Improving cochlear implant rehabilitation

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The studies presented in this thesis were conducted within the Amsterdam UMC, Vrije Universiteit Amsterdam, Otolaryngology – Head and Neck Surgery, Ear & Hearing, Amsterdam Public Health research institute, De Boelelaan 1117, Amsterdam, Netherlands.

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Improving cochlear implant rehabilitation

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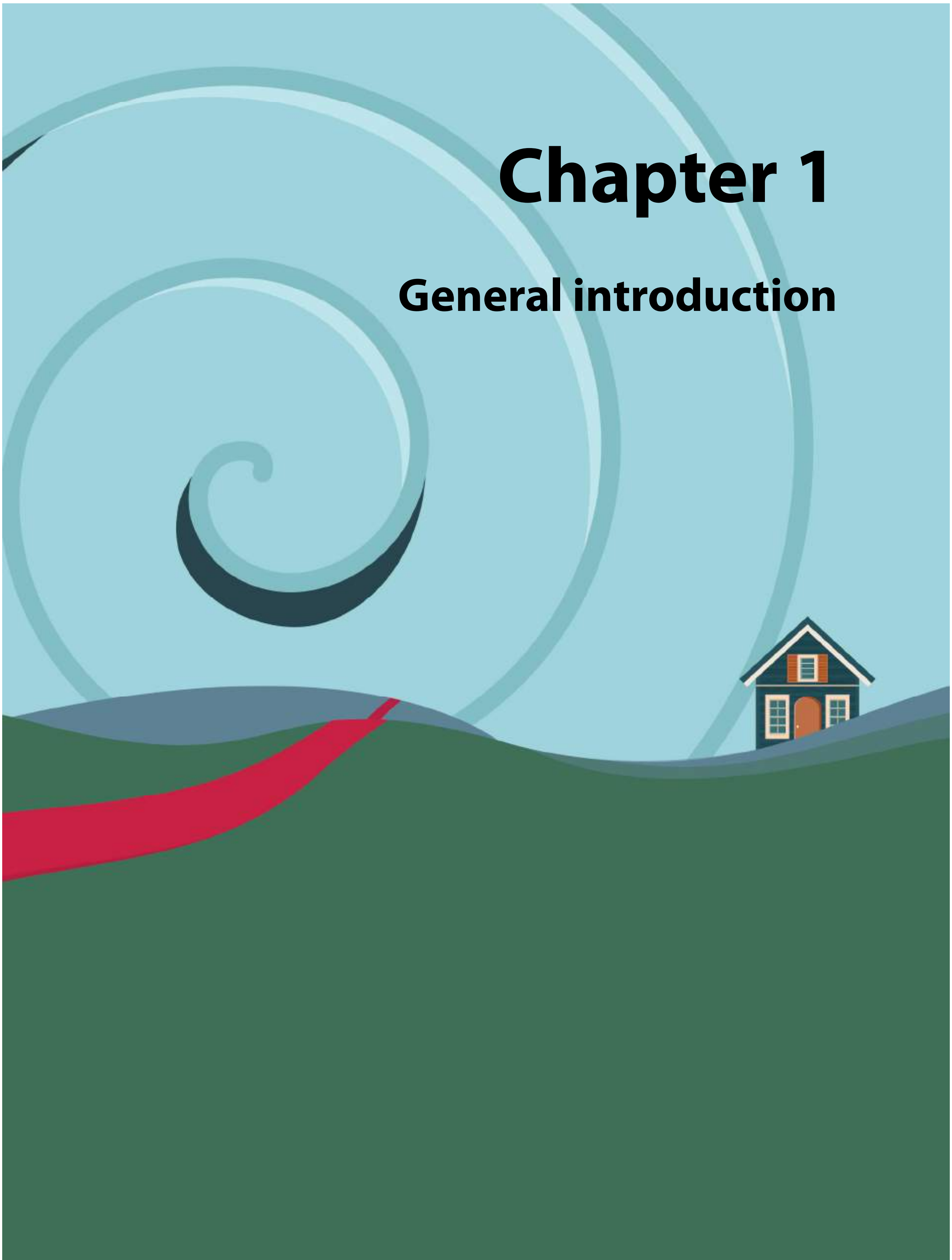
Chapter 1

General introduction



The cochlear implant (CI) is one of the most successful medical implants in medical history. It has the potential to improve the hearing status of patients with severe-to-profound sensorineural hearing loss. Researchers have long sought to find a solution for individuals with these hearing losses for whom hearing aids do not provide enough benefit. Sensorineural hearing loss is often caused by damaged inner hair cells in the cochlea. CIs bypass these damaged cells by directly stimulating the auditory nerve. A CI consists of an external and an internal part (Figures 1 and 2). The external part of the CI, the sound processor, processes sound that is captured by the microphones and transmits this through the skin via a coil to the internal receiver. The internal receiver subsequently converts the signal to electrical pulses which directly stimulate the auditory nerve via electrodes in the cochlea.

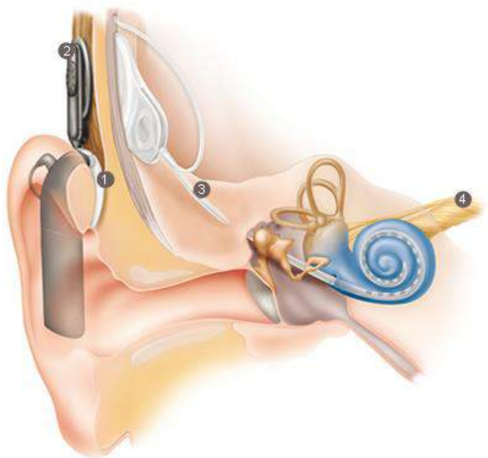


Figure 1. Cochlear implant with sound processor (1), coil (2), implant (3) and auditory nerve (4). Figure retrieved from Cochlear Ltd..

The first successful implantations of a single intracochlear electrode were performed by House and Doyle in the early 1960s. House continued his work on implants in 1967 together with Urban, which resulted in the first CI system that could be used outside of the laboratory (House & Urban, 1973). A few years later, Clark developed a multichannel implant at the University of Melbourne, which was successfully implanted in 1978 (Clark, 2003). The first successful implantation of a single channel CI in the Netherlands was performed in 1985. Further research and development worldwide, and collaborations between research groups and industry partners, ultimately led to the establishment of the main manufacturers of CIs: Advanced Bionics Corp. (USA, 1993), Cochlear Ltd. (Australia, 1981), MED-EL GmbH (Austria, 1977) and Neurelec (France, 1986, has been acquired by Oticon) (Eshraghi et al., 2012). Cochlear Ltd. holds the majority of the CI market share, with approximately 60% (Intelligent Investor).

Figure 2. The newest sound processor from Cochlear Ltd., the Nucleus® CP1000 or Nucleus® 7 sound processor. Figure retrieved from Cochlear Ltd..

Cochlear implant candidacy

CIs and its technology have evolved considerably since their first implantation and use, from a single intracochlear electrode providing merely a sensation of sound, to a multichannel device enabling speech recognition to the majority of CI users (Eshraghi et al., 2012). These technical improvements and the success of cochlear implantation have led to changing regulations and expanding candidacy criteria (Leigh et al., 2016; Snel-Bongers et al., 2018). Initially, the criteria were very strict and only unilateral implantation was done. Then, solely adults with postlingual bilateral profound hearing loss were considered for implantation. After the successful implantation and use of CIs in adults, implantation of children with severe-to-profound hearing loss was considered and performed. Nowadays, bilateral implantation of prelingually deafened children is considered to be the standard of care. The selection criteria for adults were expanded as well and now also include adults with substantial acoustic residual hearing in one or both ears. Furthermore, bilateral implantation in adults is routinely performed in some countries (e.g., United States and Australia) (Peters, Wyss, and Manrique, 2010).

In the Netherlands, adults with postlingual bilateral severe to profound hearing impairment are considered candidates for CI if well fitted hearing aids do not provide satisfactory results. The criteria include, for instance, less than 50% speech recognition in quiet but vary slightly between implant centres. In addition, patients have to be in good health without medical obstacles for implantation (e.g., intact auditory nerve and cochlea), and must be motivated for implantation and the intensive rehabilitation program, and have realistic expectations (OPCI, 2018).

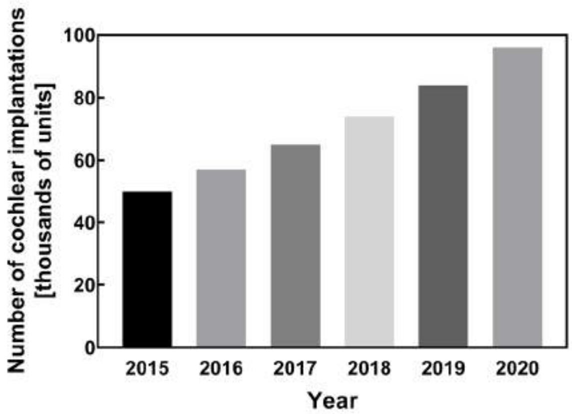


Figure 3. Number of cochlear implantations and estimations on the number of implantations up to 2020. Information retrieved from Technavio.

Because of technical improvements, expanding candidacy criteria and changing regulations, the number of CI users is increasing rapidly. Figure 3 shows the number of cochlear implantations worldwide. In the Netherlands, the number of CI users increased from approximately 1500 patients in 2005 to 7000 patients in 2017 (OPCI, 2018).

Cochlear implant care in the Netherlands

Cochlear implantations in the Netherlands are performed in each university medical centre by eight highly specialized teams. The CI teams consist of ENT surgeons, audiologists, speech and language therapists, and social workers. CI candidates have to get through an extensive selection procedure, encompassing appointments with a ENT surgeon, an audiologist, a speech and language therapist, and a social worker. Currently, adult CI candidates often find themselves on the waiting list for a prolonged period, due to limitations on the number of implantations stemming from boundaries set by the different healthcare system stakeholders.

The clinical care pathway of new and experienced CI users

The rehabilitation trajectory of the eight implant centres show slight differences. In this chapter, the rehabilitation trajectory offered to CI users in Amsterdam UMC, location VUmc is presented.

The first visit to the clinic for the rehabilitation trajectory after surgical implantation generally takes place three to five weeks after surgery to ensure adequate recovery of the surgical site. This visit marks the beginning of an intensive rehabilitation period, which covers the first three months post-implantation and is very intensive and demanding. It requires patients to frequently visit the clinic and to perform auditory training exercises at home. The schedule is often fixed and comprises weekly visits until six weeks after CI activation, followed by visits in weeks 9, 13, 28 and 52. The visits in the clinic include appointments with audiologists and speech and language therapists, in which speech recognition is assessed, the sound processor is fitted or fine-tuned, auditory training is provided, and the CI user is counselled on device use and maintenance.

The first visit after implantation comprises a fitting session with the audiologist and the first auditory training session with the speech and language therapist. The appointment with the audiologist is often split in two sessions on the same day with a break in between. The first session is aimed at providing the CI user with a basic fitting. Subsequently, the CI user is instructed to walk around the hospital and get somewhat accustomed to the sound of the CI. The basic fitting is fine-tuned in the second session, after which some patients might already be able to understand some speech. In the following sessions with the audiologist (e.g., week 1, 3, 5, 9, 12, 28, and 52 after activation of the sound processor), the fitting of the sound processor is fine-tuned. Generally, most of the adjustments to the fitting of the sound processor take place over the first few months, after which the fitting parameters remain relatively stable (Gajadeera et al., 2017).

To be able to hear sounds and understand speech with a CI, auditory training is provided by speech and language therapists in the clinic. Auditory training has been shown to improve speech recognition performance (Henshaw & Ferguson, 2013; Sweetow & Palmer, 2005). In addition to the training in the clinic, CI users are required to perform auditory training exercises at home with a training partner who is expected to attend the appointments in the clinic as well. During these appointments, training is provided by the speech and language therapist by means of exercises. The difficulty of these exercises is built up during the auditory training. The trajectory starts with the detection of sounds, followed by discrimination and identification of different sounds, and finally recognition of speech. Speech and language therapists also provide training in the use of accessories and counsel the CI user on device use and maintenance. The appointments with the speech and language therapist are provided weekly until six weeks after activation of the sound processor, followed by appointments in week 9, 12, 28 and 52.

The clinical care pathway of experienced CI users comprises annual visits to the clinic. During these visits, speech recognition is assessed and, if indicated, the fitting of the sound processor is adjusted or optimized. Training by speech and language therapists is generally no longer provided as part of standard clinical care after the first year of CI use.

Speech recognition assessment

Difficulties understanding speech in daily life is one of the biggest problems for people with hearing loss. As such, speech recognition ability is an important outcome measure. It is assessed approximately six times during the rehabilitation of newly-implanted CI users, and annually for experienced CI users. The results can be used both for fine-tuning of the fitting, but also to adjust the level and type of auditory training exercises.

In Dutch clinical practice, speech recognition is often assessed with monosyllable consonant-vowel-consonant (CVC) words in quiet (Bosman & Smoorenburg, 1995). Usually, the score is expressed as the percentage of correctly recognized phonemes. Speech recognition in quiet, however, does not reflect communication requirements in daily life. Therefore, speech recognition is also assessed in adverse conditions, with either sentences (Plomp & Mimpen, 1979; Versfeld et al., 2000) or digit-triplets (Kaandorp et al., 2015; Smits, Goverts, and Festen, 2013) in a background of steady-state masking noise. These tests use an adaptive procedure to estimate the speech reception threshold (SRT), which is defined as the signal-to-noise ratio (SNR) where a listener correctly recognizes 50% of the presented stimuli. The tests are performed in the clinic with a clinician and calibrated equipment. Test administration generally occurs in a sound-treated booth, with stimuli presented via a loudspeaker. The CI user is asked to repeat the presented speech stimuli verbally, after which a clinician judges the correctness of the response.

Fitting of the sound processor

The process of changing or fine-tuning the CI sound processor settings is referred to as programming or fitting. Vaerenberg et al. (2014) conducted a global survey on the current state of art for CI programming and concluded that CI programming practices vary considerably, both worldwide, and within countries. Despite the importance of CI fitting and rehabilitation, there is no evidence on the existence of good clinical practice.

The goal of programming or fitting of a CI is to maximize the use of the electrical dynamic range of the auditory nerve, to ensure both the audibility of soft sounds and comfort of loud sounds. The dynamic range is the difference between the threshold level (e.g., the minimal amount of electrical stimulation that is required to perceive sound) and comfortable level (e.g., the upper limit of stimulation judged to be most comfortable, or loud but comfortable). For CIs of Cochlear Ltd., these levels are referred to as T and C levels. During fitting sessions, emphasis is put on setting T and C levels (Vaerenberg et al., 2014). The T and C levels are often psychophysically determined, thereby requiring the CI users' feedback. T levels are determined by presenting a stimulus in a descending procedure where CI users are instructed to raise their hand or say "yes" when they hear the stimulus. C levels are determined by gradually increasing the presentation level of a stimulus where CI users are asked to indicate their loudness percept by pointing to categories on a 10-point loudness scale. The C level is set at a level that is “loud, but comfortable”. Once the C levels are determined, the levels are decreased by a certain percentage of the dynamic range. Subsequently, the sound processor is switched to live speech mode in which the clinician increases the C levels to find the user’s most comfortable level.

The adjustment of fitting parameters is often preceded by the assessment of auditory nerve and electrode functioning at the beginning of the fitting session. Assessment of the electrode functioning occurs by means of an electrode impedance measurement. The functioning of the auditory nerve can be assessed using the electrically evoked compound action potential (ECAP) (He, Teagle, and Buchman, 2017). The ECAP represents the response of the auditory nerve after electrical stimulation and can be recorded with the intracochlear electrodes. Here, one intracochlear electrode is used to stimulate the auditory nerve and another intracochlear electrode is used to record the neural response. Cochlear Ltd. provides an automated ECAP threshold measurement, called automatic neural response telemetry (autoNRT).

Although the majority of experienced CI users are generally positive about their device, it has been well documented that speech recognition remains a challenge in many listening conditions. In particular, difficulties are often experienced in noisy listening environments. In clinical practice, the CI user can be fitted with multiple programs for various listening environments. These programs often have specific names referring to specific listening environments (e.g., quiet, noise, music). With multiple programs for various listening environments, the user is required to characterize the listening environment and subsequently select the most appropriate program using a button on the sound processor or the remote control. Although CI users are regularly fitted with multiple programs for various listening environments, research has shown that users of hearing devices often leave their devices in the default setting (Banerjee, 2011; Cord et al., 2002; Searchfield et al., 2018; Van den Heuvel, Goverts, and Kapteyn, 1997).

In addition to multiple programs for various listening environments, modern CIs have the possibility to switch automatically through multiple settings for various listening environments. Here, the sound processor analyses the listening environment and decides whether the current settings have to be changed. Recent studies have shown that automatic program selection can benefit CI users (De Ceulaer et al., 2017; Gilden et al., 2015; Mauger et al., 2014; Wolfe et al., 2015). Therefore, automatically switching programs can be a good alternative for CI users who are not able or willing to switch between multiple programs for various listening environments.

In most modern CI sound processors, the listening environments encountered by the CI user are stored in so called datalogs, together with information about the daily usage of the device and the programs used in these listening environments (Busch, Vanpoucke, and van Wieringen, 2017; Mauger et al., 2014). In clinical practice, this datalog information can help clinicians to consider which CI users benefit from multiple programs and/or an automatically switching program.

