented in Figure 8.3-6C for the frequencies 250, 500, 1000, 2000, 4000, 6000 and 7000 Hz. No reliable estimates could be obtained for 8000 Hz since the 8 ms and 500 ms-thresholds are very similar and the 8 ms-threshold is slightly higher (0.7 dB) than the 500 ms threshold. A and *G* estimates show a similar pattern to the parameters found in the right ear.

Iso Forward Masking Curves: IFMCs were measured in both ears (Figure 8.3-6E & J). In the right ear, probe frequencies 250, 500, 1000, 2000, 4000 and 6000 Hz were measured. For comparison reasons, IFMCs were determined in the left ear for probe frequencies 500, 1000, 2000 and 4000 Hz. The masker frequencies (fm) are 0.5, 0.7, 0.9, 1, 1.1, 1.3, and 1.6 * f_p. The 4000-Hz IFMC contains extra masker thresholds for masker frequencies 0.1, 0.2, 0.3, 0.4, 1.7, 1.8, 1.9 and 2 * f_p. These extra measurements were aimed at exploring the surrounding frequency regions. For all conditions, the probe level was set at 10 dB SPL above the 8-ms threshold of the probe. Thresholds are generally the mean of 3 measurements.

Both ears show normal IFMCs up to 1000 Hz. From 2000 Hz onwards IFMCs are shallow or show erratic patterns.

In the right ear (Figure 8.3-6J), the 2000-Hz IFMC is flat at the high-frequency tail and only a minor increase in the masker threshold is seen at the low-frequency tail. The lowest masker threshold is situated at probe frequency. The 4000-Hz IFMC shows a discrete V-shape with the lowest masker threshold at 3600 Hz, indicating that a 3600 Hz-masker is most effective at masking a 4000-Hz probe. The extended low-frequency slope follows a normal pattern of rising masker thresholds as masker frequency moves further away from the probe frequency. The high-frequency tail shows a more erratic pattern displaying a second V-shape at a masker frequency of 6400 Hz. The 6000-Hz IFMC also displays a V-shape with a tip frequency of 6600 Hz. Q10 values are respectively 1.17, 1.19, 2.32, 1.52 and 2.43 for frequencies 250, 500, 1000, 4000 and 6000 Hz. No Q10 was calculated for the 2000-Hz IFMC.

In the left ear (Figure 8.3-6E), IFMCs at 500 and 1000 Hz are normal, opposed to an erratically shaped 2000-Hz IFMC and a shallow 4000-Hz IFMC. The high-frequency slope of the 2000-Hz IFMC traces the 4000-Hz IFMC.

Additional IFMC measurements: To further investigate the pattern of the 6000-Hz IFMC in the right ear, an IFMC was measured with a probe frequency of 6600 Hz. Figure 8.3-7 shows a detail

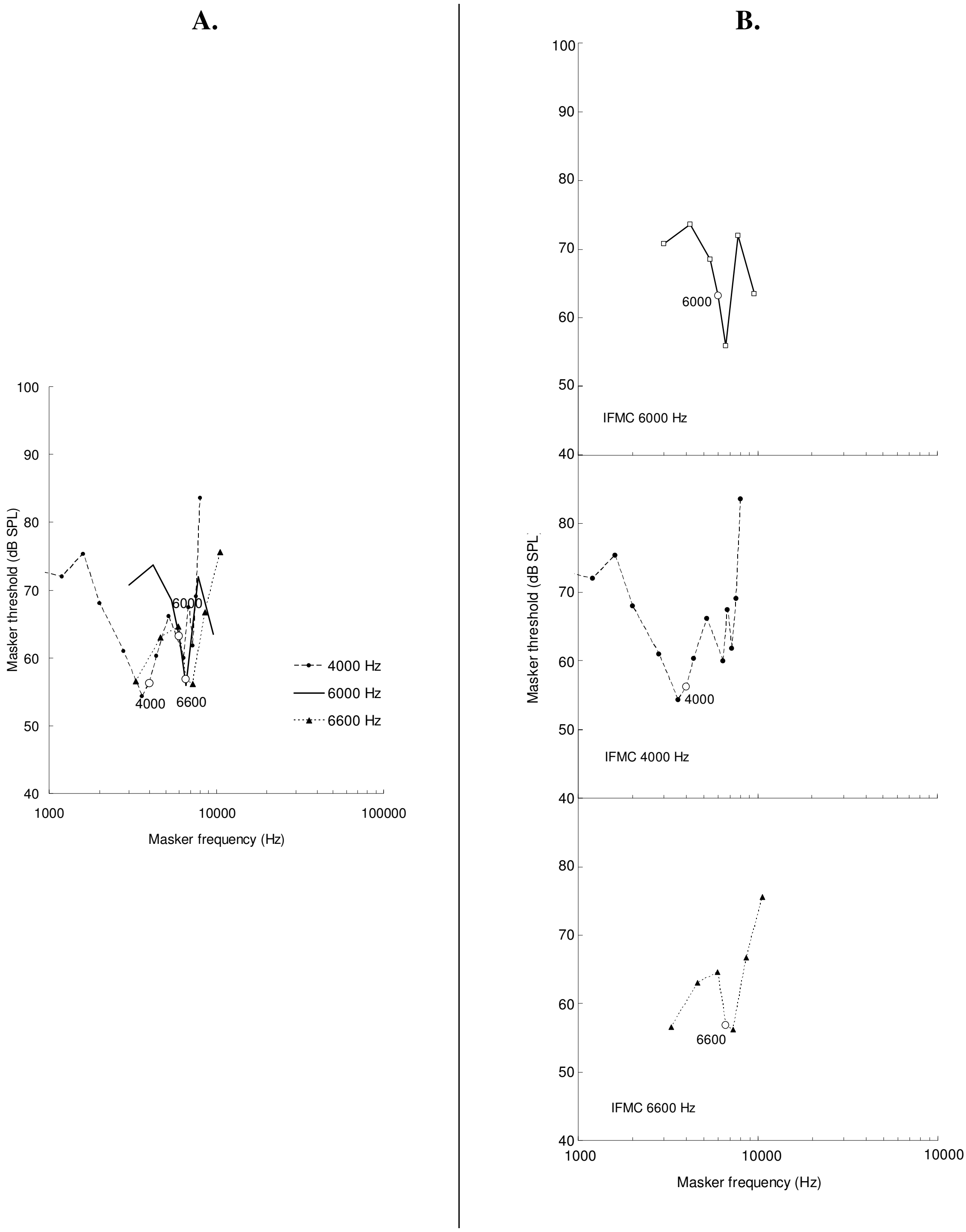


Figure 8.3-7. A. Right ear: IFMCs with probe frequency 4000, 6000 and 6600 Hz. Full thick line represents the 6000 Hz IFMC. The masker threshold at probe frequency is represented by the large open circle. The accompanying value indicates the probe frequency. All but one data point of this IFMC have been omitted for clarity. Dotted line with full circles represents the IFMC with a probe frequency of 4000 Hz. Dotted line with full triangles shows the IFMC with a BF of 6600 Hz. B. Same as A but IFMCs separated per frequency.

of the IFMCs with probe frequency 4000, 6000 and 6600 Hz. The 6600 Hz-IFMC is V-shaped with the lowest masker threshold (tip) at 7260 Hz (1.1 * f_p). The 6000 Hz-IFMC coincides nicely with the 6600 Hz-IFMC suggesting the 6600 Hz-probe and the 6000 Hz-probe are picked up by the same filter (possibly 7260 Hz).

Temporal Masking Curves: TMCs were measured in the right ear (Figure 7.3-6I) for probe frequencies 500, 1000, 2000, 4000 and 6000 Hz. All functions represent the on-frequency condition with masker frequency (f_m) = probe frequency (f_p). The linear reference (dotted line) was found by fitting a straight line to the off-frequency thresholds with a probe frequency of 6000 Hz and a masker frequency (f_m) = 3300 Hz (0.55 * f_p). For all conditions the probe level was set at 10 dB SPL above the 8-ms threshold of the probe. Gap durations varied from 0.01 to 0.09 in steps of 0.01. Thresholds are generally the mean of three measurements. To allow for closer inspection, the TMCs at each frequency are shown separately in Figure 8.3-8.

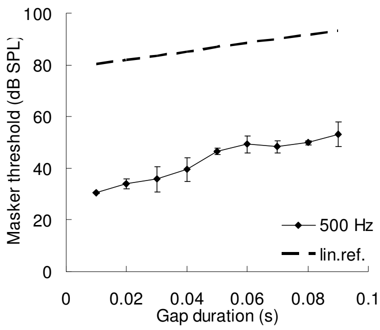
Based on visual inspection, it can be concluded that only the TMC at 1000 Hz shows a normal pattern with a shallow section followed by a steep section. The TMCs for 500, 2000, 4000 and 6000 Hz however do not show the typical steep sloping section and are shallow, suggesting that compression is reduced at these frequencies. This finding is in line with the previous tests suggesting normal hearing up to 1000 Hz and cochlear damage from 1000 Hz onwards.
The compression coefficient and compression range found in the 1000-Hz TMC are in agreement with values found in normal listeners. For the 2 highest frequencies (4000 and 6000 Hz) the coefficients are raised and the compression range is reduced at all frequencies. The TMCs for 500 and 2000 Hz return compression coefficients which are high but could still be classified within the normal range. The compression range however is severely reduced at both frequencies. This is in agreement with the data reported by Plack et al. (2004) where they found

that, in listeners with a mild to moderate hearing loss, the compression ratio did not change significantly compared to normal listeners but the gain at lower input levels was reduced. This would result in a reduced compression range.

250 Hz

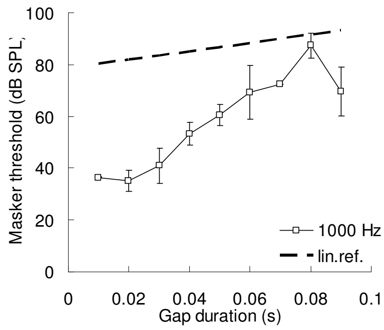
Not measured

500 Hz



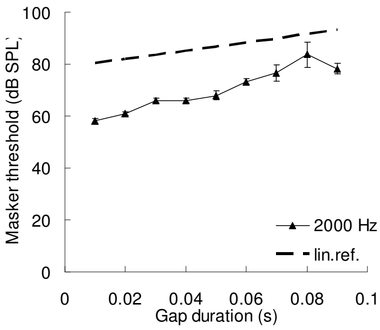
Coefficient	0.38
Range	31-47 dB SPL

1000 Hz



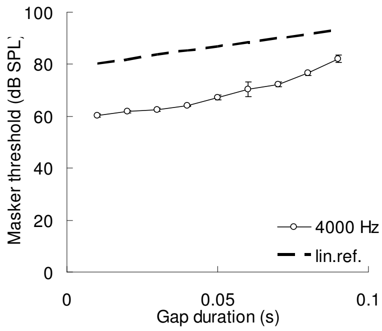
Coefficient	0.18
Range	32-74 dB SPL

2000 Hz



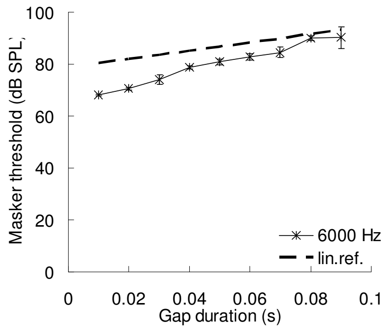
Coefficient	0.30
Range	67-78 dB SPL

4000 Hz



Coefficient	0.45
Range	65-87 dB SPL

6000 Hz



Coefficient	0.55
Range	67-89 dB SPL

Figure 8.3-8. TMCs for listener SFo for probe frequencies (f_p) 500, 1000, 2000, 4000 and 6000 Hz, measured in the right ear. Data points represent masker thresholds for gap durations from 0.01 to 0.09 (in steps of 0.1 s). Masker frequency (f_m) is set equal to the probe frequency (f_p) (on-frequency condition). The linear reference (dotted line) was found by fitting a straight line to the off-frequency data for a probe frequency of 6000 Hz and a masker frequency (f_m) = 3300 Hz (0.55 * f_p). Thresholds are generally the mean of 3 measurements. Error bars represent 1 standard error. Compression coefficient and range for each test frequency are presented below the relevant TMC-graph and were calculated using the method described in chapter 5.5.

Interpretation

A standard clinical screening confirmed that SFo has normal hearing up to 1000 Hz and a moderate loss at the high frequencies. There is no indication of an outer/middle ear problem. DPOAE responses are normal up to 1.4 kHz but absent from 2 kHz onwards. Threshold/duration functions generate abnormal *G* and *A* estimates due to the raised thresholds and reduced threshold/duration effects from 2000 Hz onwards. IFMCs show normal sharpness at the low frequencies but an erratic IFMC at 2000 Hz suggests an unresponsive region at that frequency. At 4000 Hz the IFMC is V-shaped but reduced in sharpness. The tip frequency of this IFMC is situated at 3600 Hz. This is an indication that the 4000-Hz probe is picked up at the 3600-Hz site of the BM. The tip shifts in the 6000-Hz IFMC and the 6600-Hz IFMC suggests that probes at 6000 and 6600 Hz are in fact detected at the 7260-Hz site of the BM. Compression is present up at 1000 Hz but reduced to absent from 2000 Hz onwards. Compression at 500 Hz is also reduced.