The use of telehealth in the clinical care pathway of CI users

The care as usual for both new and experienced CI users has basically remained unchanged since the first implantations in 1985 in The Netherlands. However, the growing number of CI users increases the workload of CI centres. In addition, all CI centres are located in larger cities, which requires a substantial number of patients to travel considerable distances to reach their CI centre. The current rehabilitation schedule requires patients to visit their CI centre, even if there is no clinical need. Telecare provision models, where parts of the clinical routine of CI care are being moved out of the clinic to the patient's home, might be attractive for CI users as well as for the clinic. It could be used to monitor either progress or decline, and to identify those CI users who require visits to the clinic for further adjustments or optimization. This could result in time and cost savings for both CI centres and patients, and possibly more appropriate care adjusted to the patient's needs. Various applications of remote tests have been studied in the past, including intraoperative testing (Shapiro et al., 2008) and programming (Botros, Banna, and Maruthurkkara, 2013; McElveen et al., 2010; Ramos et al., 2009; Wesarg et al., 2010). In addition, the assessment of speech recognition either at a remote location (Goehring et al., 2012; Hughes et al., 2012) or at home (Cullington & Aidi, 2017) has been studied. However, these remote applications have so far only been applied to a limited extent in CI care and are currently not part of care as usual for CI users in the Netherlands.

Outline of the thesis

This thesis describes studies related to improvements in the clinical care pathway of new and experienced CI users. The thesis is divided in three sections, each focusing on different aspects of CI rehabilitation. The first section focuses on home self-assessment of speech recognition via a telehealth solution. In section 2, clinical data (fitting parameters) from adult CI users are used to predict speech recognition performance with the aim to improve fitting practices. The final section focuses on the use of automatic and manual switching programs to optimally adapt settings to listening environments encountered in daily life.

Section 1

This section describes the development, validation, and use of self-administered speech recognition tests at the CI users' home. **Chapter 2** addresses the technical challenges that were encountered in the development of self-administered speech recognition tests for experienced adult CI users at home. The effect of different types of masking noises (continuous versus discontinuous) on speech recognition in noise scores was examined. Furthermore, the use of an audio cable as an alternative to a loudspeaker for speech recognition testing was investigated, and a method to calibrate the home self-administered test setup was developed. Subsequently, the comparison of the self-administered speech recognition tests in quiet and in noise in the CI users' home with the standard tests in the clinic are described in **Chapter 3**. Potential effects of stimuli presentation modes (loudspeaker or audio cable) and assessment (by a clinician in the clinic or self-assessment at home) on speech recognition were investigated. With the successful outcomes of the self-administered speech recognition tests in experienced CI users, we integrated the tests in the clinical care pathway of newly-implanted CI users by means of a telehealth application, the MyHearingApp (MHA). **Chapter 4** presents a study that evaluated the use and feasibility of the home self-administered test functionality as part of the MHA, with newly-implanted CI users during the first three months of rehabilitation. User compliance of the newly-implanted CI users with the instructions to repeatedly perform speech recognition tests (twice a week during the first three months of rehabilitation) was evaluated. In addition, the progression in speech recognition performance during the first three months of rehabilitation is described. **Chapter 5** presents the results of a study in which the home self-administered speech recognition test setup was combined with the newly developed Australian digits-in-noise test to assess speech recognition in noise of bimodal and bilateral CI users. Speech recognition in noise of bimodal and bilateral CI users was assessed in different conditions to determine the binaural benefit and the effect of different masking noises (steady-state versus 16-Hz interrupted masking noise) on speech recognition in noise. This study was conducted at the Ear Science Institute in Perth, Australia.

Section 2

This section uses clinical data that was gathered during the annual follow-up visits of adult CI users to find possible sources of variability in speech recognition outcomes. In the study described in **Chapter 6** we examined the relationship between speech recognition in quiet and in noise, fitting parameters (i.e., T and C levels, dynamic range) and objective measurements (i.e., impedances and NRT thresholds) to find mapping rules to optimize speech recognition.

Section 3

Section 3 concerns the use of multimemory or automatically switching devices that are increasingly being used in clinical practice to enable CI users to use different settings for various listening environments. A review of the available literature on the use of manual and automatically switching multimemory devices by hearing aid CI users is provided in **Chapter 7**. This chapter further synthesizes the literature to evaluate whether hearing aid and CI users appreciate and adequately use the ability to switch between programs in various listening environments. The findings of the scoping review were used to design an experimental study which was conducted in experienced CI users to gather objective evidence concerning the use of manually or automatically switching programs for various listening environments. Datalogs are stored in the sound processor and contain information about the daily usage, encountered listening environments and program use. These data logs were used to examine whether CI users are able to select the most appropriate program in specific listening environments. The preliminary results of 15 participants are presented in the general discussion.

Chapter 8 summarizes and discusses the main findings of the studies in the different sections of this thesis. In addition, implications for clinical practice and directions for future research are presented.

Chapter 2

The Development of Remote Speech Recognition Tests for Adult CI Users: The Effect of Presentation Mode of the Noise and a Reliable Method to Deliver Sound in Home Environments

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Abstract

The number of cochlear implant users is increasing annually, resulting in an increase in the workload of implant centres in ongoing patient management and evaluation. Remote testing of speech recognition could be time-saving for both the implant centres as well as the patient. This study addresses two methodological challenges we encountered in the development of a remote speech recognition tool for adult CI users. First, we examined whether speech recognition in noise performance differed when the steady-state masking noise was presented throughout the test (i.e., continuous) instead of the standard clinical use for evaluation where masking noise stops after each stimulus (i.e., discontinuous). A direct coupling between the audio port of a tablet computer to the accessory input of the CI processor with a Personal Audio Cable (PAC) was used. The setup was calibrated to facilitate presentation of stimuli at a predefined sound level. Finally, differences in frequency response between the audio cable and microphones were investigated.

Keywords: Cochlear implant, remote testing, speech recognition, digits-in-noise, CVC recognition, personal audio cable, direct connect, discontinuous noise.

Introduction

The number of adult cochlear implant (CI) users is increasing annually. This annual growth is speeding up because the population of CI candidates increases due to changing regulations, expanding candidacy criteria, and technical improvements in CIs. The growing number of adult CI patients, both newly implanted and experienced users, increases the workload of implant centres and therefore promotes the search for new and innovative ways to provide healthcare for these CI users.

Remote testing and programming could be time and cost saving for both the audiologist and patient. Various remote applications for CI patients have been studied in the past, such as intraoperative testing (Shapiro et al., 2008) and programming (Botros et al., 2013; McElveen Jr et al., 2010; Ramos et al., 2009; Wesarg et al., 2010). Hughes et al. (2012) and Goehring et al. (2012) investigated the use of telehealth to measure speech recognition abilities, in quiet and in noise, in CI users. They found poorer speech recognition scores when testing in remote sites (i.e., small conference rooms with videoconferencing technology) compared to regular testing in a sound booth.

We developed remote tests to measure speech recognition in quiet and noise for adult CI recipients using a direct connection between the sound processor and audio port of a tablet computer. The current study addresses several challenges that we encountered in the development of a remote speech recognition tool. This article focuses on (1) the possible interaction of the advanced sound processing features in the Cochlear™ Nucleus® CP910 processor when the steady-state masking noise was presented throughout the test (continuous) instead of noise that stops after each stimulus (discontinuous) and (2) a reliable way to deliver sound to the processor.

The effect of discontinuous noise on the results of speech-in-noise testing

Speech-in-noise tests generally use steady-state noise that starts 0.5 to 1s before and ends after the speech stimulus (e.g., a word, digit-triplet, or sentence). The use of discontinuous noise allows the subject to respond to the experimenter during the quiet period. However, modern hearing instruments, including CIs, contain adaptive sound processing features such as noise reduction and adaptive algorithms. During testing, the relatively slow-acting advanced sound processing features need time to become fully active and may remain in a transition state after switching from the quiet response period to the short period where the stimulus is presented in noise. The results of the speech-in-noise test with discontinuous steady-state noise may be affected by these sound processor features and, therefore, may not reflect speech-in-noise recognition abilities in daily life. Thus, the first aim was to study the effect of the sound processor features on speech recognition test outcomes in CI users for a speech-in-noise test with the steady-state noise presented as either continuous or discontinuous.

We also investigated the effect of these two different presentation modes of the noise on speech recognition outcomes in normal-hearing individuals to rule out possible other factors (i.e., absence of an auditory cue, habituation to noise, and startle reflexes). In the case when the noise is presented continuously during the test, subjects receive no auditory cue about the start of the speech stimulus. In contrast, when the discontinuous noise is used, subjects expect the speech stimulus to be presented approximately 1s after the start of the noise. The absence of this auditory cue could result in poorer speech-in-noise outcomes in continuous noise compared to when presented in discontinuous noise. Therefore, in the continuous noise condition, we have implemented a visual cue on the tablet screen at the exact moment of presentation of the speech stimulus. It is, however, unclear whether this visual cue supports the attention processes in the same way as the auditory cue in discontinuous noise. It is possible that an increase in speech recognition outcomes might occur due to auditory habituation to continuous noise, and the onset of the noise in the test with discontinuous noise could cause a startle reflex each time the noise starts playing, which may in turn influence speech recognition scores. Thus, the second aim was to explore whether the differences between the presentation modes of the steady-state noise yield an effect on speech recognition scores in normal-hearing listeners.

Methods

Subjects

Twelve normal-hearing subjects (2 males and 10 females; mean age 27 years; range: 22-41 years) and 16 CI users (8 males and 8 females; mean age 64 years; range: 44-83 years) participated in this study. Normal-hearing subjects had pure-tone thresholds not exceeding 15 dB HL at any octave frequency between 500 and 4000 Hz. All participants were native Dutch speakers. The CI patients acquired their severe hearing impairment after the age of seven, had at least one year experience with their CI, and used the CP910 sound processor. This study was approved by the local medical ethical committee. All participants signed informed consent prior to the subject's participation.

Procedure

The digits-in-noise test (Smits et al., 2013) uses 24 digit-triplets (e.g., 6-5-2) presented in steady-state noise, which are randomly chosen from a list of 120 triplets, to estimate the speech reception threshold (SRT). The SRT represents the signal-to-noise ratio (SNR) where the listener recognizes 50% of the triplets correctly. The digit-triplets were presented at varying SNRs following an adaptive strategy, with a 2 dB step size. The overall presentation level was fixed at 65 dBA, with a start SNR of -4 dB for normal-hearing subjects and 0 dB for CI users. A digit-triplet was judged to be correct when all digits were entered correctly. The SNR of a subsequent triplet depended on the correctness of the response on the previous triplet. For the normal-hearing participants, the SRT was calculated as the average SNR over triplet 5 to 25. Triplet 25 was not presented, but the SNR was based on the response correctness and SNR of the preceding digit-triplet. For the CI patients, two dummy triplets with a fixed SNR were added to make sure that the noise reduction algorithms had settled (approximately five seconds), while also keeping the attention of the test subject. The SRT was calculated as the average SNR over triplet 7-27. Again, triplet 27 was not presented but the SNR was based on the response correctness and SNR of the preceding digit-triplet. All tests were performed in two test conditions, with the noise presented either continuously during the whole test or discontinuously with quiet response periods, in a sound-treated room. Participants performed the digits-in-noise test three times (practice, test and retest) in each condition. Analyses were based on the outcomes of the test and retest measurements.

The tests for the normal-hearing subjects were performed with a laptop, with sound played by an external sound card (creative sound blaster X-FI HD SB1240, Creative Labs) and delivered monaurally to the (subjectively) better ear through Sennheiser HDA200 headphones. The participants had to enter the response on a keyboard. The SRT for the continuous and discontinuous noise condition was measured in one session.

The speech recognition tests for the CI patients were administered with a tablet computer (Lenovo ThinkPad 10). The participants were seated in front of a loudspeaker (Genelec HT 205), which was connected to an external sound card (creative sound blaster X-FI HD SB1240, Creative Labs). The CI users who normally used a contralateral hearing aid did not use this hearing aid during the tests. The contralateral ear was not occluded, but thresholds were at a level where no contribution to speech recognition could be expected from the unaided ear. The participants verbally repeated the digit-triplets they recognized, which were entered by the experimenter. The SRT in continuous and discontinuous noise was measured in different sessions, as the measurements for this study were part of a larger study in which more speech recognition tests were assessed. As a consequence the digits-in-noise tests with continuous noise were administered in the first session.

Results

The SRT scores of the two conditions averaged for the 12 normal-hearing and 16 CI participants are shown in Figure 1.

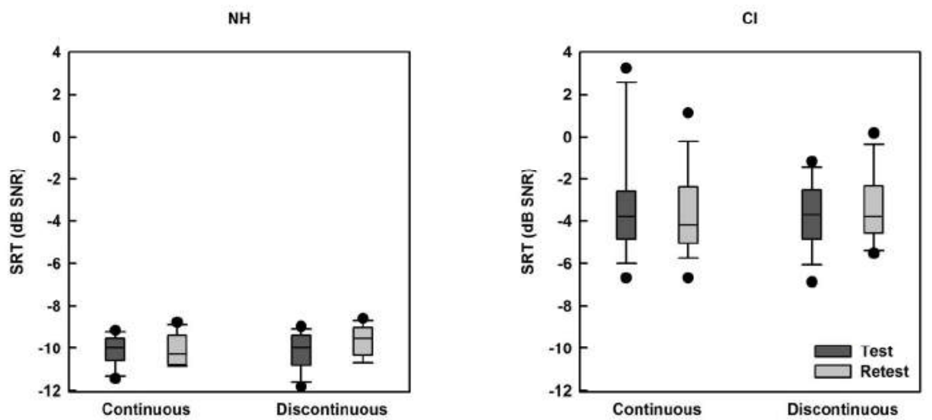

Figure 1. Boxplots representing the distribution of digits-in-noise SRT scores with the noise presented either continuously during the whole test or discontinuously with quiet response periods, in twelve normal-hearing subjects (left) and sixteen CI users (right). The plots show upper and lower quartiles (boxes), median (crossbar), and extremes of more than 1.5 times the inner quartile range (circles).

The data were analyzed using a repeated measures ANOVA with condition (noise continuously during the whole test vs. discontinuously) and measurement number (test, retest) as within-subject factors. The ANOVA for normal-hearing subjects yielded no significant main effect for condition ($F[1,11] = 1.237$, $p = 0.290$) and test number ($F[1,11] = 1.249$, $p = 0.288$), and no interaction between condition and measurement number ($F[1,11] = 2.798$, $p = 0.123$). The ANOVA on the SRT data from the CI patients showed no significant main effect of condition ($F[1,15] = 0.047$, $p = 0.832$) and test number ($F[1,15] = 0.002$, $p = 0.966$), and no interaction between condition and test number ($F[1,15] = 2.122$, $p = 0.166$). These results suggest that there is no significant difference between SRTs measured with steady-state noise that is presented continuously during the whole test and steady-state presented with quiet response periods after each stimulus, for normal-hearing subjects and CI users with the CP910 sound processor.

Stimulus delivery

Remote testing of speech recognition requires that the quality of loudspeakers of desktop computers, tablet computers or smartphones has no effect on the test results. Background noise and room acoustics are other important factors that are difficult to control in home-settings and could affect the test results (Goehring et al., 2012; Hughes et al., 2012). In our study, the audio signal was presented directly through the accessory input provided on the CP910 sound processor to avoid these unwanted effects. The direct coupling between

Figure 2. Experimental setup with a CI connected to a tablet computer using a PAC.

the audio port of the tablet computer and the accessory input of the sound processor using a Personal Audio Cable (PAC) was examined as a means to achieve the strict requirements needed for reliable home testing. Special attention was given to the calibration of the system and the frequency responses of the microphones and PAC. The experimental setup is shown in Figure 2.

Calibration

The system was calibrated to assure that speech stimuli were presented at a predefined sound level (measured in dBA or dB SPL) to the sound processor at specified volume and sensitivity settings through the PAC. A CP910 sound processor, fitted with a map in slot 1 with volume and sensitivity settings of 10 and 12, respectively, and mixing ratio set to 'accessory only' was connected to the tablet computer with a PAC. The calibration noise file available for the speech materials used (digits-in-noise and consonant-vowel-consonant (CVC) words in quiet) were presented through a loudspeaker in a sound booth. The sound pressure levels were readout directly from the internal sound level meter of the sound processor. These levels were compared to measurements done by a Brüel and Kjaer Type 2250 sound level meter positioned at the same location in the room as the sound processor. The broad-band differences in levels (dB SPL and dBA) were within 1 dB for calibration noise. These differences were assumed to be due to the position of the processor microphones and their slightly different frequency response compared to the A-weighting curve. Therefore, it was concluded that the internal sound level meter of the sound processor can be used for accurate calibration of the level of incoming signals. For this purpose, the broad-band oot mean square level of the input signal is adjusted (depending upon the speech material) such that the desired level is achieved when the input signal is presented through the PAC.

Frequency response

When the PAC is connected, the sound processor shapes the incoming signal with a frequency response to mimic the frequency response of the microphone signal (pre-emphasis). The frequency response was measured using different pure tones and the levels were read from the sound processor. A small difference was found between the frequency response of the signals that were delivered by the PAC and the microphones with a maximum deviation of approximately +3 dB at 6000 Hz. A difference in frequency response could possibly have an effect on speech recognition scores. Therefore, we performed an experiment in which we compared the speech recognition scores in quiet and in noise with a PAC using the original speech material, and speech material that was filtered to compensate for the difference in frequency response.

Methods

Subjects

Seven CI users participated in this study, all were part of the group of 16 CI users described earlier.

Procedure

Speech recognition tests included a test with monosyllabic words in quiet (Bosman and Smoorenburg, 1995) and digits-in-noise with continuous noise (Smits et al., 2013). The monosyllable words had a consonant-vowel-consonant structure, thus containing three phonemes. The words were spoken by a female speaker and presented in quiet at an intensity of 65 dB SPL. Three lists of 12 words were assessed. The procedure for the digits-in-noise test for CI users was explained earlier. For this experiment, the noise was presented continuously during the whole test.

A digital equalization filter was constructed to compensate for the differences in frequency characteristics between microphone input and PAC input. The audio files were filtered with this filter to ensure that the frequency response of the stimuli presented via the PAC matched the frequency response of the stimuli presented via the microphone. The speech recognition tests were first performed with the non-filtered audio files, followed by the tests with filtered audio files. All tests were performed in a sound-treated booth with the stimuli presented via a PAC.

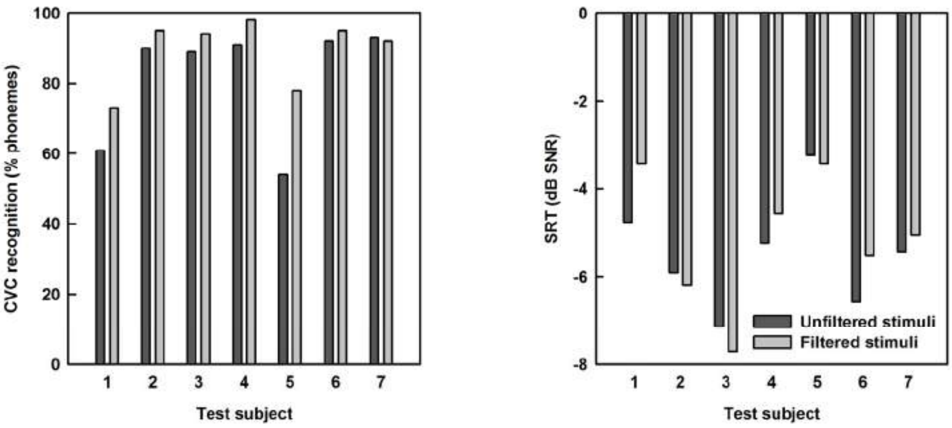

Figure 3. CVC recognition in quiet in percentage phonemes correct (average of three lists), measured with unfiltered and filtered stimuli presented via a PAC in seven CI patients (left). Digits-in-noise SRT score (average of test and retest) measured with unfiltered and filtered stimuli presented via a PAC in seven CI patients (right).

Results

The results of the speech recognition tests in quiet and in noise, measured with original and filtered stimuli presented with a PAC, are illustrated in Figure 3.

A Wilcoxon signed-rank test showed that there was a significant difference in the CVC recognition scores obtained with the unfiltered (mean = 81.4%; SD = 16.5) and filtered (mean = 89.3%; SD = 9.7%) audio files ($Z = -2.201$, $p < 0.05$). The results of the speech recognition tests in noise were subsequently analyzed. A Wilcoxon signed-rank test revealed no significant difference in SRT scores acquired with the PAC with unfiltered (mean = -5.5 dB SNR; SD = 1.3) and filtered (mean = -5.1 dB SNR; SD = 1.5) audio files ($Z = -1.183$, $p = 0.237$).

Discussion

The current study addresses two methodological challenges we encountered in the development of a remote speech recognition tool for adult CI users. First, we examined whether speech recognition in noise performance differed when the steady-state masking noise was presented continuously during the whole test instead of the standard use where the masking noise stops after each stimulus (discontinuous). We found no significant difference between those conditions for normal-hearing individuals, suggesting that the visual cue is appropriate to alert the listener that the next stimulus is to be presented. It also suggests that habituation to the noise or a startle reflex in the discontinuous noise tests with onset of the noise do not exist or have no effect on the test results. For the CI patients, lower (better) SRTs for the test with continuous noise were expected because several features in the CP910 sound processor (i.e., AGC, ADRO, ASC, SNR-NR) are slow-acting features with time constants in the order of seconds, and some have shown to improve speech recognition scores in noise (see Wolfe et al. (2015) for an overview). The results of the current study, however, showed no significant differences in speech recognition scores between continuous or discontinuous noise for CI patients using the CP910 sound processor. A possible reason for this finding could be the low SNRs for the signals in the digits-in-noise test that may reduce the effectiveness of noise reduction systems. Versfeld and Goverts (2013) demonstrated the effect of a carrier phrase on speech recognition scores in quiet and reported differences in scores for some hearing aids. This effect may also exist for CI users.

Second, we investigated the direct coupling between the tablet computer and sound processor using a PAC, by focusing on the calibration of the setup and differences in frequency response between PAC and microphones. Previous studies by Goehring et al. (2012) and Hughes et al. (2012) indicated that the characteristics of the test environment (i.e., a combination of background noise and reverberation) have an effect on speech recognition in CI patients. We used a PAC to deliver sound directly to the sound processor and could therefore bypass possible negative effects of the test environment. The PAC, however, introduces a small difference in the frequency response compared to the frequency response of the microphones. This difference is typically not noted when listening through the PAC to, for instance, music, but it may affect formal audiological testing. The effect of the difference in frequency responses on speech recognition outcomes was examined with the use of filtered audio files. Performance of speech recognition in quiet was better with the filtered audio files than with the unfiltered audio files, whereas no difference was found in speech recognition in noise test scores. The difference in speech recognition scores in quiet might be due to a training effect throughout the different test sessions. However, previous to the CVC tests with filtered audio files, participants performed a multitude of CVC tests since the assessments were part of a larger study, which makes the likelihood of a training effect very small. There might also be an effect of the specific CVC list used in a test, despite the minimal phonemic differences among the lists (Bosman and Smoorenburg, 1995). Different lists were used for each test in a different condition, but were the same for all participants. Another underlying cause might be that CI recipients normally use the microphone and may need to adapt to the slightly different spectrum of the sound that is delivered by the PAC. Possibly, this effect is more prominent for speech-in-quiet testing than for speech-in-noise testing where the masking noise is the main factor that corrupts the signal.

This study is part of a project targeting older users (60 years or older) of hearing implants and aims at allowing them to more effectively use their hearing implants in daily life. Implant centres are often located in larger cities, which requires many patients to travel long distances. The latter might be a problem when patients become older and potentially less mobile. Remote testing has the potential to reduce the need for appointments and might therefore reduce the need for transportation. The average age of our participants was 64 years (range: 44-83 years). All participants were able to perform the remote tests without any problems and all participants reported positive experiences with the remote testing. It was considered easy to connect the PAC, launch the application and perform the tests, and the complete test block took no more than half an hour. One participant was not able to execute the tests himself (plug in the PAC and type the response) due to a medical condition; however, with the help of a family member who entered the answers, he was able to perform the tests.

In conclusion, the setup as used in this study gives promising results and provides a solid base for future studies on remote assessment of speech recognition abilities in CI users in quiet and in noise. We have shown that speech-in-noise test outcomes are not influenced by the use of steady-state noise when presented continuously during the whole test. In addition, we have shown that stimuli can be presented with predefined levels using a direct connection via a PAC between a tablet computer and sound processor. In a future study, we will examine whether home assessment of speech recognition in CI users yields valid outcomes by comparing these to the outcomes of speech recognition assessment in a clinical setting. In addition, possible differences between stimulus presentation by a loudspeaker or PAC and assessment by an experimenter or self-assessment by the CI user will be investigated. Digital streaming of stimuli to the processor either via the remote control or Bluetooth might become available in the near future and may increase the ease of remote testing even further.

Chapter 3

Assessment of Speech Recognition Abilities in Quiet and in Noise: A Comparison Between Self-Administered Home Testing and Testing in the Clinic for Adult Cochlear Implant Users

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Abstract

Self speech recognition tests in quiet and noise at home are compared to the standard tests performed in the clinic. Potential effects of stimuli presentation modes (loudspeaker or audio cable) and assessment (clinician or self-assessment at home) on test results were investigated. Speech recognition in quiet was assessed using the standard Dutch test with monosyllabic words. Speech recognition in noise was assessed with the digits-in-noise test. Sixteen experienced CI users (aged between 44 and 83 years) participated. No significant difference was observed in speech recognition in quiet between presentation modes. Speech recognition in noise was significantly better with the audio cable than with the loudspeaker. There was no significant difference in speech recognition in quiet at 65 dB and in speech recognition in noise between self-assessment at home and testing in the clinic. At 55 dB, speech recognition assessed at home was slightly but significantly better than that assessed in the clinic. The results demonstrate that it is feasible for experienced CI users to perform self-administered speech recognition tests at home. Self-assessment by CI users of speech recognition in quiet and noise within the home environment could serve as an alternative to the tests performed in the clinic.

Keywords: Cochlear implant, self-administered home tests, speech recognition, consonant-vowel-consonant, digits-in-noise, audio cable, direct connect.

Introduction

The care of cochlear implant (CI) patients is intensive, and it is provided by specialized cochlear implant centres. Throughout the first year after implantation, CI users visit their implant centres frequently, often followed by annual visits. The Netherlands has eight cochlear implant centres (one in each university medical centre), all located in larger cities. A substantial number of patients must travel considerable distances to reach their cochlear implant centres. The number of newly implanted CI users is increasing rapidly due to changing regulations, expanding candidacy criteria and technical improvements in CI, which results in an increase in the number of CI candidates. In the Netherlands, the number of CI users has increased from approximately 1500 patients in 2005 to 7000 patients in 2016, while the number of cochlear implant centres has remained the same (OPCI, 2018). The growing number of CI users (both new and experienced) is increasing the workload of cochlear implant centres. Remote testing and programming or self-administered home testing could result in both time and cost savings for cochlear implant centres, as well as for patients. Various applications of remote tests have been studied in the past, including intraoperative testing (Shapiro et al., 2008) and programming (Botros, Banna, and Maruthurkkara, 2013; McElveen et al., 2010; Ramos et al., 2009; Wesarg et al., 2010). Hughes et al. (2012) and Goehring et al. (2012) demonstrated that testing of speech recognition at remote sites yield unreliable test results. The objective of the current study is to compare newly developed self-administered tests of speech recognition abilities in quiet and in noise within a home environment to standard clinical tests.

Speech recognition testing is part of a clinical routine for CI users, and it is typically performed in a soundbooth. Speech recognition can be assessed in quiet and in noise. In the Netherlands, such tests are based on sentences (Plomp & Mimpen, 1979; Versfeld et al., 2000), monosyllabic words (Bosman & Smoorenburg, 1995), or digit-triplets (Smits, Goverts, and Festen, 2013). In newly implanted users, speech recognition is assessed approximately six times during the intensive rehabilitation program (i.e., the first 3-6 months after implantation). For experienced CI users, it is assessed annually, in order to monitor either progress or decline. Speech recognition tests are typically performed with stimuli presented through a loudspeaker, after which the CI patient is asked to verbally repeat what he or she has heard. A clinician judges the correctness of the response and registers the score.

Self-administered home testing of speech recognition abilities in CI users requires technical adjustments if it is to become a viable alternative to testing in the clinic. First, the consistent and accurate presentation of stimuli is of paramount importance to the ability of detecting changes in speech recognition scores and meeting clinical standards. In studies by Hughes et al. (2012) and by Goehring et al. (2012), poorer speech recognition scores were obtained for stimuli presented with a loudspeaker when assessed in remote sites (e.g., a small conference room with videoconference technology), when compared to regular testing in a soundbooth. Background noise and reverberation were the most important factors that had a negative influence on speech recognition in the remote condition. Second, the home tests require the CI patients to connect the audio cable and type their responses on the tablet computer. It is therefore important that the automated evaluation of the response by the application is similar to the manual evaluation by clinicians when scoring verbal responses in the clinic (Francart, Moonen, and Wouters, 2009). The algorithms used to judge the patient's response in self-administered home tests should distinguish between deviations of the target word and the typed response due to misunderstandings of phonemes and those due to the individual's spelling ability.

To overcome the challenges identified in the studies by Goehring et al. (2012) and Hughes et al. (2012), the use of an audio cable for sound delivery to the CI in speech recognition testing was examined (De Graaff et al., 2016). An audio cable was used to create a direct coupling between the sound processor and the audio port of a tablet computer. It provided a direct mode for presenting the stimuli to the CI without the interference of background noise and reverberation. The study showed that stimuli can be presented at predefined levels using an audio cable. Although the study revealed a slight difference in frequency response between the microphone input and audio cable input, this difference did not have any significant effect on speech recognition scores in quiet or in noise. The authors therefore concluded that the use of an audio cable is a viable alternative to the use of a loudspeaker and that it can thus be used to detect changes in speech recognition scores.

The study by De Graaff et al. (2016) focused on the technical setup, calibration and the use of an audio cable instead of a loudspeaker to present stimuli during speech recognition testing. The objective of the current study was to compare the outcomes of self-administered home tests of speech recognition abilities in quiet and in noise to the outcomes of standard tests in the clinic. We hypothesized that the results of the home speech recognition tests would not be significantly different from the results of tests performed in the clinic. Potential effects of different stimuli presentation modes (i.e., loudspeaker or audio cable) and test format (soundbooth testing by a clinician or self-administered home testing) were investigated separately.

Materials and Methods

Study participants

A power analysis (G*Power 3.1) showed that a minimum of 12 participants were needed to detect a 1 dB difference in speech recognition in noise (based on the SEM of 1.1 dB for the digits-in-noise test in CI users by Kaandorp et al. (2015)) between conditions (power = 0.82). Sixteen adult unilateral CI users (8 males; 8 females) participated in this study. The average age was 64 years (range: 44 to 83 years). All participants were native speakers of Dutch. All CI users were postlingually deaf, with the onset of severe bilateral hearing impairment after the age of seven years. As a criterion for inclusion, all participants must have scored at least 60% on the consonant-vowel-consonant (CVC) test (phoneme scoring) during the last annual visit to the clinic. This inclusion criterion was set because a minimum speech recognition score in quiet is required to be able to assess speech recognition in noise with the digits-in-noise test (Kaandorp et al., 2015; Smits, Goverts, and Festen, 2013). No selection criteria were set in terms of experience with the use of computers. All CI users had at least one year of experience with the CI, and all were using the Cochlear™ Nucleus® CP910 sound processor. Participants enrolled in the study voluntarily and provided informed consent at the beginning of the study. The participants received a fee of €7.50 per hour and were reimbursed for their travel expenses. The study was approved by the Medical Ethics Committee of VU University Medical Centre. Table 1 lists the demographic characteristics of the participants.

Table 1. Demographic characteristics of the participants.

Participant	Gender	Age (years)	Duration of severe hearing impairment (years)*	CI experience (years)†	Implant type‡
1	M	74	7	2	CI24RECA
2	F	81	73§	5	CI512
3	M	58	16	2	CI24RECA
4	F	58	36	6	CI24REHybrid
5	M	51	43	6	CI512
6	M	67	53	6	CI24RECA
7	F	68	28	7	CI24RECA
8	M	67	47	6	CI24RECA
9	M	48	40§	17	CI24M
10	F	81	23	6	CI24REHybrid
11	M	63	33	5	CI512

F, female; M, male.

*Years between self-reported onset of (severe) hearing impairment and implantation; †Years of CI experience; ‡Type of implant from Cochlear Ltd; §Estimation of the duration of severe hearing impairment.

Participant	Gender	Age (years)	Duration of severe hearing impairment (years)[*]	CI experience (years)[b]	Implant type[c]
12	M	83	34	2	CI24RECA
13	[illegible]	61	21	6	CI512
14	[illegible]	44	27	2	CI24RECA
15	[illegible]	65	8	7	CI24RECA
16	[illegible]	53	8	3	CI24RECA

F, female; M, male.

[*]Years between self-reported onset of (severe) hearing impairment and implantation; [b]Years of CI experience; [c]Type of implant from Cochlear Ltd; [d]Estimation of the duration of severe hearing impairment.

Study design

This study was conducted as a prospective within-subject repeated measures study, in which the individual subjects served as their own controls. Speech recognition performance was assessed in three test sessions. The first and third session took place in the clinic, and the second session took place at the participants' home. The three sessions were completed within 1-2 weeks. Details of the three test sessions are described in the following paragraphs and listed in Table 2. All participants completed the entire study protocol across the three sessions.

The data acquired in the first two sessions were used to investigate the potential effect of stimuli presentation mode (loudspeaker or audio cable) and test format (soundbooth testing by a clinician or self-administered home testing). The third session was added as a control condition to identify potential learning effects for speech recognition in quiet. Speech recognition in noise was also assessed in the third session. However, the tests were performed with a different type of masking noise. These data were described elsewhere (De Graaff et al., 2016). The order of the test sessions was not counterbalanced across the participants, but it did follow a sequence that could easily be implemented in a standard clinical setting. The majority of the participants needed guidance and training in the use of the audio cable and tablet computer to self-administer the speech recognition tests. Therefore, the first session not only served as a test session, but also served as a training session for the self-assessment of the tests to be performed at home.

Table 2. Summary of test procedures and standard error of measurements (SEM) for speech recognition in quiet (65 and 55 dB) and noise (adaptive and fixed procedures).