It could be hypothesized that OHC damage is the origin of the hearing loss from 2000 Hz onwards, based on the absence of DPOAE responses, reduced to absent tuning and absent compression in this frequency range. An unresponsive region is present at 2000 Hz. Two isolated regions of residual tuning are present at the 3600-Hz and 7260-Hz site of the BM.

8.3.3 Listener BMe

BMe is a 46- year old female with a unilateral hearing loss (right ear). She wears a hearing aid in this ear. The origin of the hearing loss is probably conductive. BMe had an operation 30 years ago to remove cholesteatoma[6]. Her hearing loss was minimal at that time and she did not require a hearing aid. However in recent years, the complaints of hearing loss did become more severe and a hearing aid was prescribed. After the data for this person's profile was collected, BMe was once more diagnosed with cholesteatoma and a new operation has been scheduled.

DPOAE responses have not been measured in the left (normal) ear on request of the participant (for fear of infection).

[6] Cholesteatoma is an accumulation of keratin (skin) that is located within the middle ear of temporal-bone cavity. This accumulation has the potential to become infected and erode or destroy the ossicles or otic-capsule structures (Katz, 2002, p.21)

Audiological assessment

Otoscopy: Normal

Tympanometry: Flat tympanogram in right ear, left ear not tested

Rinne: Negative in right ear, positive in left ear

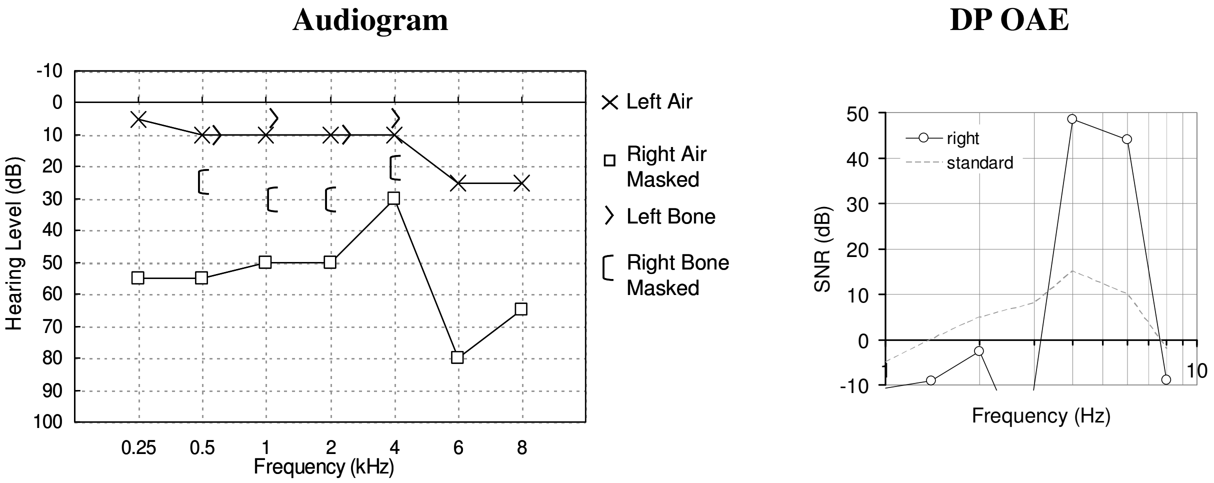


Figure 8.3-9. Audiogram: air conduction thresholds at 0.25, 0.5, 1, 2, 4, 6 and 8 kHz in left and right ear. Bone conduction thresholds at 0.5, 1, 2 and 4 kHz in left and right ear. Thresholds in the right ear are all masked. DP OAE responses for right ear for frequencies 1, 1.4, 2, 2.8, 4, 6 and 8 kHz. No DPOAE responses available in the left ear. Responses are expressed in SNR (dB). The dotted grey line represent the 5th percentile of responses of normal listeners reported by Gorga et al. (1997).

Pure tone audiometry shows normal audiometric thresholds in the left ear with a slight loss at 6 and 8 kHz (Figure 8.3-9). In the right ear a moderate to moderately severe mixed loss is seen. Tympanometry and Rinne for this ear indicate a conductive hearing loss.

Initially BMe reported the nature of her unilateral HL to be conductive. The audiogram for the right ear shows masked air conduction thresholds around 50-55 dB HL for the low frequencies with a peak at 4 kHz (30 dB loss). The high frequencies show a drop in thresholds to 80 and 65 dB HL. Masked bone conduction thresholds show a mild loss of 30 dB. This suggests that the origin of the hearing loss is most probably not limited to a conductive hearing loss. The bone conduction thresholds suggest an additional mild sensorineural problem at the lowest and highest frequencies. DP OAE-responses in the right ear are present at 4 and 6 kHz, as would be expected in a conductive hearing loss, however no responses were recorded at the other frequencies. The presence of DP OAEs indicates that the OHCs are functioning normal at those frequencies. Absent responses could possibly but not necessarily imply OHC damage. For thresholds above 50 dB HL, DP OAEs are most likely to be absent regardless of the underlying pathology (Gorga et al., 1997) so no conclusions can be drawn based on absent responses at frequencies with losses above 50 dB HL.

Psychoacoustic tests

Note: BMe did have problems with wax build-up in the ear. This could have affected threshold estimates at certain times.

Absolute thresholds: Thresholds were measured in both ears for pure tones with tone durations of 500 and 8 ms at frequencies 250, 500, 1000, 2000, 4000 and 8000 Hz (Fig. 8.3-10A and B). Thresholds are based on a single measurement.

In the left ear, thresholds for long tones are within normal limits. At 8000 Hz the threshold is raised. The difference between the thresholds for a 500 ms and a 8-ms tone is large up to 2000 Hz but slightly reduced for 4000 and 8000 Hz.

The right ear has raised thresholds at all frequencies with the highest thresholds at the low frequencies and at 8000 Hz. Thresholds are lowest at 2000 and 4000 Hz. The high thresholds at low frequencies are in line with a conductive hearing loss where the lower frequencies are expected to be affected most severely. The difference between the thresholds for the long and short tone is reduced at 4000 and 8000 Hz.

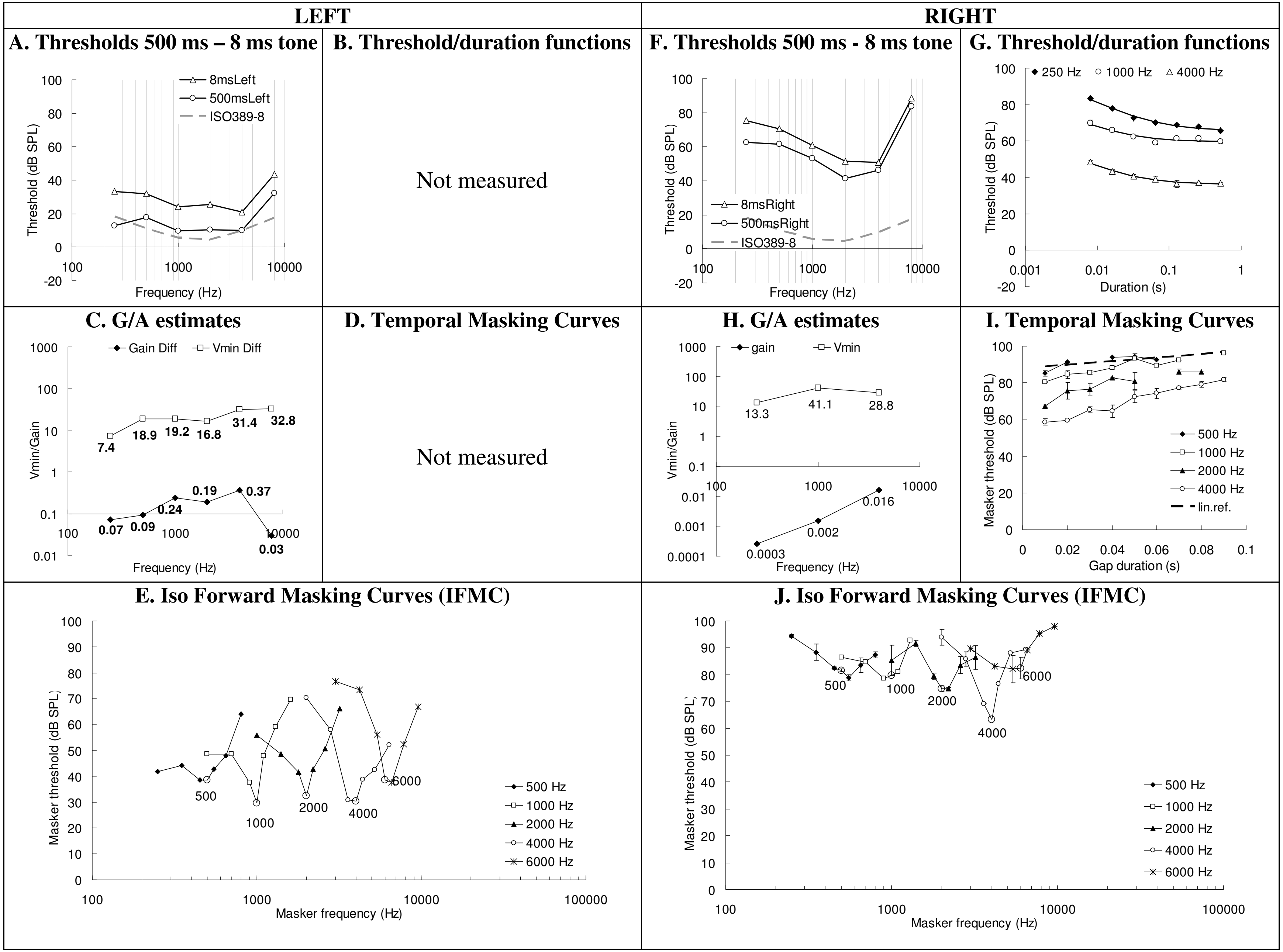


Figure 8.3-10: Impaired listener BMe.
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Figure 8.3-10. Psychoacoustic profile of impaired listener BMe.

A. Left ear: Absolute thresholds as a function of frequency for 500-ms tones (open circles) and 8-ms tones (open triangles) at frequencies 250, 500, 1000, 2000, 4000 and 8000 Hz. Thresholds are based on a single measurement.

C. Left ear: estimates of parameters *G* (filled diamonds) and *A* (open squares) estimates based only on the difference in absolute threshold between the 8 ms and 500-ms tones reported in graph A.

E. Left ear: Iso Forward Masking Curves (IFMC) for probe frequencies (f_p) 500, 1000, 2000, 4000 and 6000 Hz. The masker frequencies (f_m) are 0.5, 0.7, 0.9, 1, 1.1, 1.3, and 1.6 * f_p. The masker threshold at probe frequency is represented by the large open circle. The accompanying value indicates the probe frequency. Thresholds are based on a single measurement. Error bars represent 1 standard error.

F. Right ear: Same as A.

G. Right ear: Thresholds as a function of tone duration at frequencies 250, 1000 and 4000 Hz. Each threshold is the mean of 3 measurements. Error bars represent 1 standard error. The full lines represent predictions made using the reciprocal function described in section 6.4. Each fitted function provides an estimate of parameter *A* and *G* presented in Graph C.

H. Right ear: *G* (filled diamonds) and *A* (open squares) estimates at frequencies 250, 1000 and 4000 Hz, associated with the predictive functions (full lines in graph B) to the threshold/duration functions reported in graph B.

I. Right ear: Temporal Masking Curves (TMC) representing masker thresholds as a function of gap duration for probe frequencies (f_p) 500, 1000, 2000 and 4000 Hz. All TMCs represent the condition where the masker frequency f_m = f_p (on-frequency condition). The linear reference (dotted line) represents the off-frequency condition for f_p = 3000 Hz and f_m = 1650 Hz (0.55 * f_p). Gap durations vary from 0.01 s to 0.09 s in steps of 0.01. Thresholds are generally the mean of 3 measurements. Error bars represent 1 standard error.

J. Right ear: Same as E. for frequencies 500, 1000, 2000, 4000 and 6000 Hz. Thresholds are based on 3 measurements.

Threshold/duration functions: were measured in the right ear (Figure 8.3-10G) at frequencies 250, 1000 and 4000 Hz. The 8000 Hz condition could not be measured at that particular time due to thresholds exceeding clipping threshold. Tone durations were 0.008, 0.016, 0.032, 0.064, 0.128, 0.256 and 0.512 s. Thresholds are the mean of three measurements.

The threshold/duration functions are raised at all frequencies and steep for 250 and 4000 Hz. The 1000- Hz-function is shallower than expected. The mean slope (based on the difference between thresholds for the 0.512 and 0.008 s thresholds) is 2.2 dB/octave across all test frequencies and 3.0, 1.7 and 2.0 respectively at 250, 1000 an 4000 Hz. These values are similar to the values found in the normal listeners. The value for 1000 Hz is lower than normal.