Session	Location	Presentation mode	Tester	Test	Presentation level	CVC lists	SEM
1	Clinic	LS	Clinician	DIN adaptive	65 dBA		1.2 dB
				DIN fixed	65 dBA		11.2%
				CVC	65 dB	1-3	7.7%
				CVC	65 dB	4-6	6.5%
		AC	Clinician	DIN adaptive	65 dBA		1.7 dB
				DIN fixed	65 dBA		8.3%
				CVC	65 dB	7-9	8.3%
				CVC	65 dB	10-12	7.2%
2	Home	AC	Subject	DIN adaptive	65 dBA		1.2 dB
				DIN fixed	65 dBA		6.3%
				CVC	65 dB	13-15	7.1%
				CVC	65 dB	16-18	6.3%
3	Clinic	LS	Clinician	CVC	65 dB	19-21	6.9%
				CVC	65 dB	22-24	6.6%

AC, audio cable; CVC, Consonant-vowel-consonant; DIN, Digits-in-noise; LS, loudspeaker; SEM, Standard error of measurement.

3

In the remainder of this article, the tests with the loudspeaker in the clinic will be referred to as clinic – loudspeaker, with the terms clinic – audio cable and home – audio cable referring to the tests performed with an audio cable in the clinic and at home, respectively.

Test procedures

Speech recognition was assessed using monosyllabic words in quiet and digit-triplets in noise. The order of the tests was fixed, and each test was performed three times in each session.

Speech recognition in quiet

The monosyllabic words in the speech recognition tests have a CVC structure and were pronounced by a female Dutch speaker (Bosman & Smoorenburg, 1995). The CVC words were presented in quiet at 55 and 65 dB SPL. The two different presentation levels of 55 and 65 dB SPL were selected, as they are most representative of speech levels in daily life and are commonly used to assess speech recognition in quiet within the context of clinical practice in the Netherlands. At each presentation level, three lists of 12 CVC words were presented in each condition. Different lists were used for each test in a different condition, but they were the same for all participants. The subjects were instructed to repeat or enter everything they understood, even if it was a single phoneme or a nonsense word. The response on the first word was not included in the calculation of the test score. Scores for speech recognition in quiet were calculated as the percentage of 33 phonemes that had been recognized correctly.

The responses were typed by the CI patient on the tablet computer and automatically compared to the presented CVC word. Firstly, the software identified the middle phoneme by selecting it from a list of all possible vowels and vowel combinations, and then scored it by comparing it to the middle phoneme of the presented word. Finally, the first and last phonemes were identified and scored. A set of rules was defined in order to allow the software application to determine the correctness of the typed responses, allowing for specific deviations in the spelling of the target response. For example, graphemes that represented the same phoneme in Dutch (e.g., 'ei' and 'ij' for the diphthong /ɛi/, or 'd' and 't' for the word final consonant /t/) were both assessed as correct representation of the target phoneme.

Speech recognition in noise

The standard digits-in-noise speech recognition test (Smits, Goverts, and Festen, 2013) uses 24 digit-triplets (e.g., 6-5-2) presented in steady-state speech-shaped noise. The digit-triplets are selected at random from a list of 120 digit-triplets. The test is designed to estimate the speech reception threshold (SRT), which represents the signal-to-noise ratio (SNR) at which the listener can recognize 50% of the triplets correctly.

The digit-triplets were presented at varying SNRs following an adaptive strategy, with the overall presentation level fixed at 65 dBA (i.e., the level of the mixed speech and noise signals was kept constant). The current study used masking noise which was presented continuously throughout the test. The initial SNR was set at 0 dB SNR. The SNR of each digit-triplet depended on the correctness of the response on the previous digit-triplet, with the subsequent digit-triplet presented at a two dB higher SNR after an incorrect response, and at a two dB lower SNR after a correct response. To be scored as a correct response, all three digits must be repeated in the order presented.

In this study, the actual test was preceded by two dummy digit-triplets with a fixed SNR of 0 dB in order to ensure that the noise reduction algorithms of the sound processor had settled, while maintaining the attention of the test subject. Each test thus consisted of a total of 26 digit-triplets. The SNR of the virtual 27th digit-triplet was based on the SNR and correctness of the response of the 26th digit-triplet. The SRT was calculated as the average SNR of digit-triplets 7 to 27, omitting the first six digit-triplets of the test (two dummy digit-triplets and the first four digit-triplets of the test). In each condition, three digits-in-noise tests were performed. The first measurement was a practice list, which was not included in the final analysis, and the final two measurements were treated as test and retest.

A fixed SNR procedure was used in addition to the standard adaptive procedure. This procedure uses a fixed SNR to assess the percentage of correctly recognized digit-triplets. The average SRT of the test and retest of the adaptive digits-in-noise tests in the clinic + loudspeaker condition was used as the fixed SNR, for all conditions and was therefore different for each participant. The fixed SNR procedure was chosen because the speech recognition score is most sensitive to change around 50% speech recognition (i.e., the SRT measured with the adaptive procedure). At this point, the slope of the psychometric curve is at its steepest.

The score of the fixed procedure was calculated as the percentage of correctly identified digit-triplets in the series of digit-triplets 7-27 (see description above). In this context as well, three measurements of the digits-in-noise test were administered, with the first measurement (i.e., the practice test) excluded from the analyses.

Technical setup

The speech recognition tests were administered with a tablet computer (Lenovo ThinkPad 10, Lenovo) using software developed by Cochlear Technology Centre (Mechelen, Belgium). A visual cue was provided along with the presentation of the stimulus, in order to alert the participant to the digit-triplets or words. The loudspeaker and the audio cable set-ups were calibrated to ensure equal presentation levels, regardless of presentation mode

(De Graaff et al., 2016). Briefly, the accuracy of the internal sound level meter of the CP910 sound processor was determined by comparing the sound pressure levels read out directly from the internal sound level meter to those measured with a Bruël and Kjaer Type 2250 sound level meter. Pure tones and speech-shaped noise were used as calibration signals. The differences that were found were all within 1 dB. The internal sound level meter of the sound processor was subsequently used to adjust the output signal of the tablet computer in order to achieve the designated level in the CP910 sound processor. The loudspeaker set-up was calibrated at the distance where CI patients are seated during testing (approximately 70 cm). The calibration procedure ensured that the intensity of the internal signals in the sound processor were identical to the acoustic signals delivered through the loudspeaker and the electrical signals delivered through the audio cable. For the CI user, therefore, an acoustic signal of 65 dB SPL delivered through the loudspeaker should be equal to the 65 dB signal delivered through the audio cable. Throughout this article, the levels of the signals delivered through the loudspeaker and audio cable are expressed in dB, thus corresponding to the assumed equivalence in signal level for the CI user.

For the tests with the loudspeaker, the tablet computer was connected to an external sound card (Creative Sound Blaster X-FI HD SB1240, Creative Labs) which was connected to a loudspeaker (Genelec HT 205). The tablet computer was connected to a monitor screen to ensure that the participants could see the visual cue on the monitor screen, while the clinician used the tablet computer to record the participant’s responses. For the tests with the audio cable, the tablet computer was disconnected from the external sound card and monitor screen. The participants could look at the tablet computer screen to see the visual cue. The audio cable was connected to the tablet computer and inserted into the accessory socket of the CI.

Prior to the measurements, the participants were asked about the program and volume settings they preferred in daily life. If a participant used multiple programs and/or changed volume and sensitivity settings, a single program with defined volume and sensitivity settings was chosen for the assessment of all speech recognition tasks. Participants were instructed to use the same settings throughout the different test conditions. To ensure that the participants received sound only through the audio cable and not mixed with sound received through the microphones when using the audio cable, the accessory mixing ratio was set to ‘accessory only’. Finally, the microphone covers of each participant’s CI were replaced before testing. The CI users who used contralateral hearing aids in daily life did not use them during the tests. The contralateral ear was not occluded, but thresholds were at a level at which no contribution to speech recognition could be expected from the unaided ear.

Experimental setup: Clinic

The tests in the clinic were performed in a soundbooth with a clinician, who typed the responses of the participant. In the clinic, the participants were tested in two conditions. First, the tests were performed with a loudspeaker, with the participants seated in front of the loudspeaker at a distance of approximately 70 cm. The audio cable was subsequently connected to the sound processor and tablet computer, in order to perform the same tests with the audio cable.

Experimental setup: Home

At home, participants connected the audio cable to the tablet computer and their CI processor, and then launched the application. Participants entered their responses on the tablet computer. To overcome potential problems with the use of the tablet computer or the execution of the tests in the home environment, the participants received a manual and brief instructions during the first session in the clinic.

Results

Reliability

The test-retest reliability of the tests in the various conditions was assessed according to the standard error of measurement (SEM) for a single test, which reflects the agreement between measurements. The sum of squares of the standard deviations (SD) of the different measurements for each subject was divided by the number of test subjects (n). Subsequently, the SEM was calculated by taking the square root of this number.[1] The SEM for the adaptive digits-in-noise test (1.2 dB, 1.7 dB and 1.2 dB for the clinic + loudspeaker, clinic + audio cable, and home + audio cable conditions, respectively) is similar to the value of 1.1 dB observed in the study by Kaandorp et al. (2015). The SEM values for speech recognition in quiet and noise are listed in Table 2.

1 $SEM = \sqrt{(\sum SD^2)/n}$

Speech recognition in quiet

For statistical analyses, the speech recognition in quiet scores were transformed to rationalized arcsine units (RAU) (Sherbecoe & Studebaker, 2004; Studebaker, 1985) to normalize variance across the range of scores. Paired samples t-tests were conducted to test for significant differences between speech recognition in quiet scores in various conditions for 65 and 55 dB presentation levels separately. First, the outcome of the self-administered home test for speech recognition in quiet was compared to the standard test in the clinic. Speech recognition in quiet assessed with a loudspeaker and audio cable at 65 dB did not differ significantly from each other. At 55 dB, speech recognition in quiet in the clinic + loudspeaker condition was slightly but significantly lower than in the home + audio cable condition.

Because of the significant difference between the clinic + loudspeaker and home + audio cable condition with regard to speech recognition in quiet at 55 dB, subsequent t-tests for paired samples were conducted for 55 dB. A Bonferroni correction for multiple comparisons was used, with a p value less than 0.05/3 = 0.017 regarded as statistically significant. First, a paired samples t-test was conducted to investigate potential effects of presentation mode (loudspeaker or audio cable). No significant difference in speech recognition in quiet at 55 dB between clinic + loudspeaker and clinic + audio cable was observed. Second, a paired samples t-test was conducted to investigate potential effects of test format (soundbooth testing by a clinician or self-administered home testing). Speech recognition in quiet assessed in the home + audio cable condition was significantly better than the CVC recognition assessed in the clinic + audio cable condition.

Finally, speech recognition in quiet assessed in the first and third session in the clinic with a loudspeaker were compared for the sound pressure levels of 55 dB and 65 dB, separately. For both presentation levels, speech recognition in quiet assessed in the first clinic + loudspeaker condition was significantly lower than speech recognition in quiet assessed in the third session. The mean (± SD) and individual scores for speech recognition in quiet at 65 dB and 55 dB presentation levels in different conditions, and the results of the statistical analyses are depicted in Figure 1 and listed in Table 3.

Table 3. Means and standard deviations (SD), and summary of paired samples t-test results for speech recognition in quiet and in noise.

	Mean	SD	Mean	SD	*t*	df	*p*
Clinic vs home testing	**Clinic – loudspeaker**		**Home – audio cable**				
Speech recognition in quiet							
65 dB	80.9%	7.6%	77.6%	9.9%	1.623	15	.125
55 dB	82.3%	9.3%	85.6%	10.5%	2.373	15	**.031***
Speech recognition in noise							
Adaptive	-3.4 dB SNR	2.2 dB SNR	-4.8 dB SNR	2.2 dB SNR	3.182	15	**.006***
Fixed	57.3%	14.3%	67.1%	17.6%	-1.998	15	.064
Presentation mode	**Clinic – loudspeaker**		**Clinic – audio cable**				
Speech recognition in quiet							
55 dB	82.3%	9.3%	81.1%	11.8%	0.322	15	.752
Speech recognition in noise							
Adaptive	-3.4 dB SNR	2.2 dB SNR	-5.0 dB SNR	2.1 dB SNR	3.218	15	**.006***
Test format	**Clinic – audio cable**		**Home – audio cable**				
Speech recognition in quiet							
55 dB	81.1%	11.8%	85.6%	10.5%	-4.343	15	**.001***
Speech recognition in noise							
Adaptive	-5.0 dB SNR	2.1 dB SNR	-4.8 dB SNR	2.2 dB SNR	-0.571	15	.576
Learning effect	**Clinic – loudspeaker (session 1)**		**Clinic – loudspeaker (session 3)**				
Speech recognition in quiet							
65 dB	80.9%	7.6%	84.6%	8.4%	-2.186	15	**.045***
55 dB	82.3%	9.3%	85.4%	8.8%	-2.952	15	**.01***

* Significant *p* value (paired samples t-test).

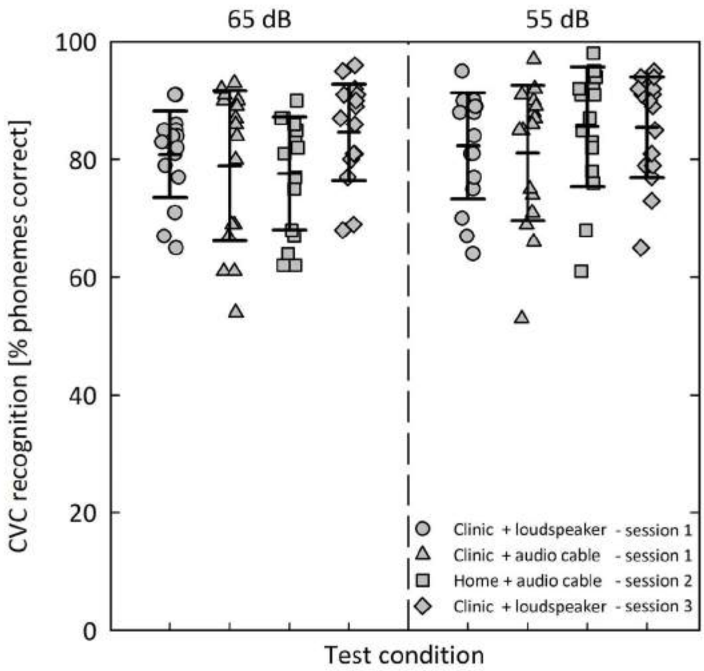

Figure 1. Speech recognition in quiet measured in four different conditions across the three test sessions. In each test condition, CVC words were presented at 65 dB (left panel) and 55 dB (right panel), respectively. The symbols represent individual scores and the horizontal lines represent mean and ± 1 standard deviation.

Speech recognition in noise

Digits-in-noise adaptive procedure

A paired samples ⍰test was conducted to compare the mean SRT assessed in the clinic and at home. The SRT assessed in the clinic + loudspeaker condition was significantly higher (i.e., worse) than the SRT assessed in the home + audio cable condition.[2]

Paired samples ⍰tests were conducted to investigate potential differences in presentation mode (loudspeaker or audio cable) and test format (soundbooth testing by a clinician or self-administered home testing). A Bonferroni correction for multiple comparisons was used, with a *p* value smaller than 0.05/3 = 0.017 regarded as statistically significant. The SRT assessed in the clinic + loudspeaker condition was significantly higher (i.e., worse) than the SRT assessed in the clinic + audio cable condition. No significant difference in SRT between soundbooth testing by a clinician and self-administered home testing was observed. The mean (± SD) and individual scores for the adaptive and fixed digits-in-noise test procedures are presented in Figure 2. The results of the statistical analyses are summarized in Table 3.

2 Corrected from published version.

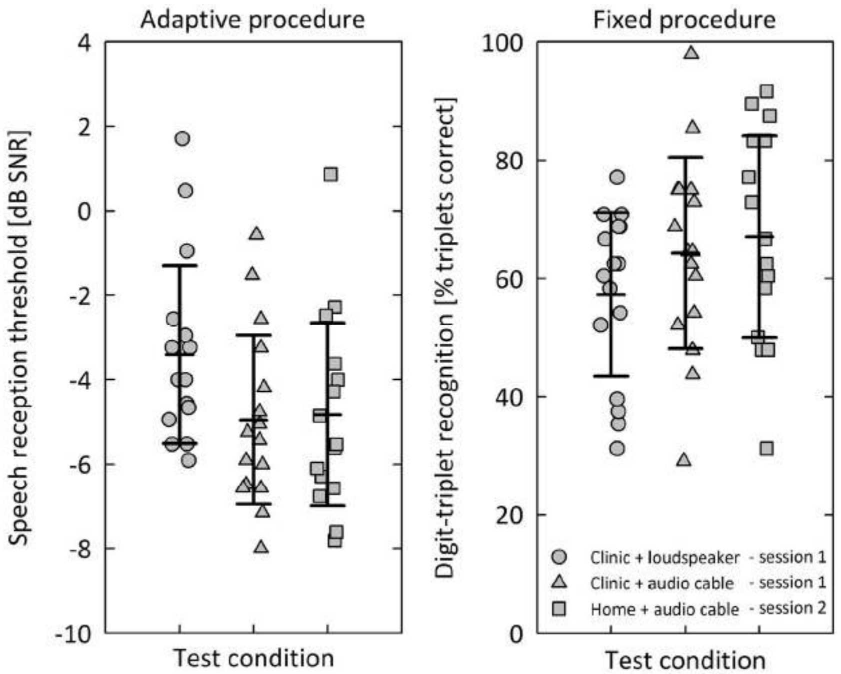

Figure 2. Speech recognition in noise measured in three different conditions using the adaptive (left panel) and fixed procedures (right panel). The digits-in-noise test was used to obtain speech reception thresholds in dB SNR from the adaptive procedure and % triplets correct scores from the fixed procedure. The symbols represent individual scores and the horizontal lines represent mean and ± 1 standard deviation.

Digits-in-noise fixed procedure

Before the statistical analyses were performed, the digit-triplet recognition scores were transformed to RAU scores. A ￼ test for paired samples was conducted to compare the digit-triplet recognition assessed in the clinic and at home. The digit-triplet recognition was not significantly different between the clinic + loudspeaker and home + audio cable conditions.[3]

3 The fixed SNR in all conditions was based on the SRT of the adaptive digits-in-noise tests in the clinic + loudspeaker condition. This results in higher percentages of correct scores for the audio cable conditions compared to the loudspeaker conditions. The SRT in the cable conditions were almost 1.5 dB better than those of the SRT in the clinic + loudspeaker condition. Given the 20%/dB slope of the psychometric curve for normal hearing subjects (Smits et al., 2013), a 15-20% higher score for the cable conditions could be expected. Although the data suggests a trend, it was not significant.

Discussion

The objective of this study was to compare the outcomes of self-administered home tests of speech recognition abilities in quiet and in noise to the outcomes of standard tests performed in the clinic. We investigated potential effects of stimuli presentation modes (loudspeaker or audio cable) and test format (soundbooth testing by a clinician or self-administered home testing). We hypothesized that the results of the home speech recognition tests would not be significantly different from the results of tests performed in the clinic. The study revealed no significant difference between measurements taken with a loudspeaker and those taken with an audio cable for speech recognition in quiet. The measurements for speech recognition in noise, however, were significantly lower (i.e., better) with the audio cable than they were with the loudspeaker, but there was no significant difference between the SRT assessed in the clinic or at home.

Despite the fact that more than half of our participants were 60 years or older, all were able to perform the self-administered home tests without any reported problems. All of the participants reported positive experiences with the self-administered home tests: they considered it easy to connect the audio cable, launch the application, and perform the tests. The results are thus promising, and they indicate that self-administered home testing is a feasible option for the standard clinical adult CI population. Several patients indicated that they were not proficient in the use of computers. Even though we emphasized to prospective candidates that participation in the study did not require any experience in computer use, bias might have been present in terms of cooperation, intelligence, and computer use, as all of the participants of this study were volunteers. In addition, all of our study participants had relatively good average SRTs. The average SRT -3.4 (± 2.2) dB SNR in the loudspeaker condition is lower (i.e., better) than the average SRT of -1.8 (± 2.7) dB SNR reported by Kaandorp et al. (2015) for a group of 18 adult CI users. The average monaural SRT for a group of 12 normal-hearing adults was -9.3 (± 0.7) dB SNR (Kaandorp et al., 2015). Kaandorp et al. (2015) presented the stimuli with a loudspeaker and, in contrast to our study, they used discontinuous noise with silent periods after each digit-triplet. The relatively slow-acting advanced sound processing features (i.e., noise reduction and adaptive algorithms) incorporated into CI devices need time to become fully active during speech recognition testing. The results of speech recognition tests with discontinuous noise may be affected by these sound processor features.

The test-retest reliability was assessed by means of the SEM, which reflects the agreement between measurements. The SEM values were calculated separately for all conditions, and showed good overall agreement between measurements (Table 2). Most importantly, the SEM values indicate that the test-retest reliability of the home tests is in no way inferior to the test-retest reliability of the measurements taken in the clinic.

The first part of this study concerns the evaluation of potential effects of stimuli presentation modes on speech recognition scores. Contrary to our hypothesis, significantly lower (i.e., better) SRTs (digits-in-noise, adaptive procedure) were observed in the clinic + audio cable condition than in the clinic + loudspeaker condition. In a previous study, De Graaff et al. (2016) found lower (i.e., better) SRTs measured in the audio cable condition with original signals compared to SRTs measured in the audio cable condition with signals that were shaped to match the exact frequency characteristic of the loudspeaker condition exactly. The difference, however, was not significant. This finding does not support the hypothesis of this study that the presentation modes (loudspeaker or audio cable) yield equal results. It is likely that many factors make some individual contribution to the difference in SRT scores assessed with the loudspeaker and those assessed with the audio cable. The acoustics of the soundbooth may have had a slight negative effect, and the aforementioned difference in frequency characteristics may also have had an effect. Larger differences between audio cable input and microphone input could arise between individuals due to head movements and differences in head diffraction. No significant differences were found for different presentation modes for the fixed digits-in-noise test procedure, or for CVC recognition at 65 and 55 dB SPL.

Given the significant difference in the SRT scores obtained with the loudspeaker and those obtained with the audio cable in the present study, the SRT results obtained in the clinic cannot be compared directly to those obtained at home through self-assessment with an audio cable. A reference measurement with the home setup is therefore needed in order to compare the speech recognition assessed at home to the speech recognition assessed in the clinic. This reference measurement should be obtained in the clinic with an audio cable. One benefit of using an audio cable to assess speech recognition in the clinic is that it eliminates the necessity of taking the measurements in soundbooths. Furthermore, the use of an audio cable bypasses the possible negative effects of the environmental characteristics that were identified in the studies by Hughes et al. (2012) and by Goehring et al. (2012). One possible disadvantage of the direct administration of sound through an audio cable is that it is not possible to assess the functionality of the microphone or to demonstrate the advantage of directional microphones. Microphone covers are likely to become dirty, thereby negatively affecting speech recognition in daily life. If an audio cable is used to assess speech recognition, therefore bypassing the microphone, this could result in speech recognition scores that are more favorable than those produced with a loudspeaker. It is thus possible that the speech recognition assessed with an audio cable may not always reflect actual speech recognition in daily life.

The second part of this study concerns the evaluation of potential effects of test format (soundbooth testing by a clinician or self-administered home testing) on speech recognition. No significant differences in speech recognition were identified between these two formats for speech recognition in noise (both adaptive and fixed procedures) or quiet at 65 dB. At 55 dB, however, the results indicate that the home tests generate slight, but significantly better speech recognition scores than did the tests performed at the clinic. This effect was unexpected, particularly in light of the lack of significant differences for any of the other conditions. It is therefore unlikely that the observed difference was caused by differences in test format, test environment or stimuli presentation mode. Two factors that could potentially explain the significant difference are: (1) the list equivalence of the CVC words and (2) learning effects.

The CVC lists were originally created with equal intelligibility for normal hearing individuals (Bosman & Smoorenburg, 1995). However, it is unclear whether the CVC lists are equally intelligible for CI users as well. Exploration of the data revealed that, within the same condition, measurements of speech recognition with different lists systematically yielded significantly different outcomes. In line with findings reported by Bierer et al. (2016) for the consonant-nucleus-consonant word lists, this observation may suggest that the lists are not equally intelligible for CI users.

Another factor that was explored was the occurrence of either procedural learning effects (e.g., effects associated with increasing familiarity with the task, listening environment, and the speaker’s voice) or content learning effects. Results from previous studies have been inconclusive regarding content and procedural learning effects in speech recognition tests. Wilson et al. (2003) found procedural learning effects with the repeated use of sentences to assess speech recognition in quiet. In contrast, Yund and Woods (2010) found limited procedural learning effects, but found that content learning significantly improved speech recognition in noise with the repeated use of sentences. In the present study, participants performed CVC tests that are also used to assess speech recognition ability in the standard care setting. Because we included experienced CI users in our study, the participants were familiar with the task, the listening environment and the speaker’s voice. The likelihood of any procedural learning effects in this study is therefore quite small.

The observed significant improvement in speech recognition in quiet between session 1 and session 3 suggests a content learning effect. In all, 45 lists are available for use with CVC tests, with only 15 of these lists containing unique words. The remaining lists contain the same words as the first 15 lists, but in a different order. In the present study, 24 lists were used, the first 15 of which were unique (see Table 2 for an overview of the lists used for each condition). The lists of words that were used within the home + audio cable condition at

55 dB had already been used in a previous condition, albeit with the words in a different order. Unfortunately, it is not possible to test for actual differences in speech recognition in quiet, because the set of CVC lists used in this study was not counterbalanced across conditions and participants. Nonetheless, further exploration of the data revealed that the repeated use of a combination of words within lists (e.g., lists 1-3 vs. lists 16-18) yielded better speech recognition in quiet, as compared to the use of lists with unique words. This finding persisted independent of test condition and stimulus presentation level. The repeated use of the same words in session 3 might also explain the significantly better speech recognition in quiet obtained in session 3 as compared to session 1.

In summary, the significant improvement in speech recognition in quiet assessed in the 55 dB at home condition could be explained by the fact that the lists were not equally intelligible and/or by content learning effects relating to the CVC lists. Based on these results, it would be advisable either to counterbalance lists across participants or to randomize the order of presenting the lists when using the current set of Dutch CVC lists. This is an important recommendation for future research, as well as for formal audiological testing in the clinical setting. It might be worthwhile to either construct CVC lists that are both phonetically balanced and have equal intelligibility for CI users and normal hearing individuals. Another option could be to omit certain lists when testing CI users. Still, it should be noted that the significant differences in speech recognition in quiet observed between some conditions are small and approximately half the SEM for a single test.

The set of rules that were defined for speech recognition in quiet testing, which allowed specific deviations in the spelling of the target response (e.g., graphemes that represent the same phoneme in Dutch), appeared to return a correct assessment of the majority of typed responses. Upon further investigation, however, a few cases occurred in which the scoring of the typed response by the algorithm was not in accordance with the expected score that a clinician would have recorded if the response had been given verbally. A common substitution occurred with the stimulus ‘tien’ (/tin/), to which participants responded ‘team’ (/tim/). Because the grapheme ‘ea’ was not defined as a valid alternative for the grapheme ‘ie’, the remote assessment tool assessed the response as containing only one correct phoneme, instead of two. Another response that was not scored in accordance with the expected scoring by a clinician was the response ‘gym’ (/ɣɪm/) to the stimulus ‘ging’ (/ɣɪŋ/). In this case as well, the response was assessed as only one correct phoneme instead of two.

As illustrated by the examples presented above, defining rules of exception at both word level (e.g., 'team') and grapheme/phoneme level (e.g., 'i' and 'y') would increase the accuracy of the scoring procedure for typed responses in the CVC task. Given the possibility of an unlimited set of responses, however, slight discrepancies in scoring between the algorithm and a clinician seem inevitable. Even though the scoring by clinicians is considered the gold standard, it is plausible that there are differences in scoring between clinicians as well (e.g., due to a certain extent of subjectivity in the auditory assessment of verbal responses).

We are currently conducting a clinical study in which the speech recognition tests are offered to newly implanted CI users in a home environment, in order to monitor the progression in speech recognition closely during the first few months after cochlear implantation. The results of the speech recognition tests are visible to both the patients and clinicians. In the future, the self-administered home tests could be used to gather information prior to the regular clinical appointments of CI users, or to gather additional test results independent of scheduled appointments. The scores obtained could be forwarded to the cochlear implant centres for review, enabling appropriate action in the event of deterioration. This could lead to patients visiting the CI centres only when there is a clinical need. In the near future, digital wireless streaming of stimuli to the processor will become available. This could further increase the ease of self-administered home tests of speech recognition. In addition, self-administered home tests of speech recognition could be combined with remote programming in order to enhance self-care for CI users.

Conclusions

In conclusion, the results of this study indicate that there are no differences in speech recognition in quiet between measurements taken with a loudspeaker and those taken with an audio cable. In noise, the speech recognition scores obtained with the audio cable were significantly better than those obtained with the loudspeaker, but home self-assessment had no significant effect on speech recognition in noise. The results indicate that it is feasible for experienced CI users to perform speech recognition tests in the home environment, in both quiet and noise. Self-administered home testing for adult CI users could be of great use in daily clinical practice as a comparative assessment when a reference measurement in the clinic with the home test setup is available.

Chapter 4

Our Experience With Home Self-Assessment of Speech Recognition in the Care Pathway of 10 Newly-Implanted Adult Cochlear Implant Users

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Under review

Keypoints

- The number of cochlear implant (CI) users has grown rapidly, resulting in an increased workload for CI centres and a need for new and innovative ways to provide healthcare to users of a CI.

- A telehealth application was developed with a functionality to self-administer speech recognition tests at home, which was evaluated in 10 newly-implanted patients.

- Speech recognition in quiet and in noise improved steadily during the first few weeks of rehabilitation, after which it stabilized.

- The home tests provided a good alternative to testing in the clinic for newly-implanted patients who were able and willing to perform part of their CI care from home, and felt confident in using the technology required.

- Frequently administered speech recognition self-tests provide fine-grained progress details which enable clinicians to monitor their CI user's speech recognition ability over time without the need for users of a CI to visit the clinic.

Introduction

The number of newly-implanted cochlear implant (CI) patients is increasing rapidly, due to changing regulations, expanding candidacy criteria, and technical improvements in CIs. This results in an increased workload for CI centres, and opens the door for new and innovative ways to provide healthcare to users of a CI. Rehabilitation after cochlear implantation is very demanding and time consuming for newly-implanted patients. It requires frequent and long visits to the clinic within the first year after implantation. The sound processor is fitted or fine-tuned during these visits and auditory training is provided. Counselling on how to use and maintain the CI is provided as well as speech recognition testing. Speech recognition is an important outcome measure during rehabilitation and is typically assessed in the clinic by a clinician with calibrated equipment.

Within the 'Supporting Hearing in Elderly Citizens' project, we developed a telehealth application, the MyHearingApp. The application comprises a user interface for a tablet computer with, among other functionalities, a functionality to self-administer speech recognition tests at home. We demonstrated that experienced users of a CI were able to perform self-administered speech recognition tests at home and that the home tests provide a valid alternative to testing in the clinic (De Graaff et al., 2018; De Graaff et al., 2016). The MyHearingApp usability was assessed and observed to be satisfactory in a random group of 16 senior (60+) experienced users of a CI. They ranked the ability to perform home tests as the most relevant functionality (Philips et al., 2018).

The main objective of the current study was to investigate the use and feasibility of the MyHearingApp self-test functionality in care-as-usual of newly-implanted patients. We evaluated whether newly-implanted patients would comply with instructions to repeatedly perform speech recognition tests at home and we collected their experiences with the self-test. Another objective was to describe the progress in speech recognition performance during the first three months of rehabilitation in a more fine-grained manner than in current rehabilitation care.

Materials and Methods

Ethical considerations

The study was approved by the Medical Ethics Committee of VU University Medical Centre Amsterdam. The participants enrolled into the study voluntarily and provided informed consent.

Study participants

Ten consecutive newly-implanted adult patients (7 males; 3 females) participated in this study (Table 1). They were postlingually deaf (onset of severe hearing impairment after the age of seven years), were unilaterally implanted with the Cochlear™ Nucleus® CI24RE implant with Contour Advance Electrode and used the Cochlear™ Nucleus® CP910 sound processor. No selection criteria were set in terms of computer experience.

Table 1. Demographic characteristics, total number of scheduled speech recognition tests and number of tests performed by each participant.

Participant	Gender	Age (years)	Total number of scheduled tests[a]	Tests performed	
				Speech recognition in quiet	Speech recognition in noise
S1	F	77	22	17 (77%)	17 (77%)
S2	M	64	22	19 (86%)	19 (86%)
S3	M	78	24	23 (96%)	23 (96%)
S4[b]	M	78	-	-	-
S5[b]	M	74	-	-	-
S6	M	67	22	20 (90%)	20 (90%)
S7	F	33	20	14 (70%)	15 (75%)
S8	M	49	20	8 (40%)	8 (40%)
S9	M	67	22	19 (86%)	18 (82%)
S10	F	20	22	5 (23%)	5 (23%)
Total			174	125 (72%)	125 (72%)

F, female; M, male

[a]The total number of scheduled speech recognition tests for each participant is different, because of differences in the rehabilitation schedule; [b]S4 and S5 withdrew from the study prematurely.

Procedures

The study was conducted according to a prospective within-subject design in conjunction with care-as-usual. Self-tests were done using a tablet computer (Lenovo Thinkpad 10, Lenovo) and an audio cable that directly presented stimuli to the sound processor. The mixing ratio of the sound processor was set to ‘accessory only’, ensuring that participants only received sound coming from the audio cable. The participants received the tablet computer with the MyHearingApp in the week after activation of their sound processor. Participants were instructed to assess their speech recognition at home twice weekly during the first three months of rehabilitation. The tests were scheduled by a clinician and subsequently appeared in the task list in the MyHearingApp. Participants were allowed to perform the tests on different days, as long as they were performed prior to the next scheduled test. Otherwise, tests were no longer accessible to the participant.

After three months, participants returned the tablet computer at their regular visit to the clinic and were asked to complete a questionnaire to elaborate on their experiences with the self-test functionality (see Table 2 for details). Speech recognition was assessed again in the clinic with the tablet computer after six months of rehabilitation.

Speech recognition tests

Speech recognition in quiet was assessed using monosyllabic words with a consonant-vowel-consonant (CVC) structure, pronounced by a female Dutch speaker (Bosman & Smoorenburg, 1995). Lists of 12 CVC words, each word containing three phonemes, were presented at 65 dB SPL. The score was calculated as the percentage of phonemes recognized correctly. The response on the first word was not included in the calculation of the score.