G and *A* parameters in the right ear (Fig. 8.3-10H) were estimated by fitting the reciprocal function (as described in Chapter 6.4) to the threshold/duration functions in Figure 8.3-10G. At all frequencies the *G* values are abnormal; this reflects the raised absolute thresholds. *A* estimates are normal at all frequencies. At 1000 Hz the *A* estimate is high but still within normal limits. This implies that all 3 frequencies are situated in Quadrant II (see section 6.4), which corresponds with a conductive hearing loss.

G and *A* parameters in the left ear (Fig. 8.3-10C) were estimated by fitting the reciprocal function only to the absolute thresholds for the 500ms and the 8-ms tones presented in Figure 8.3-11A. The estimates of the parameters for all frequencies are within normal range except at 8000 Hz. At this frequency, an abnormal *G* parameter is found which reflects the raised threshold at that frequency. The *A* parameter however is normal which situates the *G/A* combination at 8000 Hz in Quadrant II.

Iso Forward Masking Curves: IFMCs were measured in both ears (Fig. 8.3-10E and J). In the right ear, probe frequencies 500, 1000, 2000, 4000 and 6000 Hz were measured. For comparison reasons, IFMCs were determined in the left ear for probe frequencies 500, 1000, 2000, 4000 and 6000 Hz. The masker frequencies (f_m) are 0.5, 0.7, 0.9, 1, 1.1, 1.3, and 1.6 * f_p. For all conditions, the probe level was set at 10 dB SPL above the 8-ms threshold of the probe. Thresholds are generally the mean of 3 measurements.

IFMCs in the normal (left) ear (Fig. 8.3-10E) are normal V-shaped curves. Q10 values are respectively 1.7, 3.5, 2.7, 2.5 and 3.5 for frequencies 500, 1000, 2000, 4000 and 6000 Hz. These values are consistent with the Q10 values found in our normal listeners (Chapter 7).

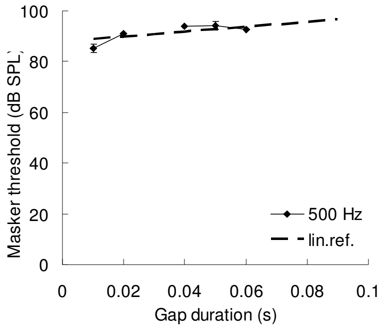
The IFMCs in the impaired (right) ear (Fig. 8.3-10J) show raised and shallower curves at all frequencies except at 4000 Hz. At that frequency the IFMC is raised but the sharpness is normal. Q10 values are 0.94, 1.24, 1.53, 4.41, 1.31 for frequencies 500, 1000, 2000, 4000 and 6000 Hz respectively. The Q10 value at 4000 Hz is actually higher than the values found in our normal listeners. The remaining IFMCs exhibit Q10 values below normal values, representing the shallower-than-normal functions.

Temporal Masking Curves: TMCs were measured in the right ear (Figure 8.3-10I) for probe frequencies 500, 1000, 2000 and 4000 Hz. All functions represent the on-frequency condition with the masker frequency (f_m) = probe frequency (f_p). No masker thresholds could be determined for the on-frequency condition at 6000 Hz due to thresholds exceeding clipping threshold. The linear reference (dotted line) was found by fitting a straight line to the off-frequency thresholds with a probe frequency of 3000 Hz and a masker frequency (f_m) = 1650 Hz (0.55 * f_p). The off-frequency condition could not be measured at 4000 Hz. Therefore the off-frequency for a 3000 Hz-probe was used to obtain the linear reference. For all conditions the probe level was set at 10 dB SPL above the 8-ms threshold of the probe. Gap durations varied from 0.01 to 0.09 in steps of 0.01. Thresholds are generally the mean of 3 measurements. To allow for closer inspection, the TMCs at each frequency are shown separately in Figure 8.3-11.

250 Hz

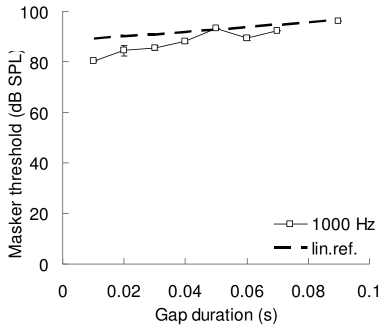
Not measured

500 Hz



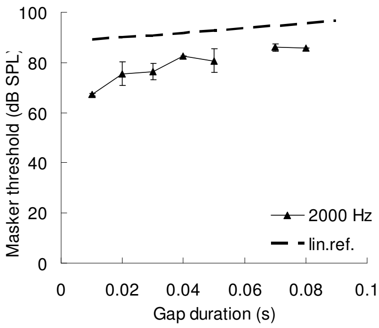
Coefficient	0.17
Range	85-91 dB SPL

1000 Hz



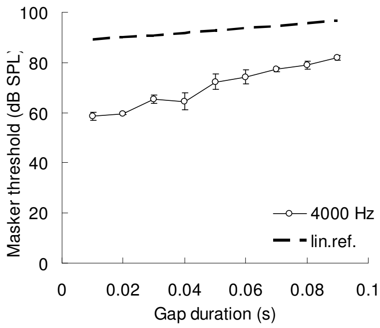
Coefficient	0.43
Range	79-91 dB SPL

2000 Hz



Coefficient	0.27
Range	64-82 dB SPL

4000 Hz



Coefficient	0.12
Range	64-76 dB SPL

6000 Hz

Not measured

Figure 8.3-11. TMCs for listener BMe for probe frequencies (f_p) 250, 500, 1000, 2000 and 4000 Hz, measured in the right ear. Data points represent masker thresholds for gap durations from 0.01 to 0.09 (in steps of 0.1 s). Masker frequency (f_m) is set equal to the probe frequency (f_p) (on-frequency condition). The linear reference (dotted line) was found by fitting a straight line to the off-frequency data for a probe frequency of 3000 Hz and a masker frequency (f_m) = 1650 Hz (0.55 * f_p). Thresholds are generally the mean of 3 measurements. Error bars represent 1 standard error. Compression coefficient and range for each test frequency are presented below the relevant TMC-graph and were calculated using the method described in chapter 5.5.

Upon visual inspection, little indication of compression could be found at the test frequencies. The TMC at 4000 Hz shows a shallow section followed by a short steeper section which in turn progressed in a second shallower section. Minor steep sections can also be seen at most of the other frequencies. Based on this detailed analysis, the compression coefficients established for the 4 test frequencies are within normal limits except at 1000 Hz. Normal coefficients could be obtained because at all frequencies a steep section was present in the TMC. The steep section was usually very small which is represented in the seriously reduced compression ranges at all frequencies. It can be argued that a straight line would also fit the data-points well. This would result in compression coefficients which are not within normal limits and indicate limited compression at these frequencies.

Interpretation

All measures in the clinical screening suggest that the main component in BMe's hearing problem in the right ear is of conductive origin. The tympanogram is flat, the Rinne is negative and there is a gap between the air conduction and the bone conduction thresholds. However, the bone conduction thresholds are slightly raised at the lower frequencies which could indicate an additional sensorineural component.

The psychoacoustic measure showed that the threshold/duration functions are raised but steep as expected in a conductive hearing loss except at 1000 Hz where a shallower function was found. IFMCs are shallower than normal except at 4000 Hz. Compression is limited to a narrow range of intensity levels. The TMCs appear very shallow due to the limited intensity range the participant is operating in but some compression can be seen at most frequencies. At 1000 Hz, the compression coefficient is higher than normal.

The normal IFMC at 4000 Hz agrees with the steep threshold/duration function and the strong DPOAE responses at this frequency and supports the hypothesis that the hearing loss might be purely conductive at 4000 Hz. At the remaining frequencies, there is indication that a conductive loss is combined with some OHC damage.

It needs to be noted that the masker thresholds in both the TMCs and the IFMCs were very high, which made it difficult to obtain thresholds in all conditions. It is possible that these high thresholds actually induce a ceiling effect making IFMCs shallower and TMCs less steep/almost linear.

It is hypothesized that a conductive loss is present at all frequencies in the right ear. This is supported by the presence of an airbone-gap in the audiogram, a flat tympanogram, a negative Rinne and strong DP OAE response at 4000 and 6000 Hz. Furthermore, steep threshold/duration functions were found at 250 and 4000 Hz and a sharp IFMC was observed at 4000 Hz.
However an OHC problem is also contributing to the hearing loss at most frequencies except at 4000 Hz. This is supported by the presence of slightly raised bone conduction thresholds at 500, 1000 and 2000 Hz in the audiogram, a shallow threshold/duration function at 1000 Hz and shallow IFMCs at 500, 1000, 2000 and 6000 Hz.

8.3.4 Listener MFr

MFr is a 44 years old hearing impaired male. He wears hearing aids bilaterally. The cause of his hearing impairment is linked to Alport's syndrome[7].

[7] Alport's disease is a hereditary nephropathy with an accompanying bilateral, progressive, sensorineural hearing loss. The basis of this syndrome lies in a dysfunction in the formation of a protein called Type IV collagen. This protein is involved in the formation of the renal, ocular and cochlear basement membranes. The hearing loss is not congenital and is usually detected by late childhood or early adolescence in boys. In females hearing loss is less frequent and tends to occur later in life. In the early stages the hearing deficit emerges in a bilateral reduction in sensitivity to tones in the 2000 – 8000 Hz range. The loss progresses and eventually extends to other frequencies, resulting in a flat audiogram (Kashtan, 1999). Clinical observations on SNHL and Alport syndrome generally show excellent speech discrimination and a hearing loss not more that 60 to 70 dB (Rintelmann, 1976, Gleeson, 1984). The exact effect of the Type IV collagen defect in the cochlea is still not very clear but a histological study showed a separation between the basilar membrane and the organ of Corti and cellular in-filling of the tunnel and extracellular spaces of the organ of Corti (Zehnder et al., 2005). It is thought that this will affect cochlear micromechanics. Possibly an inadequate adhesion between the cells of the organ of Corti and the basilar membrane may result in inadequate tuning by the outer hair cells. Other studies reported abnormalities of the stria vascularis, organ of Corti and hair cell loss in a number of patients (Johnsson and Arenberg, 1981, Westergaard et al., 1972).

Audiological assessment:

Otoscopy: Normal

Tympanometry: Normal

Rinne: Positive

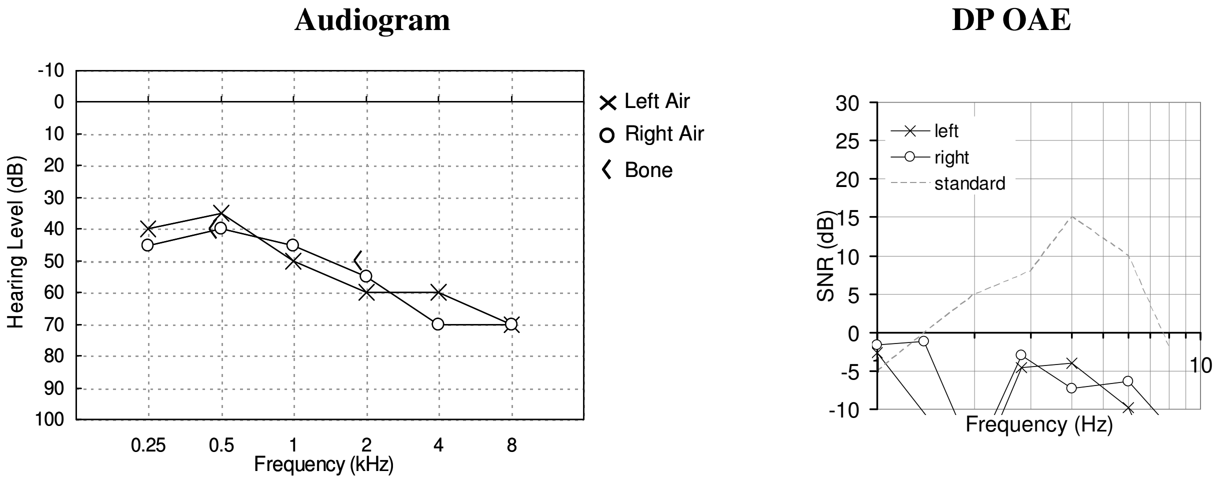


Figure 8.3-12. Audiogram: air conduction thresholds at 0.25, 0.5, 1, 2, 4 and 8 kHz for left and right ear. Bone conduction thresholds at 0.5, 1, 2 and 4 kHz. DP OAE responses for frequencies 1, 1.4, 2, 2.8, 4, 6 and 8 kHz for left and right ear. Responses are expressed in SNR (dB). The dotted grey line represent the 5th percentile of responses of normal listeners reported by Gorga et al. (1997).

Pure tone audiometry shows a moderate to moderately severe sensorineural hearing loss (Figure 8.3-12). There is no indication of an outer/middle ear problem. No DP OAE's could be recorded in left or right ear.

Psychoacoustic tests

Absolute thresholds: Thresholds were measured in both ears for pure tones with tone durations of 500 ms and 8 ms at frequencies 250, 500, 1000, 2000, 4000, 6000 and 8000 Hz (Figure 8.3-13A & F). Thresholds are based on a single measurement.

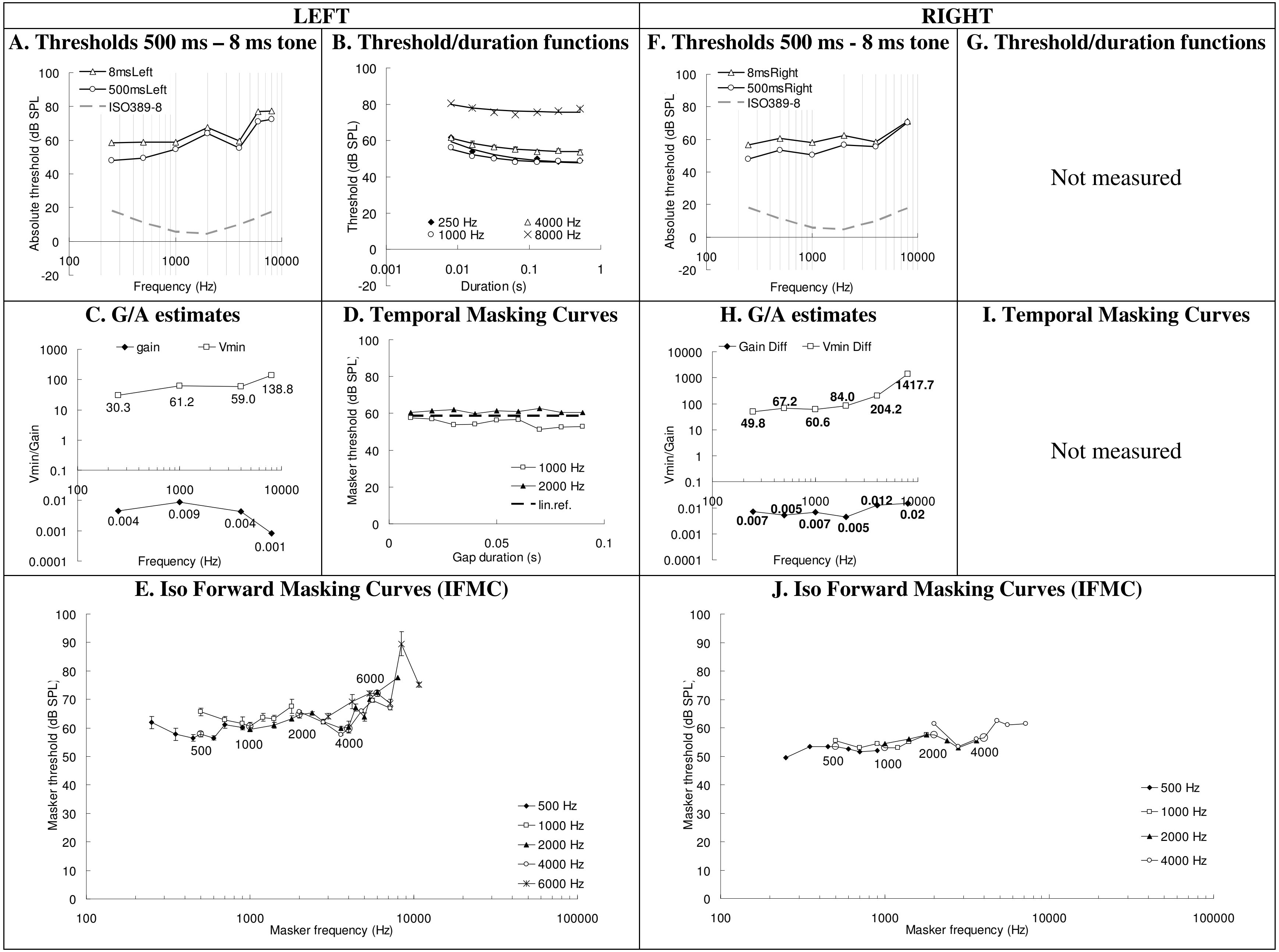


Figure 8.3-13: Impaired listener MFr.
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Figure 8.3-13. Psychoacoustic profile of impaired listener MFr.

A. Left ear: Absolute thresholds as a function of frequency for 500-ms tones (open circles) and 8-ms tones (open triangles) at frequencies 250, 500, 1000, 2000, 4000, 6000 and 8000 Hz. Thresholds are based on a single measurement.

B. Left ear: Thresholds as a function of tone duration at frequencies 250, 1000, 4000, and 8000 Hz. Each threshold is the mean of 3 measurements. Error bars represent 1 standard error. The full lines represent predictions made using the reciprocal function described in section 6.4. Each fitted function provides an estimate of parameter *A* and *G* presented in Graph C.

C. Left ear: *G* (filled diamonds) and *A* (open squares) estimates at frequencies 250, 1000, 4000 and 8000 Hz, associated with the predictive functions (full lines in graph B) to the threshold/duration data reported in graph B.

D. Left ear: Temporal Masking Curves (TMC) representing masker thresholds as a function of gap duration for probe frequencies (f_p) 1000 and 2000Hz. All TMCs represent the condition where the masker frequency $f_m = f_p$ (on-frequency condition). The linear reference (dotted line) represents the off-frequency condition for $f_p = 1000$ Hz and $f_m = 550$ Hz ($0.55 * f_p$). Gap durations vary from 0.01 s to 0.09 s in steps of 0.01. Thresholds are based on 1 measurements. Error bars represent 1 standard error.

E. Left ear: Iso Forward Masking Curves (IFMC) for probe frequencies (f_p) 500, 1000, 2000, 4000 and 6000 Hz. The masker frequencies (f_m) are 0.5, 0.7, 0.9, 1, 1.2, 1.4, and 1.8 * f_p (Note: these ratios differ slightly from the ratios described in the Method Section). IFMC with probe frequency 2000 Hz shows extra masker thresholds for masker frequencies 2, 2.2, 2.5, 2.7, 3, and 4* f_p. The masker threshold at probe frequency is represented by the large open circle. The accompanying value indicates the probe frequency. Thresholds are generally the mean of 3 measurements. Error bars represent 1 standard error.

F. Right ear: same as A for frequencies 250, 500, 1000, 2000, 4000 and 8000 Hz.

H. Right ear: *G* (filled diamonds) and *A* (open squares) estimates based only on the difference in absolute threshold between the 8 ms and 500-ms tones reported in graph F.

J. Right ear: same as E. for frequencies 500, 1000, 2000 and 4000 Hz. Thresholds are result of single measurement.

Both the left and the right ear show raised thresholds with an increasing loss at the high frequencies. The functions for long and short tones are parallel and show a small dip in threshold at 4000 Hz. The threshold difference between the 500-ms and the 8-ms tones is small at all frequencies.

Threshold/duration functions: were measured in the left ear (Figure 8.3-13B) at frequencies 250, 1000, 4000 and 8000 Hz. Tone durations were 0.008, 0.016, 0.032, 0.064, 0.128, 0.256 and 0.512 s. Thresholds are the mean of three measurements.

The threshold/duration functions are shallow with a mean slope (based on the difference between the 0.512 s and 0.008 s-threshold) of 1.3 dB/octave across all test frequencies and 2.1, 1.3, 1.4 and 0.5 respectively at 250, 1000, 4000 and 8000 Hz. The slope at 250 Hz is similar to values found in normal listeners but the slopes for the three other frequencies are lower than normal.

G and *A* parameters in the left ear (Figure 8.3-13C) were estimated by fitting the reciprocal function (as described in section 6.4) to the threshold/duration functions in Figure 8.3-13B. All frequencies show an abnormal (reduced) *G* which represents the raised thresholds at these frequencies. The *A* value at 250 Hz is within normal limits but *A* estimates are higher than normal for 1000, 4000 and 8000 Hz. These 3 frequencies can be situated in Quadrant III (Chapter 5.4).

For the right ear, *G* and A parameters were obtained (Figure 8.3-13H) by fitting the same reciprocal function only to the absolute thresholds of the 500-ms and 8-ms tones at different frequencies presented in Figure 8.3-13F. *G* and *A* parameters are abnormal at all frequencies situating all test frequencies in Quadrant III.

Iso Forward Masking Curves: IFMCs were measured in both ears (Fig. 8.3-13E & J). In the left ear, probe frequencies 500, 1000, 2000, 4000 and 6000 Hz were measured. IFMCs in the right ear were determined for probe frequencies 500, 1000, 2000 and 4000 Hz. The masker frequencies (f_m) are set at 0.5, 0.7, 0.9, 1, 1.2, 1.4 and 1.8 * f_p[8]. For all conditions, the probe level was set at 10 dB SPL above the 8-ms threshold of the probe. Masker thresholds measured in the left ear are the mean of three measurements and based on a single measurement in the right ear.

The IFMCs generally show shallow to flat curves in both left and right ear, indicating that a masker is equally efficient at masking the probe no matter what the masker frequency is. For the left ear, discrete V-shapes can be seen at 500, 1000 and 4000. Q10 values for these 3 frequencies are respectively 0.75, 0.64 and 1.13. The 2000-Hz IFMC shows a W-shape with its low-frequency slope tracing the high-frequency slope of the 1000-Hz IFMC. Its high-frequency slope traces the low-frequency slope of the 4000 Hz-IFMC. The 6000 Hz-IFMC also shows this W-shape with its low-frequency slope following the pattern of the high-frequency slope of the 4000-Hz IFMC. In the right ear, flat IFMCs are found for 500 and 1000 Hz. At 2000 and 4000 Hz, a pattern is found similar to the ones in the left ear.

To further explore the high-frequency tail of the 2000-Hz IFMC in the left ear, additional masker thresholds at higher frequencies were measured in the left ear (Figure 8.3-14). The additional data-points represent masker thresholds for masker frequencies 4000, 4400, 5000, 5400, 6000 and 8000 Hz (2, 2.2, 2.5, 2.7, 3 and 4 * f_p)

[8] These masker frequencies are slightly different from the standard masker frequencies. The IFMCs for participant MFr were collected in the early stages of the research project, after which small changes to the measurement conditions were implemented. These differences however do not affect the overall conclusion on the frequency selectivity in this impaired listener.

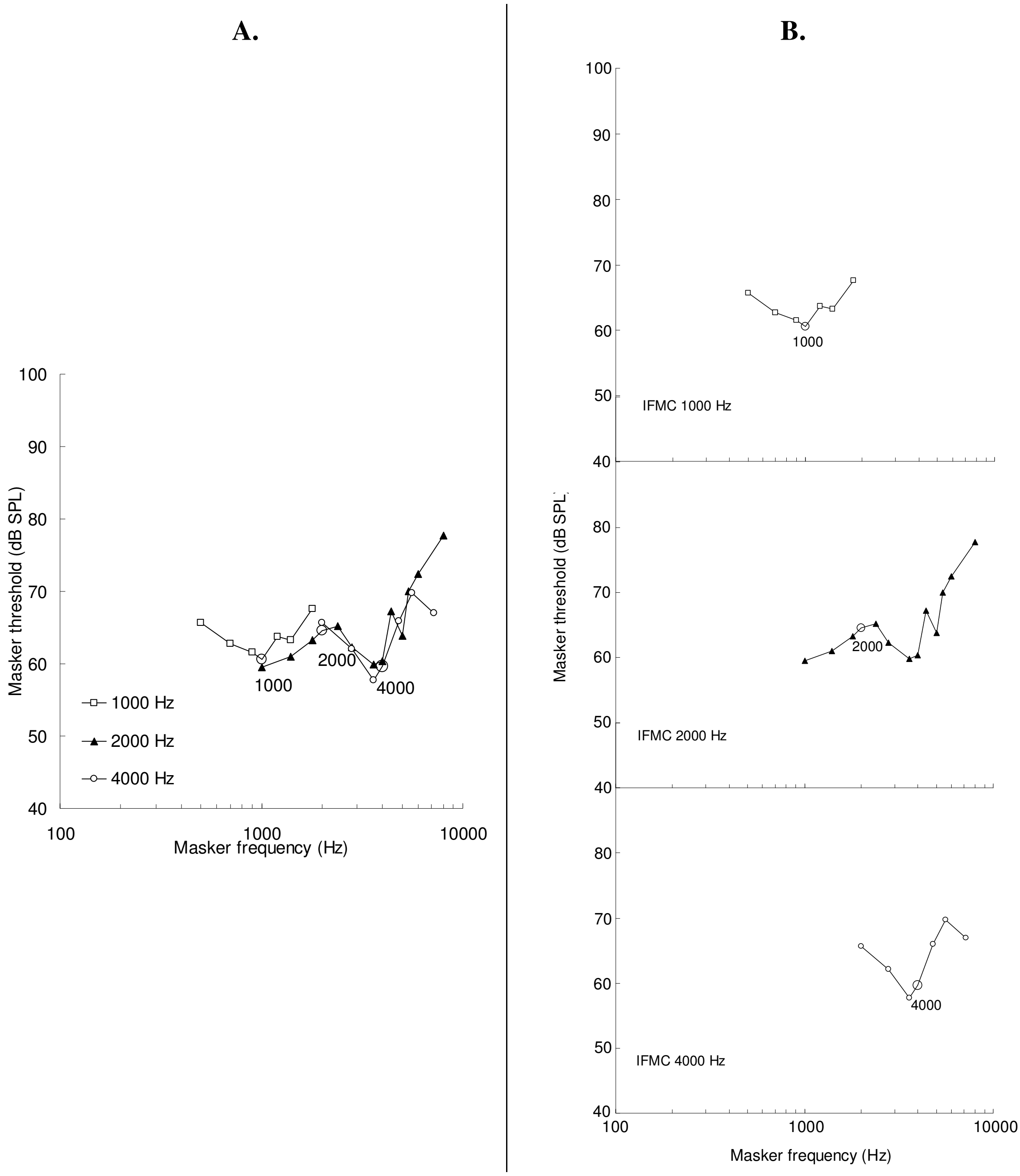

Figure 8.3-14. A. Left ear: IFMCs with probe frequency 1000, 2000 and 4000 Hz. The masker frequencies (f_m) are 0.5, 0.7, 0.9, 1, 1.2, 1.4, and 1.8 * f_p. IFMC with probe frequency 2000 Hz shows extra masker thresholds for masker frequencies 2, 2.2, 2.5, 2.7, 3 and 4* f_p. The masker threshold at probe frequency is represented by the large open circle. The accompanying value indicates the probe frequency.