Speech recognition in noise was assessed with the digits-in-noise test (Kaandorp et al., 2015; Smits, Goverts, and Festen, 2013). The test estimates the speech reception threshold (SRT) via an adaptive procedure using series of digit-triplets (e.g., 6-5-2) presented against a background of continuous steady-state speech-shaped masking noise. The SRT represents the signal-to-noise ratio (SNR) at which the listener recognizes 50% of the digit-triplets correctly. Lower SRTs represent better speech recognition in noise.

At the start, two CVC tests were listed in the task list. Once the participant reached an average phoneme-correct score of at least 40% on the two CVC tests, they were instructed to perform one digits-in-noise test as well. Hereafter, participants performed two CVC tests and one digits-in-noise test. The task list took less than 10 minutes to complete.

Test results were made visible in the MyHearingApp for the CI user in two separate graphs, for speech recognition in quiet and in noise. The graphs showed new and previous results (Appendix A). For speech recognition in quiet, the mean score of two CVC tests with the standard deviation was plotted. For speech recognition in noise, the SRT with standard error of measurement (1.1 dB, based on previous research (Kaandorp et al., 2015)) was plotted. As SRTs may be difficult to interpret for lay people, we opted for an interval scale with categories varying from < -7.5 to > 7.5 dB SNR (Appendix A). The detailed test results were remotely visible for clinicians.

Results

Feasibility and Compliance

One participant withdrew after two weeks (S5) and another one after two months (S4). Both withdrew because they felt that they had insufficient computer knowledge and skills to perform tests at home. The remaining eight participants performed approximately 75% of the scheduled tests (Table 1). Two participants performed less than half of the scheduled tests, because of time constraints and lack of motivation (S8) or fatigue and tinnitus (S10).

Speech recognition

The speech recognition test results are shown in Figure 1. The dashed lines show exponential fits to the data points. All participants, except two (S1 and S7), showed a clear steady increase in speech recognition in quiet and in noise during the first 4-5 weeks. Thereafter, speech recognition stabilized. Half of the participants (S3, S6, S7, S9) reached a score of 80% of phonemes correct or higher three months after activation. One participant (S7) even reached 100% speech recognition in quiet in week 5.

Questionnaire

Responses to the questionnaire are listed in Table 2. Overall, participants gave a mean score of 8 for the possibility to self-assess their speech recognition. Half of participants reported to prefer home testing over testing in the clinic. Presentation of the test results was considered useful and clear, and helped to motivate participants to improve their performance further. Most of the participants considered the results reliable, except one (S1), mainly due to variation in her test results (Figure 1).

Table 2. Specific questions from the questionnaire addressing the home self-assessment of speech recognition with the individual and mean scores.

	Questions	Scores								
		S1	S2	S3	S6	S7	S8	S9	S10	Mean
1	How clear do you find the graph with the speech recognition in quiet results? *Very unclear (0) - Very clear (10)*	10	8	10	8	9	7	8	5	8.1
2	How clear do you find the graph with the speech recognition in noise results? *Very unclear (0) - Very clear (10)*	10	8	10	6	8	7	8	7	8.0
3	How useful do you think it is to see the progress in speech recognition over time? *Not useful (0) - Very useful (10)*	10	8	9	8	10	9	9	6	8.6
4	How reliable do you think the results of the home tests are compared to the tests in the clinic? *Very unreliable (0) - Very reliable (10)*	6	7	7	8	9	8	8	7	7.5
5	How useful do you think it is to be able to self-assess your speech recognition at home? *Not useful (0) - Very useful (10)*	6	9	10	8	10	10	8	7	8.3
6	What do you prefer for speech recognition testing in the future? *Always at home (1) - Frequently at home, sometimes in the clinic (2) - Frequently in the clinic, sometimes at home (3) - Always in the clinic (4)*	2	2	2	1	3	4	3	3	

4

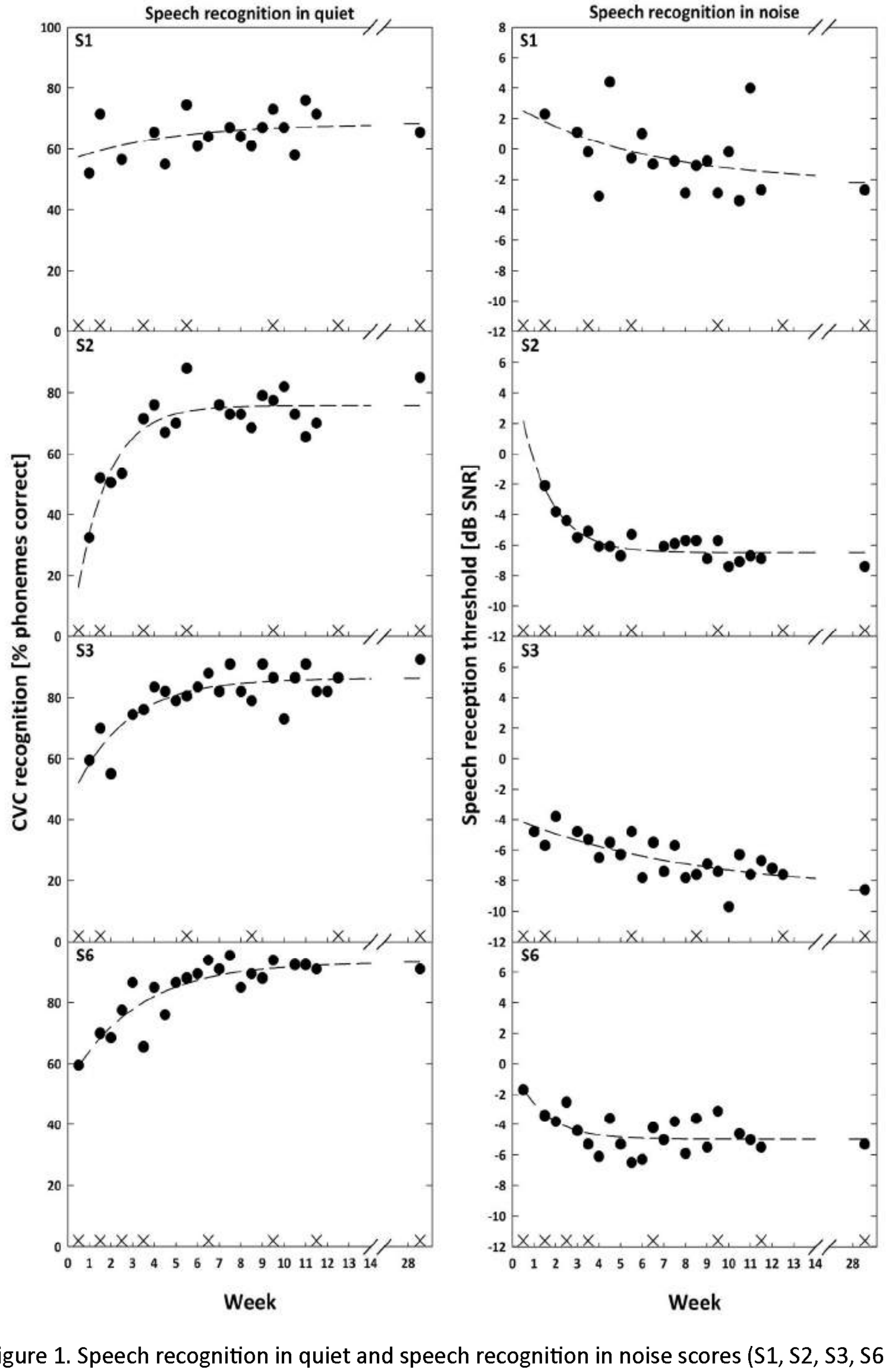

Figure 1. Speech recognition in quiet and speech recognition in noise scores (S1, S2, S3, S6), assessed twice a week during the first three months of rehabilitation and again after six months. The crosses on the x-axes represent fitting appointments with the audiologist in the clinic. The dashed lines show exponential fits to the data points.

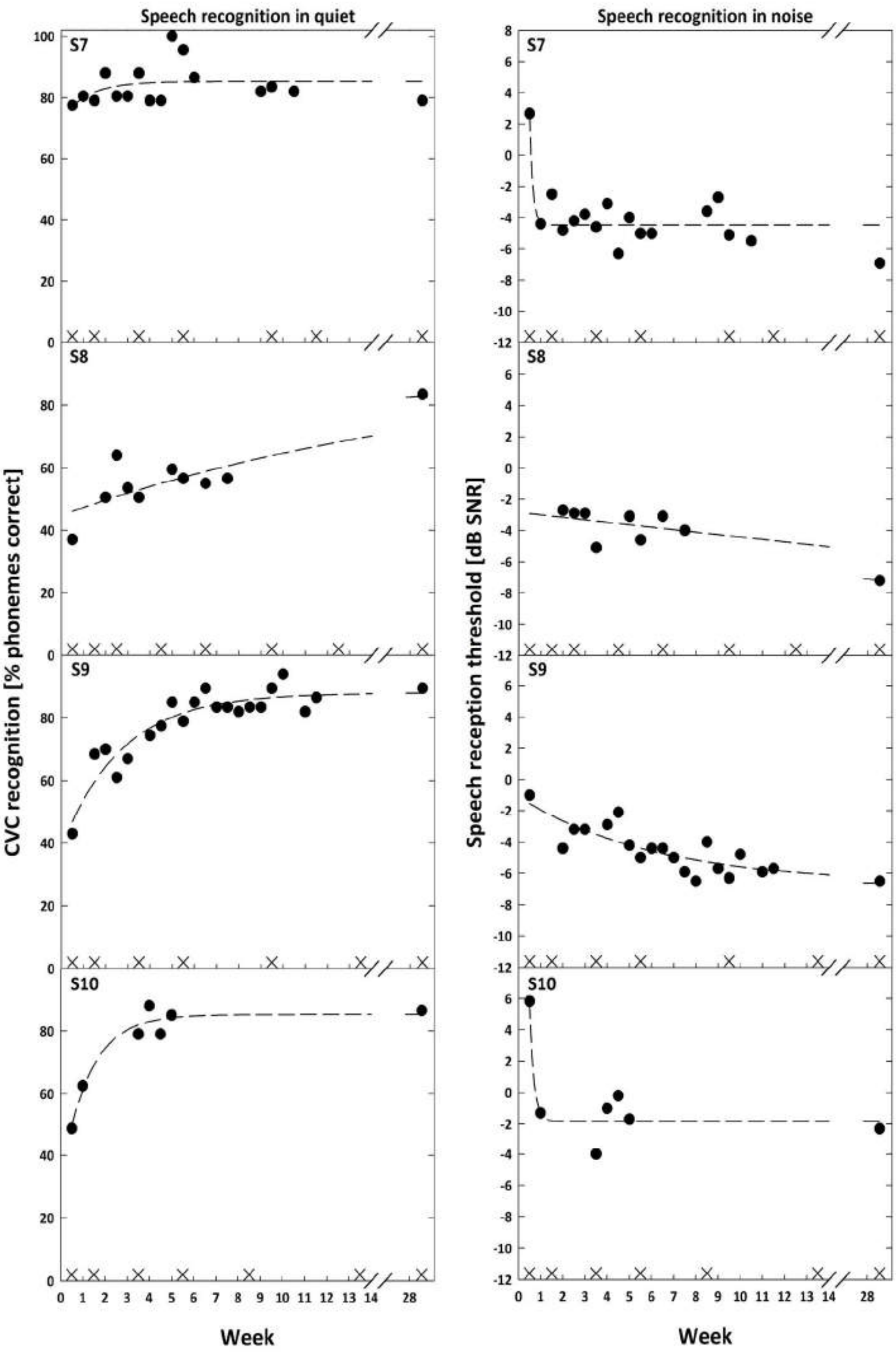

Figure 1 continued. Speech recognition in quiet and speech recognition in noise scores (S7-S10), assessed twice a week during the first three months of rehabilitation and again after six months. The crosses on the x-axes represent fitting appointments with the audiologist in the clinic. The dashed lines show exponential fits to the data points.

4

Discussion

The purpose of this study was to evaluate the use and feasibility of self-administered home speech recognition testing in newly-implanted patients. The results show that home tests might be suitable for approximately 80% of newly-implanted patients. Because we included ten consecutive newly-implanted patients without any selection criteria, we think that our participants are representative of newly-implanted patients, despite the small number. In current care, patients visit our clinic nine times during the first three months for a total of 19 hours. For those patients eligible for home self-assessment, a large reduction in visits and time seems achievable while information about speech recognition progress is even more detailed for clinicians than currently.

Clinical applicability

To our knowledge, this is the first study showing progress in speech recognition over the first three months of rehabilitation in such detail. The data reveal improvements in speech recognition over time, without a clear relation to fitting appointments with an audiologist. Detailed progress information has not been available to clinicians before, because in care-as-usual speech recognition is assessed only once or twice in the equivalent period. Detailed progress information enables clinicians to identify -at a much earlier stage- for whom the level of auditory training would need to be intensified (e.g. for S1 with unsatisfactory progress in the first weeks). The detailed results also indicate for whom visits to the clinic would become unnecessary or could be reduced (e.g. for S2, S3, S6, S9 for whom progress was satisfactory). The current data are promising and indicate that there is more potential for home self-assessment. It could be used to try out different settings of the sound processor at home while simultaneously allowing clinicians to examine effects on speech recognition without the need for additional visits to the clinic. To illustrate, the sound processor could be programmed with two different programs and the CI user could perform multiple tests at home while acclimatizing to the new settings.

Conclusion

In conclusion, home self-testing has the potential to change and improve the CI care pathway substantially, eventually leading to a significant reduction in the required number of visits of patients to the clinic. Not only would this result in cost- and time savings for both clinics and patients, it would even improve the quality and richness of data obtained during rehabilitation. A reduction in the number of technical operations (e.g. digital streaming of stimuli) might improve the usability of the home tests for even more newly-implanted patients.

Chapter 5

Binaural Speech Recognition in Steady-State Noise and Interrupted Noise in Bimodal and Bilateral Cochlear Implant Users

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Under review

Abstract

Hypothesis: The addition of a hearing aid (HA, bimodal), or a second cochlear implant (CI, bilateral), in the contralateral ear can provide CI users with some of the benefits associated with having two ears.

Background: Candidacy criteria for CI have changed considerably over the years. CI users with residual hearing in the contralateral ear regularly opt to wear a HA. Furthermore, bilateral CI is being considered more often. Although the number of bimodal and bilateral CI users is increasing, assessment of binaural hearing is not always part of the standard test battery.

Methods: Ten bimodal (29-84 years) and 10 bilateral CI users (20-91 years) participated in the study. Speech recognition in noise was assessed with the digits-in-noise test in different conditions (monaural, diotic and dichotic) with different masking noises (steady-state and interrupted), to assess binaural benefit, binaural unmasking, and fluctuating masker benefit.

Results: There were no significant differences between monaural and binaural speech recognition for bilateral and bimodal CI users. Speech recognition with CI alone was significantly better than with HA alone for bimodal CI users. Speech recognition was not significantly different between the diotic and dichotic conditions. Speech recognition in noise was significantly better with interrupted noise than with steady-state masking noise for both bilateral and bimodal CI users.

Conclusions: Binaural speech recognition was not significantly better than monaural speech recognition. There was no binaural unmasking, but bilateral and bimodal CI users experienced a fluctuating masker benefit.

Keywords: Cochlear implant, speech recognition in noise, bimodal, bilateral, binaural benefit, binaural unmasking, fluctuating masker benefit.

Introduction

Candidacy criteria for cochlear implantation have changed considerably over the years, mainly due to technical improvements and the success of cochlear implantation. Consequently, the number of people eligible to receive a cochlear implant (CI) is increasing, including people with residual hearing in one or both ears (Leigh et al., 2016; Snel-Bongers et al., 2018). These CI users regularly opt to wear a contralateral hearing aid (HA) to obtain the benefits of bimodal hearing (Devocht et al., 2015; van Loon et al., 2017). In addition, the attitude towards those with bilateral severe-to-profound hearing loss is also changing. Bilateral implantation is already considered the standard of care for prelingually deafened children, and is being considered for adults as well. The addition of a HA in the contralateral non-implanted ear (bimodal) or a second CI (bilateral) can provide CI users with some of the benefits associated with having two ears, such as sound localization and the better ear effect, experienced by normal-hearing listeners (see van Hoesel (2012) for an overview, (van Loon et al., 2014)). But bimodal and bilateral CI users benefit less from other binaural effects, such as squelch or binaural unmasking, than normal hearing listeners (Gifford et al., 2014; Kokkinakis & Pak, 2014; Schleich, Nopp, and D'Haese, 2004; Sheffield, Schuchman, and Bernstein, 2017).

In recent years it has been demonstrated that the digits-in-noise (DIN) test is a feasible, reliable, and valid test for measuring speech recognition in noise in CI users (Kaandorp et al., 2015; Smits, Goverts, and Festen, 2013). The standard DIN test used in these studies, presents digit-triplets in steady-state noise to estimate the signal-to-noise ratio (SNR) at which a listener correctly recognizes 50% of the digit-triplets. Speech recognition in noise can also be assessed with fluctuating noise (e.g., interrupted noise or multi-talker babble noise). Measurements of speech recognition in fluctuating noise are assumed to better represent daily life communication than measurements in steady-state noise. Normal-hearing listeners show an improvement in performance with fluctuating noise, which is also referred to as fluctuating masker benefit (FMB). They can use the temporarily higher SNRs for so-called "dip listening". Hearing-impaired listeners often have less FMB than normal-hearing listeners (Bernstein & Grant, 2009; Desloge et al., 2010).