The high-frequency slope of the 2000-Hz IFMC follows the pattern of the 4000-Hz IFMC closely, indicating that the 2000-Hz probe is detected at the same place on the BM as the 4000 Hz-probe. Similarly, the 6000-Hz IFMC (8.3-12E) shows that a 6000-Hz probe is also detected at that same site on the BM. However, this is not probably not the 4000-Hz site of the BM. A tip shift observed in the 4000-Hz IFMC indicates that the 3600-Hz site of the BM is most effective in detecting the 4000-Hz probe. In the right ear, a tip shift in the 4000-Hz IFMC is also observed situated at 2800 Hz.

Temporal Masking Curves: TMCs were measured in the left ear (Figure 8.3-13D) for probe frequencies 1000 and 2000 Hz. Both functions present the on-frequency condition with masker frequency (f_m) = probe frequency (f_p). The linear reference (dotted line) was found by fitting a straight line to the off-frequency thresholds with a probe frequency of 1000 Hz and a masker frequency (f_m) = 0.55 * probe frequency (f_p). For all conditions the probe level was set at 10 dB SPL above the 8-ms threshold of the probe. Gap durations varied from 0.01 to 0.09 in steps of 0.01. Thresholds are based on a single measurement.

There is no indication of compression at any of the test frequencies. The TMCs for 1000 and 2000 Hz are flat with a slope identical to the linear reference. In fact there was no indication that any recovery of forward masking was taking place. Masker thresholds, both in the on and off-frequency condition, did not increase with increasing gap duration. This effect should be observed even when the basilar membrane is completely linear.

Additional TMC measurement:

To further explore the apparent absence of recovery of forward masking, we measured the on-frequency condition for a probe of 6000 Hz and extended the range of gap durations up to 0.2 s. Gap durations were 0.01, 0.05, 0.06, 0.09, 0.13 and 0.16 s. As before, the probe level was set at 10 dB SL. Thresholds are based on a single measurement. Figure 8.3-15 shows the 6000 Hz-TMC for long gap durations. To enable comparison the TMCs for probe frequencies of 1000 and 2000 Hz and the linear reference presented earlier are included in the graph.

The TMC for a probe frequency of 6000 Hz does not show any reliable sign of an increase in masker threshold as a function of gap duration. The thresholds vary around 80 dB SPL except for the gap of 0.16 s where a distinct increase in masker threshold is observed.

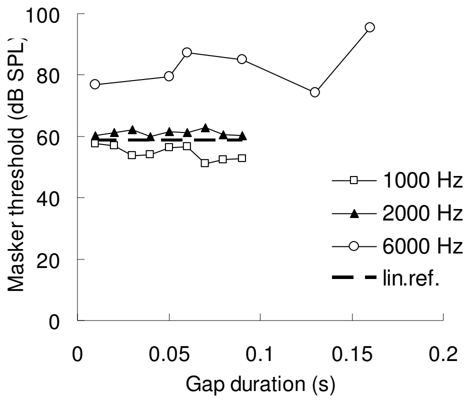

Figure 8.3-15. TMC for the on-frequency condition at 6000 Hz with longer gap durations (0.01, 0.05, 0.06, 0.09, 0.13, 0.16 s). TMCs for 1000 and 2000 Hz and linear reference are the same as presented in Fig. 8.3-12D.

The failure to observe recovery of forward masking, even at long gap durations, is very interesting. The underlying mechanism to this phenomenon is not clear.

The absence of a recovery function may mean that the IHC/AN synapse is not driven above spontaneous activity. However if this were the case one would expect to see random responses and thresholds in all measures. This is clearly not the case since MFr responds in a very consistent manner in all threshold measurements. A second explanation is that the IHC/AN synapse is permanently overactive, preventing any recovery.

Speech recognition: As part of a pilot study looking into measures for the recognition of speech in a noisy environment, a speech recognition test was performed. The data obtained from this measure has not been implemented in the hearing profile but will be presented for informative purposes.

Method: For speech recognition-testing, the Boothroyd AB-wordlists were employed (Boothroyd, 1968). The listener was presented with lists of Consonant-Vowel-Consonant (CVC) words. Each list consists of 10 words. The listener is asked to verbally repeat each presented word. Each list is scored as phonemes correct out of 30. This score is then converted into a percentage of correct phonemes. One list was presented per condition. Stimuli and noise were presented to both ears via circumaural headphones. The stimuli were recorded in the lab by a non-professional speaker. The noise used was white noise.

Results: In the first part of the experiment a speech audiogram in quiet was measured. Speech recognition (in % correct) was established for different speech levels (70, 80, 85, 90 dB SPL). One list was presented per condition. Figure 8.3-16A shows the speech performance as a function of speech level. Speech performance improves as speech level goes up. Optimal performance was found for a speech level of 85 dB SPL, obtaining more than 70% correct identification of CVC-words. From 85 dB SPL onwards however, increasing the level of the speech does not offer any additional benefit and appears to cause a worsening in performance.

In the second part of the experiment, noise was added to the speech. Again speech recognition (in phonemes correct) was evaluated. The speech level was set at 85 dB SPL, which was the level of maximal performance found in the first part of the experiment. Five different SNR-levels were tested (25, 15, 5, 0, -5). The SNR-level is defined as the difference in intensity level between the stimulus and the noise. A positive SNR represents a stimulus which is louder than the noise. A negative SNR means that the noise is louder than the stimulus.

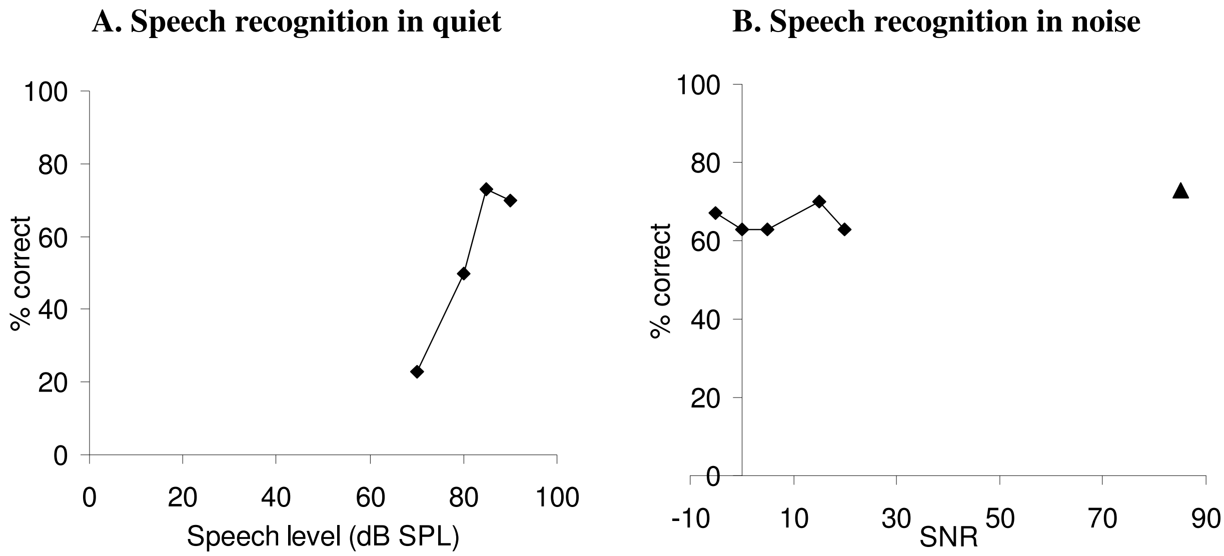

Figure 8.3-16. A. Speech recognition (in % phonemes correct) as a function of speech level (in dB SPL). Different speech levels are 70, 80, 85 and 90 dB SPL. B. Speech recognition scores (in % phonemes correct) as a function of Signal to Noise ratio (SNR). The speech level is set at 85 dB SPL. SNRs are 25, 15, 5, 0 and -5. The large triangle represents performance in quiet (as established in A).

Figure 8.3-16B shows the percentage correct scores as a function of SNR. When introducing a small amount of noise, performance drops slightly compared to performance in quiet. However increasing the level of the noise does not cause any further decrease in performance. The percentages correct vary between 70 and 63 %. Although his scores do not seem to be affected by increasing the noise level, the participant did indicate that he found the task increasingly difficult with increasing noise levels.

This result is very unusual. It can be hypothesized that, in the absence of frequency selectivity, the speech is actually masking itself. Therefore, adding noise does not result in a decrease of performance since maximal masking is already occurring.

Interpretation

The results of the clinical screening indicate that MFr has a moderate to moderately severe sensorineural hearing loss. There is no indication of an outer/middle ear problem. DPOAE responses are absent across all frequencies. The hearing loss is associated in a larger syndrome called Alport's disease.

The psychoacoustic measures showed raised thresholds, shallow threshold/duration functions, shallow to flat IFMCs in both ears and no sign of compression. Some tuning is possibly conserved at the 3600 Hz site of the BM in the left ear and at the 2800 Hz site in the right ear.

Based on these observations, it can be hypothesized that OHC damage is the origin of the impairment. However, although OHC damage seems the most straightforward interpretation of this patient's hearing profile, the apparent absence of recovery of forward masking does raise the question if other structures or mechanisms in the inner ear are affected as well, such as the IHC/AN synapse.

The protein deficiency at the basis of Alport's syndrome is usually associated with defects in the the OHC-function. A histological study showed a separation between the basilar membrane and the organ of Corti and cellular in-filling of the tunnel and extracellular spaces of the organ of Corti (Zehnder et al., 2005). Other studies reported abnormalities of the stria vascularis, organ of Corti and hair cell loss in a number of patients (Johnsson and Arenberg, 1981, Westergaard et al., 1972). All these abnormalities can indeed be associated with deficiencies in the functioning of the OHC. However abnormalities in the stria vascularis would also affect the functioning of the IHCs and could be a contributing factor to the absence of forward masking observed in listener MFr.

8.4 Conclusion

This chapter is concerned with producing hearing profiles of impaired listeners. Four impaired listeners were tested extensively and detailed information on their hearing was collected.

The first aim of testing impaired listeners was to check whether the testing protocol was suitable for use with impaired listeners. Although normal listeners did not manifest any problems when conducting the different tests, there was the possibility that impaired listeners would not be able to perform the tests. However, this was not the case. The data collection occurred without any problems. The hearing impaired listeners were comfortable in conducting the tests. The time needed to collect all the data comprised in a hearing profile was very similar to the time taken by the normal listeners. No extra measurements were required for the impaired listeners; they performed the tasks in the same way as the normal listeners. The profiles were also collected quicker because in most cases, not as many frequencies could be tested compared to the normal listeners.

The second and most important goal was to collect detailed information on each impaired listener and evaluate the benefits of the hearing profile in diagnosing the origin of the impairment. The hearing profiles revealed some very interesting features. It showed in the first place that most hearing impaired listeners are definitely different. It is no surprise that a conductive hearing loss will generate a different profile than a sensorineural hearing loss. However, the profiles presented in this chapter show that the classification of ‘sensorineural loss’ covers a broad spectrum and that listeners with a similar audiogram can generate profiles with distinct differences. Nevertheless, in a clinical setting, these two people would probably be treated in a very similar way.

A second asset of the hearing profiles is the fact that detailed information is available for each frequency in one single listener. This information is available over and above the information found in the audiogram. This can potentially enable a clinician to remediate the hearing loss in a frequency-specific manner.

The measures used have been shown to differentiate between various impairments:

- In the profiles of the impaired listeners, parameter *G/A* combinations situated in Quadrants I to III, as presented in Chapter 6.4, have been reported. The observations found in Quadrant I were associated with frequencies where normal thresholds were found. Interestingly, frequencies associated with hearing loss returned *G/A* combinations situated both in Quadrant II and III. As expected, the observations where a conductive loss was found were situated in Quadrant II (listener BMe). However, this quadrant also contained parameter combinations associated with sensorineural loss (listener JEv). More specifically, it contained observations associated with frequencies where an IHC problem was hypothesized. Observations situated in Quadrant III were thought to be associated with OHC damage (listener SFo and MFr). Further exploration of the observations situated in Quadrant II and III is required but the preliminary results show potential for the *G* and *A* parameters to be of use in differentiating between hearing impairments.
- The IFMCs do provide a clear picture of the frequency selectivity in a listener. At each frequency, one can establish whether the IFMCs are normal, shallower or if there is an indication of an un- or less responsive region. In the latter case the IFMC offers insight in which BM site is most sensitive and where a more favourable filter is at work.
- For the compression measure, we have been able to establish whether the compression is reduced or if it merely is the range over which the BM responds compressively which is reduced. This information is important when considering fitting strategies of hearing aids.

It does appear that the TMCs are not very informative at frequencies with more severe losses, displaying TMC slopes close to linearity. The hearing loss causes the masker to operate at high intensity levels which makes it more difficult to obtain a clear picture of the TMC. One needs to consider if these TMCs offer extra information over and above the information available in the absolute thresholds and the IFMCs. An initial exploration of the TMC could be suggested, measuring masker thresholds using only a short (e.g. 0.02 s) and a long gap duration (e.g. 0.08 s). If little increase in the masker threshold is found by increasing the gap duration, this is a clear indication that no steep section is present in the TMC and that the slope of a straight line represents the compression at that particular frequency. In the case that a large difference in masker threshold is found using a short and a long gap, then measurement of the masker thresholds at intermediate gap durations is required to achieve a complete TMC.

CHAPTER 9

General discussion

The work presented in this thesis has successfully explored and introduced new approaches concerning the assessment protocols used for impaired listeners. This has resulted in the development of hearing profiles for impaired listeners. The aim of the hearing profiles of the impaired listeners is to gain a wide range of information on a person's hearing and to gain insight into the possible pathology underlying the hearing impairment. In future research, computer models of impaired hearing will be developed based on these hearing profiles.

In this chapter, the achievements and novel contributions of the thesis will be discussed. Attention will also be drawn to a number of limitations of the assessment procedure. Finally, the next step in the project will be presented, discussing the implications of the hearing profiles for the development of computer models and the fitting of hearing aids.

9.1 Achievements

1. Development of a user friendly, fast and reliable threshold estimation procedure

The procedure for threshold measurements was developed to be user friendly, efficient, reliable and fast. This was accomplished by refining a Single interval threshold measurement procedure. In Chapter 2, we presented the Single-Interval-Up-Down method and described the adjustments made to accommodate for our specific measurement requirements. The most significant adjustment was that we introduced an simple one-up one-down adaptive track to the method to avoid the problems associated with the stimulus selection procedure used in the Maximum

Likelihood method (Green, 1993). Secondly, a cue was added to the test stimulus to make the task easier. Finally, we implemented a new application of catch trials by aborting the threshold run when a listener made a mistake when presented with a catch trial.

In Chapter 3, we investigated how the accuracy of threshold estimates changes with the number of trials used in a threshold run. The aim of this exercise was to find the smallest number of trials we could use when making the measurements for the hearing profiles of the normal and impaired listeners. A formula was presented which enables us to predict the number of trials needed for a certain level of accuracy. This formula was validated against computer simulations and human data. It was shown that a threshold run consisting of 10 trials (not including the initials trials and the catch trials) yields standard deviation of 1.5 dB SPL. It was decided that this level of accuracy is appropriate for the measurements we would be making and using 10 trials per threshold run was adopted for the remainder of the project.

Chapter 4 was concerned with evaluating the Single-Interval-Up-Down method on a number of issues. First, we looked at user friendliness and showed that the method is simple and hardly requires any training. Almost no training effect was found for the measurement of absolute threshold and only a minor effect was found when measuring forward-masked thresholds. Secondly, the Single-Interval-Up-Down method (SI-UD) was compared with a traditional 2I2AFC, one-up, two-down procedure in terms of threshold estimates, accuracy and speed. The threshold estimates obtained using the two procedures were comparable, the standard deviations were smaller for the SI-UD method and the time needed for a threshold run was considerably shorter using the SI-UD method.

As mentioned before, Single Interval procedures are not as widely used as forced-choice procedures. During this project, we were met with criticisms within the psychophysical community. The main concerns were related to the possible biases caused by changes in the listener's criterion. However, we feel that the SI-UD method is a valuable procedure for our specific needs. First, it is consistent with measurement methods used in clinical practise. Secondly, consistent results were obtained in all listeners using the SI-UD. Finally, comparisons with forced-choice procedures has shown that both methods gives the same answers.

2. Development of a set of psychoacoustic tests which will provide a range of information on the hearing of individuals.

Chapter 5 outlined the psychoacoustic measures that will be used to create a psychoacoustic profile of the participants. Absolute thresholds provided information on the sensitivity of the listeners. Frequency selectivity was measured using Iso Forward Masking Contours (IFMC). Compression was measured using Temporal Masking Curves (TMC).

In chapter 6, we explored how thresholds change as a function of the duration of the stimulus in normal and impaired listeners. A reciprocal function was presented to fit the threshold/duration functions and it was shown that the parameters derived from this data distinguished not only between normal and impaired listeners, but also have potential to differentiate within the group of impaired listeners.

3. Development of hearing profiles in normal and impaired listeners

In Chapter 7, the hearing profiles of 3 normal listeners were presented. The findings showed to be consistent with results reported in the literature and justify using the assessment protocol with impaired listeners. It was also shown that the protocol is sensitive to subclinical abnormalities which would be missed by normal clinical assessment.

In Chapter 8, hearing profiles of 4 impaired listeners were generated. It was demonstrated that the assessment protocol is viable for use with impaired listeners. The listeners liked the testing method and had no problem performing the measurements. The profiles showed that large differences can be found in different listeners. This finding emphasizes the need for more individual treatments of hearing impairments. The results described in the hearing profiles have shown to be indicative of an underlying pathology. This information will be crucial when a computer model of the impaired listener is developed.

Apart from forming the basis for the development of a computer model of a listener's hearing, the profiles might also have the potential to form the basis of a more sophisticated patient classification system. The additional information may prove to reveal different sub-classes of impairment. At the moment, classification of the impairment is mainly based on the audiometric thresholds. If diagnosis can be formed on the underlying pathology, impairment classification could be based on the impairment rather than on, or in addition to, the audiometric thresholds.

9.2 Limitations

Psychoacoustic measures

One can argue that the measures used to form the hearing profiles are not sufficient and that more information should be included in the profiles. For example, the hearing profile would benefit from including a speech-in-noise measure. Apart from the IFMC measure there is little information available on how a listener performs in a noisy environment. This however is the most important complaint heard from impaired listeners. A speech-in-noise test has subsequently been incorporated in the assessment protocol and is considered to be a valuable addition to the hearing profile. First it will enable us to correlate the results on the IFMCs with the speech recognition in a noisy environment. It can be hypothesized that listeners with residual tuning will perform better in this test than listeners whose frequency selectivity is severely affected. Furthermore, the comparison of the 2 measures might reveal a relationship between a specific IFMC patterns and a specific speech recognition performance pattern. Finally, and most importantly, the speech-in-noise measure will be very valuable when implemented into a computer model of the auditory periphery. It will provide a direct and real-life evaluation of the performance of the computer model and of subsequent adjustments to the model. A possible method to evaluate the speech recognition capacities of the model is to link an automatic speech recognizer to the system. An impaired computer model will generate an impaired input in the speech recognizer resulting in deteriorated speech recognition performance. This will therefore be the ultimate test when exploring new hearing aid strategies. A hearing aid design or a speech processing strategy will only be considered successful if it can improve or even normalize the results of the automatic speech recogniser.

Testing time

Although the testing time is reduced considerably by adopting the SI-UD method, collecting all the measurements for the hearing profile is still time consuming. A further reduction in time is needed for the hearing profile to be viable in a clinical setting. This can be addressed by revisiting the testing conditions and considering if measures such as IFMCs and TMCs can be reduced, for example by omitting data points, without loosing considerable information. This reduction in data points has already been implemented for the threshold/duration measure where it was shown that 2 data points produce comparable information to all 7 initial data points.

The IFMCs are currently based on 7 data points representing masker thresholds at 7 different masker frequencies. The masker threshold at probe frequency is a crucial data point in the IFMC so obviously can not be omitted. The importance of the points around probe frequency seems to differ according to the situation of the listener. In normal listeners, sharp V-shaped IFMCs are generally found. In this situation, the masker threshold at probe frequency and 1 masker threshold on each side of the tip frequency could be sufficient to generate a reliable IFMC. The masker frequencies of these two points can be up to an octave away from the probe frequency (for example 0.5 and 1.6 * probe frequency). In impaired listeners however the IFMC are expected to be shallower and shifts in the tip frequency can occur. Exploration of masker frequencies around, and close to, the probe frequency (BF) is therefore important. We would suggest that IFMCs consisting of 5 data points, representing masker frequencies of 0.7, 0.9, 1, 1.1, and 1.3 * probe frequency, might be more suitable in impaired listeners. The masker threshold at probe frequency and 2 thresholds around and close to the probe frequency would provide information on the tip of the IFMC. An additional masker threshold on each side and at frequencies further removed from the probe frequency will inform us on the sharpness (or in this case, the shallowness) of the IFMC. Since IFMCs are expected to be shallower, not much benefit can be gained from measuring masker thresholds at masker frequencies an octave away from probe frequency.

A reduced testing procedure for the TMC measure can also be suggested. Analysis of the TMCs of the impaired listeners shows that the TMCs are in most cases a shallow straight line. The slope of the straight line might be steeper than the linear reference but is still considerably reduced compared to steep slopes found in the normal TMCs. It is thought that an initial exploration of the TMCs using only a short and a long gap durations could save testing time. If the difference between the masker thresholds associated with a short gap and a long gap is small, that is a strong indication that the TMC will result in a shallow, straight line and that compression is reduced or absent. There would be no need to test all the intermediate gap durations since those would generate masker thresholds which also fall onto the straight line. If a large difference between the 2 thresholds is found then further testing of the TMC is required to identify the steep sloping section of the TMC and acquire a complete image of the function.

Listeners

In this thesis only 4 hearing profiles of impaired listeners were presented. This is a limited sample and it is difficult to draw general conclusions. However, a further thirteen profiles are in the midst of being collected as an extension of the data already gathered from this project. The findings are generally consistent with what was shown in the hearing profiles in this thesis.

The degrees of hearing impairments we can currently test are limited to mild to moderately severe hearing losses. Impaired listeners with severe hearing losses are not able to conduct the measurements due to their high thresholds. In particular, measurements of masked thresholds work at stimulus levels close to the upper limit of the testing software. It can therefore be stated that the results presented in this theses are not a good representation of the population of hearing impaired listeners. On the other hand, the group of mild to moderately severe losses is probably the group which would most benefit from the information presented in the hearing profiles, if those form the basis of a computer model of the listener's hearing and if the hearing profile is used to fine tune the hearing aid fitting. It can be expected that the hearing profiles of the listeners with a severe hearing loss do not show the same wide range of results. It is highly probable that the IFMCs are flat (suggesting that no tuning is present) and that no compression is found. These listeners would unfortunately not profit from the detailed assessment procedure in the same manner as the listeners with a moderate to moderately severe loss. Furthermore the possibilities for fitting of the hearing aid would probably be limited to applying an overall amplification and a hearing profile would add little additional information to this treatment decision.

This project is aimed at exploring the wide spectrum that hearing impairment covers. It is not the principal goal to classify people into different impairment groups. The hearing of each listener will be modelled using the individual data collected on that listener. For this reason, no assumptions or criteria are being used at the moment to evaluate the inter-subject variability. Nevertheless, an extensive set of profiles might reveal specific impairment groups in the future. It is anticipated that suitable criteria concerning intra- and inter-subject variability will emerge (and be necessary) at that point.

9.3 Future research

As mentioned earlier, this thesis is part of a larger project which aims to model the hearing of impaired individuals. The hearing profiles presented in Chapter 8 are now suitable to start the modelling component of the project. The computer model of impaired hearing will be based on an existing computer model of the normal auditory periphery (Meddis, 2006). The model comprises of a cascade of modules, each of which represents a different stage in the processing of the auditory periphery: the outer-middle ear, the basilar membrane, the inner hair cell synapse and the auditory nerve. The output of one module is passed to the subsequent stage in the sequence. Each module is characterized by a number of parameters. It is those parameters which will be crucial in modelling hearing impairments.

The computer model has been adjusted so it can perform the psychoacoustic tests in our test battery. It generates thresholds similar to a normal human listener. The modelling of impaired hearing will proceed as follows: the computer model of normal hearing will be made impaired by adjusting the parameters associated with each module of the model. The aim is to make the model similarly impaired as the listener by matching the psychophysical thresholds of the model with the thresholds in the hearing profile of the listener. The pathology hypotheses formulated for each listener offer an important initial clue as to which module (parameter) needs changing. When the model thresholds match the human thresholds an evaluation can be made as to which parameters, and therefore which modules, have been changed. This is expected to provide an insight into the origin of the impairment. Preliminary modelling work has already produced encouraging results.