Speech recognition in quiet and in noise are often assessed during visits to the implant centre. However, the assessment of binaural hearing is not always part of the standard test battery. One reason for this could be that assessing the possible benefits of bimodal hearing or having bilateral CIs is time consuming and cumbersome. A comprehensive test setup and specialized booths are required to assess binaural benefit with spatially separated speech and noise signals. In the standard test setup, stimuli are presented via a loudspeaker in a sound-treated booth and a clinician judges the correctness of the response. In previous studies we demonstrated the use of self-administered tests outside the sound booth to assess speech recognition in experienced CI users (De Graaff et al., 2018; De Graaff et al., 2016). An audio cable was used as a direct coupling between the sound processor and the audio port of a tablet computer (De Graaff et al., 2016). With the direct presentation of stimuli to the CI, possible influences of background noise and reverberation are eliminated. The setup was calibrated and the outcomes of the self-administered tests in quiet and in noise were compared to the outcomes of the standard tests in the clinic (De Graaff et al., 2018). It was demonstrated that the self-administered tests are a viable alternative to tests in the clinic. However, the setup as used in those studies only allowed assessment of monaural speech recognition with the CI. With the increasing number of bimodal and bilateral CI users, it is important to not only assess monaural speech recognition with the CI, but also to assess the benefit in speech recognition of having both a CI and HA or bilateral CI.

In this study, monaural and binaural speech recognition in noise of bimodal and bilateral CI users was assessed via direct audio input into both CI sound processors (bilateral) or the CI processor and the contralateral HA (bimodal). A test battery similar to the one used in normal-hearing adults by Smits et al. (2016) and children by Koopmans, Goverts, and Smits (2018) was used to measure monaural, diotic (i.e., no inter-aural differences), and dichotic (i.e., inter-aural phase difference) speech recognition in steady-state noise and interrupted noise. The aims were (1) to determine the binaural benefit, (2) to investigate the presence of binaural unmasking, and (3) to assess the FMB in bimodal and bilateral CI users.

Materials and Methods

Study participants

Ten bilateral CI users (3 females, 7 males; 20-91 years of age) and 10 bimodal CI users (6 females, 4 males; 29-84 years of age) participated in this study. All participants were native English speakers and had a Cochlear™ sound processor. Both the bilateral and bimodal CI users had to have at least six months of experience with their CI(s) and must have scored at least 50% phonemes correct with their CI on a speech recognition in quiet test which was assessed at a recent visit to the clinic. Participants enrolled in the study voluntarily and provided informed consent at the beginning of the study. The study was approved by the Human Research Ethics Committee of The University of Western Australia. Table 1 lists the demographic characteristics of the participants.

□□□ter□□□□users

Bilateral CI users were fitted with CP910 sound processors for the purpose of the testing if they normally used another sound processor (e.g., CP810 or CP920). Two monaural audio cables were connected to the accessory socket of both CIs, and the accessory mixing ratio of both CIs was set to accessory only.

□□□ od□□□□users

All bimodal CI users were fitted with the ReSound ENZO^2 (EN998-DW, GN Hearing) HA prior to testing. This was done because not all HAs have a direct audio input (DAI), and to ensure that the stimuli were presented at predefined levels. The fitting was done by a HA audiologist according to the NAL-NL2 prescription rule. Pure-tone audiometry was repeated prior to the fitting if audiometry was performed more than six months before. The audiogram was used as a first fit, and target measurements were verified using Real-Ear Measurements (REMs) at 65 dB SPL. Fine-tuning of the first fit based on REMs were done using the ReSound Smart Fit software (GN Hearing). The HA was programmed with two programs, of which program 1 was a regular program and program 2 was specifically set to the DAI. This program was set with the default DAI settings (i.e., 3 dB advantage to the DAI signal). CI users were fitted with a CP910 sound processor if they used another sound processor (e.g., CP810 or CP920). The accessory mixing ratio of the CI was set to accessory only. The mean pure-tone thresholds of the HA ear in bimodal CI users are shown in Figure 1.

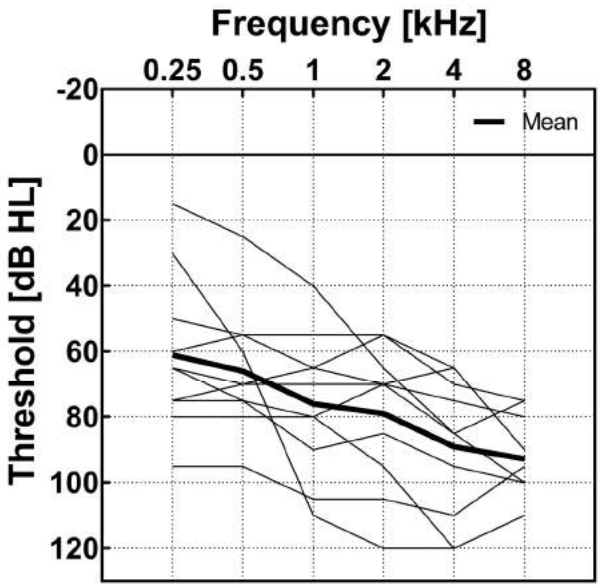

Figure 1. Pure-tone thresholds of the HA ear of 10 bimodal participants. The thick line represents the mean pure-tone thresholds.

Table 1. Demographic characteristics of bilateral AND bimodal CI users.

Bilateral					Bimodal					
Subject	Sex	Age (years)	Implant	Processor	Subject	Sex	Age (years)	CI	Implant	Processor
BIL01	M	71	CI24RECA	CP910	BIM01	M	76	R	CI24RECA	CP810
BIL02	M	77	CI422 CI512	CP810 CP920	BIM02	F	29	L	CI24RECA	CP910
BIL03	F	60	CI512 CI24RECA	CP920	BIM03	M	61	R	CI24RECA	CP910
BIL04	M	47	CI24RECA	Kanso	BIM04	M	76	R	CI24REST	CP920
BIL05	M	43	CI512 CI24RECA	CP810	BIM05	F	61	L	CI24RECA	CP910
BIL06	F	50	CI24REST CI422	CP910	BIM06	M	44	R	CI512	CP810
BIL07	M	73	CI24RECA CI24REST	CP910	BIM07	F	58	L	CI24RECA	CP910
BIL08	M	91	CI24RECA	Kanso	BIM08	F	84	L	CI512	CP920
BIL09	M	81	CI24RECA CI512	CP910 CP810	BIM09	F	80	L	CI24RECA	Freedom
BIL10	F	20	CI24R CI24RECA	CP810 CP910	BIM10	F	78	L	CI24RECA	CP910

BIL, bilateral; BIM, bimodal; F, female; M, male; R, right; L, left.

Digits-in-noise test

The digits-in-noise test used a validated set of Australian-English digit-triplets. Series of randomly chosen digit-triplets (e.g., 6-5-2) were presented in background noise to estimate the speech reception threshold (SRT). The SRT represents the SNR at which the listener correctly recognizes 50% of the triplets. In the current study steady state noise and 16-Hz interrupted noise with a modulation depth of 100% was used. The two noise types had the same spectrum and the noise was presented continuously throughout the test (De Graaff et al., 2016). Continuous presentation of background noise was required as the relatively slow-acting advanced sound processing features of the CP910 CI and ReSound ENZO^2 HA need time to fully activate during speech recognition testing (De Graaff et al., 2016). The overall presentation level was fixed at 65 dB SPL and the initial SNR was 0 dB. An adaptive procedure was used, in which the SNR of each digit-triplet depended on the correctness of the response on the previous digit-triplet, with the subsequent digit-triplet presented at a 2 dB higher SNR after an incorrect response, and at a 2 dB lower SNR after a correct response. This was the same test procedure as that used by De Graaff et al. (2016). This test procedure was specifically designed for use in CI users in that it included two dummy digit-triplets with a fixed SNR of +6 and +2 dB which preceded the actual test. These dummy triplets were used to ensure that the noise reduction algorithms of the CI sound processor had settled, while maintaining the attention of the participant. Each test consisted of a total of 26 digit-triplets. The SNR of the virtual 27th digit-triplet was based on the SNR and correctness of the response of the 26th digit-triplet. The SRT was calculated as the average SNR of digit-triplets 7 to 27.

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Procedures

The study was completed within one session and followed a within-subject repeated measures design. The test battery per participant consisted of two tests (test, retest), with two different masking noises (steady-state and interrupted noise), in four conditions (monaural [left CI and right CI or CI and HA], diotic, dichotic). The test battery was preceded by two practice tests (diotic, one with steady-state and one with interrupted masking noise). Thus, participants performed a total of 18 DIN tests: 2 practice tests + (2 tests x 2 noise types x 4 conditions). The order of the tests in the test battery was counterbalanced across participants, and half of the participants started with steady-state noise. The participants were seated in a sound-treated booth at the Ear Science Clinic, Subiaco, Australia and were instructed to repeat the digit-triplets to the experimenter who entered the responses on a tablet computer.

The setup from the study by De Graaff et al. (2018) and De Graaff et al. (2016) was used. In short, the tests were administered with a tablet computer (Lenovo Thinkpad 10, Lenovo), using software developed by Cochlear Technology Centre (Mechelen, Belgium). The stimuli were directly presented to the CI sound processor via an audio cable. The setup was calibrated, such that the signals presented through the audio cable were delivered at an intensity equal to the intensity of an acoustic signal of 65 dB SPL (De Graaff et al., 2016). For the binaural measurements, a splitter cable was used which had a stereo jack at one end that was connected to the tablet computer, and two mono sockets at the other end which were connected via an audio cable to the HA and CI or bilateral CI.

Conditions

Speech recognition was assessed monaurally (left and right ear) and binaurally (diotic and dichotic). For the diotic condition, the noise and speech were identical for the two ears (S0N0). For the dichotic condition, the noise was identical for the two ears while the speech signal presented to one of the two ears was phase inverted (SπN0). The conditions assessed in the bimodal CI users were HA only, CI only and CI + HA (diotic and dichotic). The conditions measured in the bilateral CI users were left CI only, right CI only and CI + CI (diotic and dichotic). During monaural speech recognition testing, the audio cable of the contralateral ear (either CI or HA) was disconnected.

Statistical analyses

All statistical analyses were conducted using SPSS (IBM SPSS, version 20). The data were analyzed with repeated measures analysis of variance (ANOVA). For bilateral CI users, an ANOVA was conducted to compare the effects of noise type (steady-state noise vs. interrupted noise) and condition (left CI, right CI, diotic, dichotic). For bimodal CI users, an ANOVA was conducted to compare the effects of noise type (steady-state noise vs. interrupted noise) and condition (HA, CI, diotic, dichotic). If sphericity assumptions were violated, the Greenhouse-Geisser correction was used. A p value of < 0.05 was considered statistically significant. Post hoc tests with Bonferroni correction for multiple comparisons were conducted in case of significant effects. The standard error of the mean (SEM), which reflects the agreement between measurements, was calculated for the different conditions.

Data was missing for both the steady-state and interrupted masking noise for two bilateral CI users in the dichotic condition and for one bilateral CI user in the right CI condition. In addition, the SRT was excluded for one bilateral CI user in the left CI condition with interrupted masking noise due to an unreliable measurement (SRT = 9.1 dB SNR, SD = 10.6). Therefore, the statistical analyses for bilateral CI users were done on six complete cases. For bimodal CI users, data with the HA was missing of two participants, because these bimodal CI users had no speech recognition at all with their HA. Therefore, statistical analyses for the bimodal CI users were done on eight complete cases.

Table 2. Mean SRT for the DIN tests (mean SRT for complete cases in brackets) and standard error of measurement (SEM) measured in different conditions and with different masking noises.

		Steady state noise		Interrupted noise	
		SRT [dB SNR] (complete cases)	SEM [dB]	SRT [dB SNR] (complete cases)	SEM [dB]
Bilateral	Left CI	-7.1 (-7.2)	1.3	-12.5 (-13.2)	1.1
	Right CI	-7.7 (-7.8)		-12.8 (-13.3)	
	Diotic	-8.9 (-9.0)		-14.5 (-14.6)	
	Dichotic	-9.5 (-9.7)		-15.2 (-14.6)	
Bimodal	HA	-3.2 (-3.2)	1.7	-4.3 (-4.3)	1.4
	CI	-8.6 (-8.5)		-13.8 (-13.2)	
	Diotic	-9.0 (-9.1)		-13.7 (-13.2)	
	Dichotic	-9.4 (-9.5)		-13.6 (-13.0)	

Results

The SEMs for different conditions are listed in Table 2, and the test-retest SRTs are shown in Figure 2. The SEM values for the digits-in-noise tests with bimodal and bilateral CI users in different conditions is similar to the values observed in Kaandorp et al. (2015) and De Graaff et al. (2018). Table 2 and Figure 3 show the mean SRTs for the different conditions and masking noises for all participants and for the subset of complete cases.

Bilateral CI users

The ANOVA showed a significant main effect of noise ($F[1,5] = 88.587$, $p < 0.01$). The tests with interrupted noise resulted in significantly lower (i.e., better) SRTs than tests with the steady-state noise. No significant main effect for condition and no significant interaction between noise and condition were found.

Thus, bilateral CI users benefit from interruptions in noise which yields an average FMB of 5.5 dB. There was no binaural benefit nor binaural unmasking.

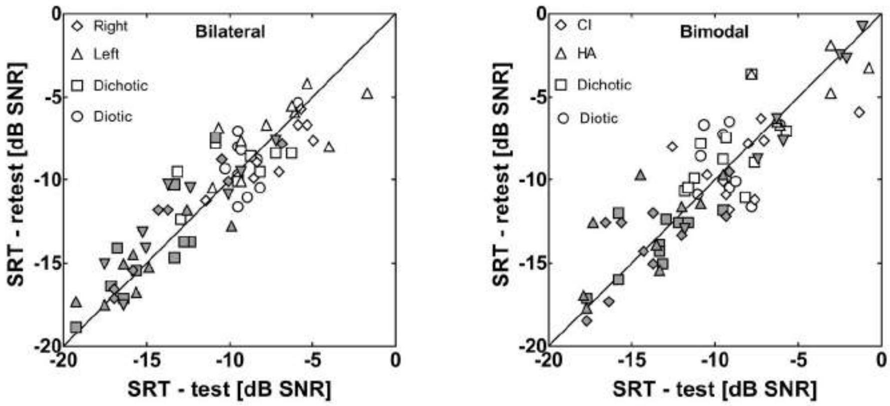

Figure 2. Scatterplots of the test-retest mean SRTs of the bilateral CI users (left) and bimodal CI users (right). The white symbols represent SRTs in continuous noise, and the grey symbols represent SRTs in interrupted noise. The different symbols represent different test conditions. The line indicates equal test-retest SRTs.

Bimodal CI users

The ANOVA showed a significant main effect of noise ($F[1,7] = 40.298$, $p < 0.01$), and condition ($F[1.124,7.866] = 32.209$, $p < 0.01$) on mean SRT. Because the interaction between noise type and condition was significant ($F[3,21] = 9.568$, $p < 0.01$), separate repeated measures ANOVAs were conducted for steady-state and interrupted noise. For steady-state noise, a significant effect for condition ($F[1.101,7.708] = 24.940$, $p < 0.01$) was found. Post hoc comparisons with Bonferroni corrections showed significant differences between the HA alone and all other conditions ($p < 0.05$), with poorer SRTs for the HA alone condition. There were no significant differences between the other conditions. For interrupted noise, a significant effect for condition ($F[1.171,8.200] = 30.081$, $p < 0.01$) was found. Post hoc comparisons showed significant differences between the HA alone and all other conditions ($p < 0.01$). There were no significant differences between the other conditions.

Thus, bimodal CI users had a significant FMB of, on average, 3.3 dB. Monaural speech recognition in noise was significantly better with their CI alone than with their HA. The HA did not contribute to binaural speech recognition. Similar to bilateral CI users, bimodal CI users did not demonstrate binaural unmasking.

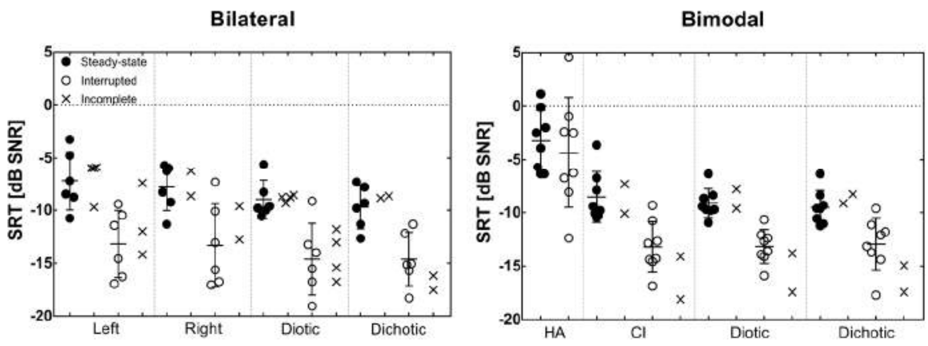

Figure 3. Speech recognition in noise for bilateral (left panel) and bimodal (right panel) CI users. The SRT was measured in four different conditions and with two different masking noises. The SRTs measured with steady-state noise are represented by the black symbols, the SRTs measured with interrupted noise are represented by the white symbols. The symbols represent individual scores and the horizontal lines represent mean and ± 1 standard deviations of all complete cases (n = 6 for bilateral CI users and n = 8 for bimodal CI users). The crosses indicate data of the participants who did not perform all the tests (incomplete cases).