A final step in the project is to use the computer model of an impaired individual to find the best fitting strategy for the hearing aid of that particular person. The model would act as some kind of ‘tailor’s dummy’: it has all the right ‘measurements’ of the patient and can be used to try out as many different hearing aid settings and strategies as needed, without needing the presence or cooperation of the listener. A ‘hearing aid-module’ will be linked up to the impaired model and the performance of this ‘aided’ model will be evaluated on the basis of the psychoacoustic measurements. Different fitting strategies can be evaluated on the basis of the psychophysical results generated by the ‘aided’ model and a decision can be formulated concerning the best fit for a specific patient. This part of the ‘Hearing Dummy Project’ is expected to start in the foreseeable future.

9.4 Summary

This thesis was aimed at exploring new approaches concerning the assessment protocols of impaired hearing. Hearing profiles of normal and impaired listeners have been presented containing information on the hearing sensitivity, frequency selectivity and compression of an individual listener. The measurements were carried out using a fast and reliable threshold estimation procedure. The assessment protocol developed in this thesis, and the resulting hearing profiles, are a sound basis for the future development of computer models of hearing for impaired individuals. A computer model of an individual's hearing can subsequently be used to extensively evaluate the suitability of different hearing aid settings and enable the exploration of novel hearing aid designs.

References

Amitay, S., Irwin, A., Hawkey, D.J.C., Cowan, J.A., and Moore, D.R. (**2006**). "A comparison of adaptive procedures for rapid and reliable threshold assessment and training in naïve listeners," J. Acoust. Soc. Am., **119**, 1616-1625.

Arehart, K., Burns, E.M., and Schlauch, R.S. (**1990**). "A comparison of psychometric functions for detection in normal hearing and hearing impaired listeners," J. Speech Hear. Res., **33**, 433-439.

Baer, T., Moore, B.C.J., and Kluk, C. (**2002**). "Effects of low pass filtering on the intelligibility of speech in noise for people with and without dead regions at high frequencies," J. Acoust. Soc. Am., **112**, 1133-1144.

Baker, R.J., and Rosen, S. (**2001**)."Evaluation of maximum-likelihood threshold estimation with tone-in-noise masking," Brit. J. Aud., **35**, 43-52.

Barry, S.J., and Larson, V.D. (**1974**). "Brief-tone audiometry with normal and deaf school-age children," J. Speech Hear. Dis., **39**, 457-464.

Boothroyd, A. (**1968**). "Developments of speech audiometry," Sound, **2**, 3-10.

BS EN ISO 389-1:2000: Acoustics: Reference zero for the calibration of audiometric equipment-Part 1: Reference equivalent threshold sound pressure levels for pure tones and supra-aural earphones, identical to ISO 398-1.1989.

BS EN ISO 8253-1: British Society of Audiology Recommended Procedure (**2004**). Pure tone air and bone conduction threshold audiometry with and without masking and determination of uncomfortable loudness levels.

Buss, E., Hall, J.W., Grose, J.H., and Dev, M.B. (**2001**). "A comparison of threshold estimation methods in children 6-11 years of age, "J. Acoust. Soc. Am., **109**, 727-731.

Carhart, R., and Jerger, J.F. (**1959**). "Preferred method for clinical determination of pure-tone thresholds," J. Speech Hear. Dis., **24**, 330-345.

Carlyon, R.P., Buus S. and Florentine, M. (**1990**). “Temporal integration of trains of tone pulses by normal and by cochlearly impaired listeners,” J. Acoust. Soc. Am., **87**, 260-268.

Carney, A.E., Nelson, D.A. (**1983**).”An analysis of psychophysical tuning curves in normal and pathological ears,” J. Acoust. Soc. Am., **73**, 268-278.

Chung, D.Y. (**1981**). “Masking, temporal integration, and sensorineural hearing loss,” J. Speech. Hear. Res., **24**, 514-520.

Chung, D.Y. (**1982**). “Temporal integration-its relationship with noise-induced hearing loss,” Scand. Audiol., **11**, 153-157.

Chung, D.Y., and Smith, F. (**1980**). “Quiet and masked brief-tone audiometry in subjects with normal hearing and with noise-induced hearing loss,” Scand. Audiol., **9**, 43-47.

Clark, J.G. (**1981**). “Uses and abuses of hearing loss classification,” ASHA, **23**, 493-500.

Cohen, M.F. (**1982**). “Detection threshold microstructure and its effect on temporal integration data,” J. Acoust. Soc. Am., **71**, 405-409.

Dempsey, J.J., Maxon, A.B., (**1982**). “Audiology in hearing education – temporal integration functions in hearing-impaired children,” Ear Hear., **3**, 271-273.

Dillon, H. (**2001**). *Hearing aids.* (Thieme, New York).

Dubno, J.R., and Ahlstrom, J.B. (**2001**). “Forward- and simultaneous-masked thresholds in bandlimited maskers in subjects with normal hearing and cochlear hearing loss,”

Elliott L.L. (**1963**).”Tonal thresholds for short-duration stimuli as related to subject hearing level,” J. Acoust. Soc. Am., **35**, 578-580.

Elliott L.L. (**1975**). “Temporal and masking phenomena in persons with sensorineural hearing loss,” Audiology, **14**, 336-353.

Evans, E.F. (**1975**). “Cochlear nerve and cochlear nucleus,” in Handbook of Sensory Physiology, Vol. V/2, Auditory system, edited by W.D. Keidel and W.D. Neff (Springer-Verlag, Berlin), p. 1-108.

Florentine, M. (**1992**). “Effects of cochlear impairment and equivalent-threshold masking in psychoacoustic tuning curves,” Audiology, **31**, 241-253.

Florentine, M., Buus, S., and Geng, W. (**1999**). “Psychometric functions for gap detection in a yes-no procedure,” J. Acoust. Soc. Am. 106, 3512-3520.

Florentine, M., Buus, S., and Geng, W. (**2000**). “Towards a clinical procedure for narrowband gap detection I: a psychophysical procedure,” Audiology, **39**, 161-167.

Florentine, M., Fastl, H. and Buus, S. (**1988**). “Temporal integration in normal hearing, cochlear impairment, and impairment simulated by masking,” J. Acoust. Soc. Am., **84**, 195-203.

Florentine, M. and Houtsma A.J.M. (**1983**). “Tuning curves and pitch matches in a listener with a unilateral, low-frequency hearing loss,” J. Acoust. Soc. Am., **73**, 961-965.

Florentine, M., Marvit, P., and Buus, S. (**2001**). “Maximum-likelihood yes-no procedure for gap detection: effect of track length,” J. Am Acad. Audiol., **12**, 113-120.

Gengel R.W., and Watson, C.S. (**1971**). “Temporal integration: I. Clinical implications of a laboratory study. II. Additional data from hearing-impaired subjects,” J. Speech Hear. Disord., **36**, 213-244.

Gerken, G.M. Bhat, V.K.H., Hutchison-Clutter, M. (**1990**). “Auditory temporal integration and the power function model,” J. Acoust. Soc. Am., **88**, 767-778.

Glasberg, B.R., and Moore, B.C.J. (**1986**). “Auditory filer shapes in subjects with unilateral and bilateral cochlear impairments,” J. Acoust. Soc. Am., **79**, 1020-1033.

Glasberg, B.R., and Moore, B.C.J. (**2000**). “Frequency selectivity as a function of level and frequency measured with uniformly exciting notched noise,” J. Acoust. Soc. Am., **108**, 2318-2328.

Gleeson, M.J. (**1984**). “Alport’s syndrome: audiological manifestations and implications,” J. Laryngol. Otol., **98**, 449-465.

Goodman, A. (**1965**). “Reference zero levels for pure-tone audiometer,” ASHA, **75**, 262-263.

Gorga, M.P., Neely, S.T., Ohlrich, B., Hoover, B., Redner, J., and Peters, J. (**1997**). “From laboratory to clinic: a large scale study of distortion product otoacoustic emissions in ears with normal hearing and ears with hearing loss,” Ear Hear., **18**, 440-455.

Green, D.M. (**1993**). “A maximum-likelihood method for estimating thresholds in a yes-no task,” J. Acoust. Soc. Am., **93**, 2096-2105.

Green, D.M., Birdsall, T.G., and Wilson, P.T. (**1957**). “Signal detection as a function of signal intensity and duration,” J. Acoust. Soc. Am., **29**, 253-531.

Gu, X. , and Green, D.M. (**1994**). “Further studies of a maximum-likelihood yes-no procedure,” J. Acoust. Soc. Am., **96**, 93-101.

Hall, J.W., and Fernandes, M.A. (**1983**). “Temporal integration, frequency resolution, and off-frequency listening in normal-hearing and cochlear-impaired listeners,” J. Acoust. Soc. Am., **74**, 1172-1177.

Halpin, C. (**2002**). “The tuning curve in clinical audiology,” Am. J. Audiol., **11**, 56-64.

Harris, J.D., Haines, H.L., and Myers, C.K. (**1958**). “Brief tone audiometry,” Arch. Otolaryngol., **67**, 699-713.

Hastings, N.A.J., and Peacock, J.B. (**1975**). *Statistical Distributions: A Handbook for Students and Practitioners.* (Butterworth, London).

Heil, P., and Neubauer, , H. (**2001**). “Temporal integration of sound pressure determines thresholds of auditory-nerve fibers,” J. Neursci., **15**, 7404-7415.

Hughes, J, W. (**1946**). “The threshold of audition for short periods of stimulation,” Proc. R. Soc. London Ser. B, **133**, 486-490.

Hughson, W., and Westlake, H. (**1944**). “Manual for program outline for rehabilitation of aural casualties both military and civilian,” Transactions of the American Academy of Ophtalmology and Otolaryngology Supplement, **48**, 1-15.

ISO 389-8: Acoustics: Reference zero for the calibration of audiometric equipment - Part 8: Reference equivalent threshold sound pressure levels for pure tones and circumaural earphones.

Jepsen, M.L., and Dau, T. (**2008**). “Estimating the basilar-membrane input output-function in normal-hearing and hearing-impaired listeners using forward masking,” J. Acoust. Soc. Am., **123**, 3859.

Johnsson, L.-G., and Arenberg, I.K. (**1981**). “Cochlear abnormalities in Alport’s syndrome,” Arch. Otolaryngol., **107**, 340-349.

Kashtan, C.E. (**1999**). “Alport syndrome: an inherited disorder of renal, ocular, and cochlear basement membranes,” Medicine, **78**, 338-360.

Kemp, D.T. (**2002**). "Otoacoustic emissions, their origin in cochlear function, and use," Br. Med. Bull., **63**, 223-241.

Kluk, K., and Moore, B.C.J. (**2005**). "Factors affecting psychophysical tuning curves for hearing-impaired subjects with high-frequency dead regions," Hear. Res., **200**, 115-131.

Katz, J. (**2002**). *Handbook of Clinical Audiology (5th edn).* Philidelphia: Lippincott, Williams and Wilkins.

Leek , M.R. (**2001**). "Adaptive procedures in psychophysical research," Perception & Psychophysics, **63**, 1279-1292.

Leek , M.R., Dubno, J.R., He, N., and Ahlstrom, J.B. (**2000**). "Experience with a yes-no single-interval maximum-likelihood procedure," J. Acoust. Soc. Am., **107**, 2674-2684.

Leek , M.R., Hanna, T.E., and Marshall, L. (**1992**). "Estimation of psychometric functions from adaptive tracking procedures," Perception & Psychophysics, **51**, 247-256.

Levitt, H. (**1971**). "Transformed up-down procedures in psychoacoustics," J. Acoust. Soc. Am., **49**, 467-477.

Liberman M.C., and Dodds L.W. (**1984**). "Single-neuron labeling and chronic cochlear pathology. III Stereocilia damage and alterations of threshold tuning curves," Hear Res., **16**, 55–74.

Lopez-Poveda, E.A., and Alves-Pinto, A. (**2008**)."A variant temporal-masking-curve method for inferring peripheral auditory compression," J. Acoust. Soc. Am., **123**, 1544-1554.

Lopez-Poveda, E.A., Plack, C.J., and Meddis, R. (**2003**). "Cochlear nonlinearity between 500 and 8000 Hz in listeners with normal hearing," J. Acoust. Soc. Am., **113**, 951-960.

Lopez-Poveda, E.A., Plack, C.J., Meddis, R., and Blanco J.L. (**2005**). "Cochlear compression in listeners with moderate sensorineural hearing loss," Hear. Res. 205, 172-183.

Marvit, P., Florentine, M., and Buus, S. (**2003**). "A comparison of psychophysical procedures for level-discrimination thresholds," J. Acoust. Soc. Am., **113**, 3348-3361.

McFadden, S. L., Quaranta, N., and Henderson, D. (**1997**). "Suprathreshold measures of auditory function in the aging chinchilla," Hear. Res., **111**, 127-135.

McKee, S.P, Klein, S.A., and Teller, D.Y. (**1985**). “Statistical properties of forced-choice psychometric functions: Implications for probit analysis,” Percept. Psychophys., **37**, 286-298.

Meddis, R. (**2006**).“Auditory-nerve first-spike latency and auditory absolute threshold: A computer model,” J. Acoust. Soc. Am., **119**, 406-417.

Møller, A.R., (**2000**). *Hearing: Its Physiology and Pathophysiology*. (Academic Press, San Diego).

Moore, B.C.J. (**2004**). *An Introduction to the Psychology of Hearing* (Academic, London).

Moore, B.C.J. (**2007**). *Cochlear Hearing Loss, 2nd edition* (Wiley, Chichester).

Moore, B.C.J., and Alcantara, J.I. (**2001**).”The use of psychophysical tuning curves to explore dead regions in the cochlea,” Ear Hear, **22**, 268-278.

Moore, B.C.J., Glasberg, B.R., and Stone, M.A. (**2004**). “New version of the TEN test with calibrations in dB HL,” Ear Hear., **25**, 478-487.

Moore, B.C.J., Huss, M., and Vickers, D.A., Glasberg, B.R., Alcántara, J.I. (**2000**). “A test for the diagnosis of dead regions in the cochlea,” Br. J. Audiol., **34**, 205-224.

Moore, B.C.J., Killen, T., and Munro, K.J. (**2003**). “Application of the TEN test to hearing-impaired teenagers with severe-to-profound hearing loss.” Int. J. Audiol., **42**, 465-474.

Moore, B.C.J., and Oxenham, A.J. (**1998**). “Psychoacoustic consequences of compression in the peripheral auditory system,” Psychol. Rev., **105**, 108-124.

Nelson, D.A. (**1991**). “High-level psychophysical tuning curves: forward masking in normal-hearing and hearing-impaired listeners,” J. Speech. Hear. Res., **34**, 1233-1249.

Nelson, D.A., and Freyman, R.L. (**1987**). “Temporal resolution in sensorineural hearing-impaired listeners,” J. Acoust. Soc. Am., **81**, 709-720.

Nelson, D.A., and Schroder, A.C. (**1997**).”Linearized response growth inferred from growth-of-masking slopes in ears with cochlear hearing loss,” J. Acoust. Soc. Am., **101**, 2186-2201.

Nelson, D.A., Schroder, A.C., and Wojtczak, M. (**2001**). “A new procedure for measuring peripheral compression in normal-hearing and hearing-impaired listeners.” J. Acoust. Soc. Am., **110**, 2045-2064.

Neubauer, H., and Heil, P. (**2004**). “Towards a unifying basis of auditory thresholds: the effects of hearing loss on temporal integration reconsidered,” J. Assoc. Res. Otolaryngol., **5**, 436-458.

Neubauer, H., and Heil, P. (**2008**). “A physiological model for the stimulus dependence of first-spike latency of auditory-nerve fibers,” Brain Res., **1220**, 208-223.

Olsen, W.O., and Carhart, R. (**1966**). “Integration of acoustic power at threshold by normal hearers, “ J. Acoust. Soc. Am., **40**, 591-599.

Olsen, W.O., Rose, D.E., and Noffsinger, D. (**1974**). “Brief tone audiometry with normal, cochlear, and eighth nerve tumor patients,” Archs. Otolaryngol., **99**, 185-189.

Oxenham, A.J., and Moore, B.C.J. (**1995**). “Additivity of masking in normally hearing and hearing-impaired subjects,” J. Acoust. Soc. Am., **98**, 1921-1934.

Oxenham, A.J., and Moore, B.C.J. (**1997**). “Modeling the effects of peripheral nonlinearity in listeners with normal and impaired hearing,” in Modeling sensorineural hearing loss edited by Jesteadt W., Lawrence Erlbaum

Oxenham, A.J., and Plack, C.J. (**1997**). “A behavioral measure of basilar-membrane nonlinearity in listeners with normal and impaired hearing.” J. Acoust. Soc. Am., **101**, 3666-3675.

Patterson, R.D. (**1976**).”Auditory filter shapes derived with noise stimuli,” J. Acoust. Soc. Am., **59**, 640-654.

Patterson, R.D., and Nimmo-Smith, I. (**1980**).”Off-frequency listening and auditory-filter asymmetry,” J. Acoust. Soc. Am., **67**, 229-245.

Pedersen, C.B., and Elberling C. (**1972**). “Temporal integration of acoustic energy in normal hearing persons,” Acta Otolaryngol. (Stockh.), **74**, 398.

Pedersen, C.B., and Elberling C. (**1973**). “Temporal integration of acoustic energy in patients with presbyacusis,” Acta Otolaryngol., **75**, 32-37.

Pedersen, C.B., and Salomon, G. (**1977**). “Conductive hearing loss evaluated by brief-tone audiometry,” Acta Otolaryngol., **83**, 424-428.

Pentland, A. (**1980**). “Maximum likelihood estimation: the best PEST,” Perception & Psychophysics, **28**, 377-379.

Plack, C.J., and Drga, V. (**2003**). "Psychophysical evidence for auditory compression at low characteristic frequencies," J. Acoust. Soc. Am., **113**, 1574-1586.

Plack, C.J., Drga, V., and Lopez-Poveda, E.A. (**2004**), "Inferred asilar-membrane respose functions for listeners with mild to moderate sensorineural hearing loss," J. Acoust. Soc. Am., **115**, 1684-1695.

Plack, C.J., and Oxenham, A.J. (**1998**). "Basilar-membrane nonlinearity and the growth of forward masking," J. Acoust. Soc. Am., **103**, 1598-1608.

Plomp, R., and Bouman, M.A. (**1959**). "Relation between hearing threshold and duration for tone pulses," J. Acoust. Soc. Am., **31**, 749-758.

Rhode, W.S. (**1971**). "Observations of the vibration of the basilar membrane in squirrel monkeys using the Mossbauer technique," J. Acoust. Soc. Am., **49**, 1218-1231.

Rhode, W.S., and Recio A. (**2000**). "Study of mechanical motions in the basal region of the chinchilla cochlea," J. Acoust. Soc. Am., **107**, 3317-3332.

Rhode, W.S., and Robles, L. (**1974**). "Evidence from Mossbauer experiments for nonlinear vibration in the cochlea," J. Acoust. Soc. Am., **55**, 588-596.

Richards, A.M., Dunn, J. (**1974**). "Threshold measurement procedures in brief-tone audiometry," J. Speech Hear. Res., **11**, 446-454.

Rintelmann, W. (**1976**). "Auditory manifestations of Alport's disease syndrome." Trans. Sect. Otolanryngol. Am. Acad. Ophthalmol. Otolaryngol., **82**, 375-387.

Robles, L., Ruggero, M.A., and Rich, N.C. (**1986**). "Basilar membrane mechanics at the base of the chinchilla cochlea. I. Input-output functions, tuning curves, and response phases," J. Acoust. Soc. Am., **80**, 1364-1374.

Rosen, S., Baker, R.J., and Darling, A. (**1998**). "Auditory filter nonlinearity at 2 kHz in normal hearing listeners," J. Acoust. Soc. Am., **103**, 2539-2550.

Rosengard, P.S., Oxenham, A.J., and Braida, L.B. (**2005**). "Comparing different estimates of cochlear compression in listeners with normal and impaired hearing," J. Acoust. Soc. Am., **117**, 3028-3041.

Ruggero M.A. (**1992**). "Responses to sound of the basilar membrane of the mammalian cochlea," Curr. Opin. Neurobiol., **2**, 449-456.

Ruggero M.A., and Rich, N.C. (**1991**). “Furosemide alters organ of Corti mechanics: Evidence for feedback of outer hair cells upon the basilar membrane, J. Neurosci., **11**, 1057-1067.

Ruggero M.A., and Rich, N.C., Recio, A., Narayan, S.S., and Robles, L. (**1997**). “Basilar-membrane responses to tones at the base of the chinchilla cochlea,” J. Acoust. Soc. Am., **101**, 2151-2163.

Schairer, K.S., Nizami, L., Reimer, J.F., and Jestaedt, W. (**2003**). “Effects of peripheral nonlinearity on psychometric functions for forward-masked tones,” J. Acoust. Soc. Am., **113**, 1560-1573.

Schmiedt, R.A., Lang, H., Okamura, H., and Schulte, B.A. (**2002**). “Effects of furosemide applied chronically to the round window: A model of metabolic presbyacusis,” J. Neuroscience, **22**, 9643-9650.

Sheeley, E.C., and Bilger, R.C. (**1964**). “Temporal integration as a function of frequency,” J. Acoust. Soc. Am., **36**, 1850-1857.

Shelton, B.R., Picardi, M.C., and Green, D.M. (**1982**). “Comparison of three adaptive psychophysical procedures,” J. Acoust. Soc. Am., **71**, 1527-1533.

Snik, A.F.M., and Horst, J.W. (**1991**). “A new measure for determining the degree of abnormality of psychophysical tuning curves in hearing-impaired subjects,” Scand. Audiol., **20**, 191-195.

Spence, M., and Feth, L.L. (**1974**). “Effects of off-frequency detection in brief-tone audiometry,” J. Speech Hear. Res., **17**, 576-588.

Stainsby, T.H., and Moore, B.C.J. (**2006**). “Temporal masking curves for hearing-impaired listeners,” Hear. Res., **218**, 98-111.

Stelmachowicz, P.G., and Jestaedt, W. (**1984**). “ Psychophysical tuning curves in normal-hearing listeners: test reliability and probe level effects,” J. Speech. Hear. Res., **27**, 394-402.

Stelmachowicz P.G., Jesteadt W., Gorga M. P., and Mott J. (**1985**).”Speech perception ability and psychophysical tuning curves in hearing-impaired listeners,” J. Acoust. Soc. Am., **77**, 620-7.

Summers, V., Molis, M.R., Müsch, H., Walden, B.E., Surr, R.K., and Cord, M.T. (**2003**). "Identifying dead regions in the cochlea: Psychophysical tuning curves and tone detection in threshold-equalizing noise," Ear Hear, **24**, 133-142.

Thornton, A.R. and Abbas, P.J. (**1980**). "Low-frequency hearing loss: Perception of filtered speech, psychophysical tuning curves, and masking," J. Acoust. Soc. Am., **67**, 638-643.

Vanden Abeele, D., van de Heying, P.H., Creten, W., Graff, A., and Marquet, J.F.E. (**1992**). "Psychoacoustical tuning curves, Normative data for clinical use," Scand. Audiol., **21**, 3-8.

Vickers, D.A., Moore, B.C.J., and Baer, E. (**2001**)." Effects of low-pass filtering on the intelligibility of speech in quiet for people with and without dead regions at high frequencies," J. Acoust. Soc. Am., **110**, 1164-1175.

Viemeister, N.F., and Wakefield, G.(**1991**). "Temporal integration and multiple looks," J. Acoust. Soc. Am., **90**, 858-865.

Watson, A.B., and Pelli, D.G. (**1983**). QUEST: A Bayesian adaptive psychometric method," Perception & Psychophysics, **33**, 113-120.

Watson, C.S., Franks, J.R., and Hood, D.C. (**1972**). "Detection of tones in the absence of external masking noise. I. Effects of signal intensity and signal frequency," J. Acoust. Soc. Am., **52**, 633-643.

Watson, C.S., and Gengel, R.W. (**1969**). "Signal duration and signal frequency in relation to auditory sensitivity," J. Acoust. Soc. Am., **46**, 989-997.

Westergaard, O., Kluyskens, P., John, H.D. (**1972**). "Alport's syndrome: Histopathology of human temporal bones," ORL J. Otorhinolaryngol Relat. Spec., **34**, 262-272.

Wojtczak, M, and Viemeister, N. (**2003**). "Suprathreshold effects of adaptation produced by amplitude modulation," J. Acoust. Soc. Am., **114**, 991-997.

Wright, H.N. (**1968**). "Clinical measurement of temporal auditory summation," J. Speech Hear. Res., **11**, 109-127.

Wright, H.N. (**1978**). Brief Tone Audiometry. In *Handbook of Clinical Audiology, 2nd Edition,* 218-232, J. Katz (Ed.). (Williams and Wilkins Company, Baltimore)

Yasin, I., and Plack, C.J. (**2008**). "Slopes of psychometric functions for forward-masked tones with and without a suppressor," J. Acoust. Soc. Am., **123**, 3857-3858.

Yates, G.K. (**1990**). "Basilar membrane nonlinearity and its influence on auditory nerve rate-intensity functions," Hear. Res., **50**, 145-162.

Yates, G.K., Winter, I.M., and Robertson, D. (**1990**). "Basilar membrane nonlinearity determines auditory nerve rate-intensity functions and cochlear dynamic range, Hear. Res., **45**, 203-220.

Young, L.M., and Kanofsky, P. (**1973**). "Significance of brief-tone audiometry," J. Aud. Res., **13**, 14-25.

Zehnder, A.F, Adams, J.C., Santi, P.A., Kristiansen, A.G., Wacharasindhu, C., Mann, S., Kalluri, R., Gregory, M.C., Kashtan, C.E., and Merchant, S.N. (**2005**). "Distribution of type IV collagen in the cochlea in Alport Syndrome," Arch. Otolaryngol. Head Neck Surg., **131**, 1007-1013.

Zwislocki, J.J. (**1960**). "Theory of temporal auditory summation," J. Acoust. Soc. Am., **32**, 1046-1060.