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Discussion

The aims of this study were to investigate the binaural benefit, presence of binaural unmasking, and the FMB in bilateral and bimodal CI users. An Australian English DIN test was used to assess monaural and binaural speech recognition in noise with direct audio input to both CIs for bilateral CI users and the CI and HA for bimodal CI users in different conditions with different masking noises.

Binaural benefit

For bimodal CI users, it was shown that monaural speech recognition with the CI was significantly better than with the HA, but binaural speech recognition was not significantly better than monaural speech recognition with the CI. This means that the HA in the contralateral ear did not help to improve speech recognition in noise. For bilateral CI users no significant difference between left and right CI performance was found. Because it is more likely to find differences in speech recognition between first-implanted CI and second-implanted CI than between left and right CI, a paired samples t-test was performed. For the participants in the current study we did not find significant differences between monaural speech recognition with the first-implanted CI and the second-implanted CI for both steady-state ($t[6] = -1.246$, $p = 0.259$) and interrupted noise ($t[5] = 0.309$, $p = 0.770$). We also did not find better speech recognition with both CIs compared to speech recognition with only one CI.

In contrast, normal-hearing listeners experienced a small but significant benefit (between 0.5 and 0.8 dB) from the binaural condition compared to the monaural condition for the Dutch and American English DIN test (Smits et al., 2016). The findings in the literature on the binaural benefit of having a contralateral HA or a second CI are somewhat contradictive (van Hoesel, 2012). For bilateral CI users, no significant differences between left, right or bilateral CI use were found when signal and noise (i.e., speech-shaped noise or interfering talkers) were both presented from the front (Laske et al., 2009; Rana et al., 2017; van Hoesel & Tyler, 2003). But other studies found that some bilateral CI users experience binaural benefit (Schleich, Nopp, and D'Haese, 2004; Tyler et al., 2002). For bimodal CI users, Morera et al. (2005) did not find a binaural benefit when using four-talker babble and a fixed SNR. Schafer et al. (2011) conducted a meta-analysis to evaluate the findings on binaural advantages in bilateral and bimodal CI users. The authors concluded that the addition of a contralateral CI or HA significantly improved speech recognition in noise. Kokkinakis and Pak (2014) also found a binaural benefit, both in bimodal and bilateral CI users, when speech was presented in four-talker babble.

Binaural unmasking

Binaural unmasking was assessed by comparing diotic (same speech and noise signal, S0N0) and dichotic (phase inverted speech signal and same noise signal, SπN0) speech recognition. The inter-aural phase difference introduced in the speech signal was found to have no effect on speech recognition for bilateral and bimodal CI users. In normal-hearing listeners, binaural unmasking results in an improvement in speech recognition of 5.5-6.0 dB (Smits et al., 2016). The finding of the current study is in line with previous research, where Sheffield et al. (2017) assessed speech recognition in noise in diotic and dichotic conditions pre- and post-implantation in bimodal CI users. They found that approximately half of the participants experienced binaural unmasking preoperatively, however, the majority of bimodal CI users had no binaural unmasking postoperatively (Sheffield, Schuchman, and Bernstein, 2017). The effect of binaural unmasking was also investigated by Ching et al. (2005), who created an inter-aural time delay by delaying the noise signal with 700 µs relative to the speech signal. An absence of binaural unmasking in bimodal CI users was reported (Ching et al., 2005). One of the widely accepted reasons for the lack of binaural unmasking is the absence of, or poorly encoded, inter-aural phase and time information in the signal processing pathway of CIs (van Hoesel, 2012).

Fluctuating masker benefit

DIN tests were performed with steady-state and interrupted masking noise to investigate whether bimodal and bilateral CI users could benefit from interruptions in noise. Both groups demonstrated a significant FMB. It is well known that normal-hearing individuals experience an improvement in speech recognition in fluctuating noise (Rhebergen, Versfeld, and Dreschler, 2008; Smits & Houtgast, 2007; Smits et al., 2016). Hearing-impaired listeners often experience less FMB than normal-hearing listeners (Bernstein & Grant, 2009; Desloge et al., 2010). The findings of the current study, where bimodal and bilateral CI users experience substantial FMB, is in contrast to previous studies. A study by Nelson et al. (2003) showed very little FMB for CI users. Cullington and Zeng (2008) and Stickney and Zeng (2004) found no FMB for different fluctuating maskers in CI users and normal-hearing listeners who listened through an implant simulation. Possible explanations for the discrepancy between our findings and the findings in the literature may be the stimulus presentation mode, the type of fluctuating noise, and the speech material used. In our study, an audio cable was used to directly present stimuli to the sound processor, whereas other studies used loudspeakers to present the stimuli. The use of an audio cable for speech recognition testing diminishes the possible effect of background noise and reverberation in speech recognition testing, which might still be present in a sound-treated booth when stimuli are presented via a loudspeaker. The acoustics of the sound-treated booth might have an effect on the characteristics of the fluctuating noise, thereby reducing the possible FMB in hearing-impaired listeners. Zirn et al. (2016), however, used an audio cable to investigate FMB in a group of high performing CI users, but they found no FMB. Unlike in our study, which uses interrupted noise, they used speech-modulated noise which means that CI users need to segregate speech from noise. In interrupted noise, CI users can use clean speech fragments during noise interruptions for recognition. The large FMB found in this study might also be explained by the speech material that is used in the current study (i.e., digit-triplets). Smits and Festen (2013) showed that the FMB decreases with increases in SNR. Both Kwon et al. (2012) and Zirn et al. (2016) used sentences as speech material. SRTs for sentences are in general higher than SRTs for digit-triplets, which implies less FMB. Another explanation for the large FMB found in our study compared to the FMB reported in other studies might be the relatively good SRTs in steady-state noise for the CI users in the current study (Smits & Festen, 2013). This might indicate that the CI users are relatively high-performing CI users. Both Kwon et al. (2012) and Zirn et al. (2016) showed FMB for part of their high-performing CI users. However, Kwon et al. (2012) created a condition that promoted FMB, because the masker was presented in gaps of the speech signal.

Limitations

The sample size of the current study is relatively small which is a limitation of the study. Another limitation is that the current set of test conditions should be changed or enlarged for further use in studies or in clinic, because they cannot demonstrate the potential benefit of bilateral CI or bimodal CI. It should be noted that bilateral and bimodal CI users will undoubtedly benefit from their second CI or contralateral HA in many real life situations due to the better-ear effect (Litovsky, Parkinson, and Arcaroli, 2009; Schleich, Nopp, and D'Haese, 2004; van Loon et al., 2014). Therefore, a condition which is a realistic representation of listening in free field is needed. In that condition, input to both hearing devices (two CIs or CI and HA) in a situation with noise and speech spatially separated, should be simulated. We expect that the binaural advantage, which could be up to 12 dB for bilateral CI users (van Loon et al., 2014), could be demonstrated with the test setup from the current study as well, when using these stimuli.

Conclusion

In conclusion, speech recognition in noise was assessed with direct audio input into both CIs of bilateral CI users, and the CI and HA for bimodal CI users. When comparing monaural and binaural speech recognition, no binaural benefit was demonstrated for the bilateral and bimodal CI users. Furthermore, no binaural unmasking was present for both bilateral and bimodal CI users when speech recognition in noise was compared for diotic (i.e., without inter-aural phase difference) and dichotic (i.e., with inter-aural phase difference) listening conditions. Both bilateral and bimodal CI users benefitted from interruptions in the masking noise, which yielded a large FMB when speech recognition assessed with steady-state masking noise was compared to interrupted masking noise. The conditions assessed in the current study do not capture all possible benefits from binaural hearing in bilateral and bimodal CI users. Therefore, future speech recognition tests in CI users with direct audio input should incorporate conditions that are representative of daily life communication (e.g., simulated spatially separated speech and noise).

Chapter 6

Predicting Speech Recognition in Quiet and Noise from Fitting Parameters, Impedances and ECAP Thresholds in Adult Cochlear Implant Users

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To be submitted

Abstract

Objectives: The objective of this study was to predict speech recognition in quiet and in noise from fitting parameters, sound-field aided thresholds, electrically evoked compound action potential (ECAP) thresholds and impedances. Ultimately, the objective was to identify parameters which may improve current fitting practices and, thus, optimize speech recognition of cochlear implant (CI) users.

Design: Adult CI users who visited the Amsterdam UMC, location VUmc, for their annual follow-up between January 2015 and December 2017 were retrospectively identified. Several inclusion criteria were applied. The final study population consisted of 138 postlingually deaf Cochlear™ adult CI users. Prediction models were built with speech recognition in quiet and in noise as the outcome measures, and fitting parameters, sound-field aided thresholds, ECAP thresholds and impedances as the independent variables. Separate analyses were performed for postlingually deafened CI users with early onset of severe hearing impairment (EO) and CI users with late onset of severe hearing impairment (LO).

Results: Speech recognition in quiet was not significantly different between groups. Speech recognition in noise was worse for the EO group compared to the LO group. For CI users in the LO group, mean aided thresholds, mean electrical dynamic range (DR) and measures to express the impedance profile across the electrode array were identified as predictors of speech recognition in quiet and in noise. For CI users in the EO group, mean T level appeared to be a significant predictor in the models for speech recognition in quiet and in noise, such that with higher T levels, speech recognition in quiet and in noise worsened.

Conclusions: Significant parameters to predict speech recognition in quiet and in noise were identified. The results of this study may guide audiologists in their fitting practices and improve the performance of CI users. Aided thresholds should be around the target level of 25 dB SPL, and the DR should preferably be between 40-60 CL. Finally, clinicians should be aware of profiles of impedances other than a flat profile with mild variations. Future research should assess the clinical relevance of predictors identified in this study.

Keywords: Cochlear implant, speech recognition, multivariable linear regression, fitting parameters, NRT thresholds, ECAP, impedance.

Introduction

Speech recognition performance varies highly among cochlear implant (CI) users. The variance in speech recognition can still not be fully explained, but many factors potentially contributing to the large variation in outcome have been identified (Blamey et al., 1996; Blamey et al., 2013; Finley and Skinner, 2008; Lazard et al., 2012). Some studies showed that patient characteristics, such as age, duration of deafness, etiology of deafness, and linguistic and cognitive factors partly explain the variance in speech recognition (Blamey et al., 2013; Kaandorp et al., 2017; Lazard et al., 2012). In addition, device and implant factors are assumed to be related to speech recognition outcomes. Examples of these factors include electrode positioning, electrode insertion depth, and the number of inserted or active electrodes (Finley and Skinner, 2008; Lazard et al., 2012; Skinner et al., 2002; Yukawa et al., 2004). Other studies have shown the effect of fitting parameters on speech recognition (Busby and Arora, 2016; Loizou, Dorman and Fitzke, 2000; Skinner et al., 1999; Van der Beek, Briaire and Frijns, 2015). The current study aims to add to previous research by predicting speech recognition in quiet and in noise from fitting parameters, the electrically evoked compound action potential (ECAP) thresholds and impedances in a group of adult Cochlear™ CI users. The rationale for using these parameters is that 1) these parameters are available to the fitting audiologist during a fitting session, and 2) these parameters can be adjusted or may change between fitting sessions. Only Cochlear™ CI users were included, because they form the largest group of adult CI candidates in our CI centre. A large group is needed because the number of variables that can potentially predict speech recognition may be high, even with the above mentioned restrictions.

Fitting of CI processors is essential to achieve optimal speech recognition for CI users. The identification of possible effects of fitting parameters on speech recognition can guide clinicians and to improve fitting practices. For fitting CI sound processors, there is no commonly accepted good clinical practice and there are no fitting rules available equivalent to the prescription rules used in hearing aid fitting (e.g., NAL or DSL prescription rules). For Cochlear™ sound processors, a huge amount of fitting parameters and other measures are available to the clinician during a fitting session. Vaerenberg et al. (2014) conducted a global survey on fitting practices and found considerable differences in fitting practices between CI centres. The authors concluded that CI centres focus on the setting of stimulation levels based on psychophysically derived measures of threshold (i.e., T level for Cochlear™) and comfort (i.e., C level for Cochlear™). Other parameters (e.g., speech coding strategy, pulse width, stimulation rate, gain, Q factor, frequency allocation table, number of maxima) are usually set at default and rarely modified.

An important goal of fitting CI sound processors is to maximize the use of the dynamic range of the auditory nerve by setting T and C levels for each electrode. The electrical dynamic range (DR) is covered by the difference between T and C levels. C levels that are set either too low or too high may have a negative impact on speech recognition and sound quality (Wolfe and Schafer, 2015). Setting T levels too low (i.e., below hearing threshold) results in the inaudibility of soft sounds, while T levels that are set too high will result in ambient sounds that may be too loud (Busby and Arora, 2016). A high variability of T levels across electrodes can negatively impact speech recognition as well, when the variability is due to variations in the electrode-to-neuron distance (Pfingst and Xu, 2004, 2005; Zhou and Pfingst, 2014). Aided sound-field thresholds are often assessed to determine the audibility of soft sounds with targets usually set at 20-30 dB SPL. The aided thresholds are related to the T-SPL (i.e., default at 25 dB SPL), which relates the minimum intensity input level to the electrical stimulation at T level, the microphone sensitivity and T levels. If T levels are set correctly, aided thresholds should be around the target level of 25 dB SPL (i.e., with T-SPL set at default and sensitivity at 12).

Multiple studies have investigated the use of ECAP as an alternative to behavioral parameters in the fitting of adult and pediatric patients (e.g., Botros and Psarros (2010a, 2010b); Brown et al. (2000); Franck and Norton (2001); Hughes et al. (2000); Smoorenburg, Willeboer and van Dijk (2002)). The ECAP represents the response of the auditory nerve after electrical stimulation, and can be measured using the technology incorporated in modern CIs. In CI users with a Cochlear™ implant, the technique is called neural response telemetry (NRT). The NRT uses one intra-cochlear electrode to stimulate the auditory nerve, and another intracochlear electrode to record the neural response. Although the majority of studies only found a weak to moderate correlation between ECAP thresholds and stimulation levels, studies have shown the clinical relevance of ECAP measurements. For instance, the ECAP threshold profile can be used by clinicians to determine stimulation profiles (Botros and Psarros, 2010a; Smoorenburg, Willeboer and van Dijk, 2002).

Electrode functioning is regularly assessed at the beginning of programming sessions by means of electrode impedance. Electrode impedance is a measure of the resistance to electrical current flow across an electrode (Wolfe and Schafer, 2015). Once electrodes are stimulated, the impedances decrease and usually remain stable up to, at least, 24 months after implantation (Hughes et al., 2001). Changes in electrode impedance can indicate changes in the surrounding tissue or electrode function (i.e., short or open circuits) which may negatively affect patient performance.

Objective of current study

The objective of the current study was to identify parameters which may improve current fitting practices and, thus, optimize speech recognition of CI users. Clinical data of postlingually deafened unilaterally implanted Cochlear™ adult CI users were used to build prediction models to predict speech recognition in quiet and in noise in two groups of CI users. Separate analyses were performed for postlingually deafened CI users with early onset of severe hearing impairment (i.e., before the age of seven years) and CI users with late onset of severe hearing impairment, because we speculated that optimal fitting parameters would be different for these groups. Prelingually deafened adults were excluded, because speech recognition is often limited in this group of patients. The prediction models were built with fitting parameters, sound-field aided thresholds and objective measures (i.e., ECAP thresholds and impedances) that are available to the clinician during a fitting session (see above). Thus, other factors (e.g., age, duration of deafness, etiology of deafness, electrode position, cognitive and linguistic abilities) were not included in the models.

Materials and methods

Rehabilitation and general fitting procedures for adult CI users in Amsterdam UMC, location VUmc

Because large variations exist between CI centres on all aspects of fitting (Vaerenberg et al., 2014), we briefly describe the general fitting procedure and rehabilitation program of CI users in our clinic. Although small differences in fitting procedures between audiologists in our CI centre may exist, the fitting procedures used for the patients in the current study can be considered largely similar.

In our CI centre, the rehabilitation program for newly implanted CI users comprises weekly visits to the clinic up to six weeks after initial activation of the sound processor, three visits in the following five months, and annual follow-up visits thereafter. During the first weeks of rehabilitation, emphasis is put on fitting of the sound processor and auditory rehabilitation. Two basic principles guide the fitting, mainly by changing T and C levels. First, we want to use the entire dynamic range of the auditory nerve and, second, soft sounds should be audible. In general, the speech coding strategy and its specific parameters are initially set at default and are rarely modified. In our clinic, the default speech coding strategy is ACE, stimulation mode is MP1+2 (monopolar) with a stimulation rate of 900 Hz, pulse width of 25 µs, 8 maxima, standard frequency allocation table, and a Q factor of 20. In addition, we normally use no channel gain, set the sensitivity at 12 and the volume at 10. Next to fitting of the sound processor, speech recognition performance and aided thresholds are assessed and the outcomes are used for optimization of the fitting.

Each fitting session generally starts by measuring electrode impedances in all four electrode coupling modes to identify open or short circuits. Subsequently, stimulation levels are psychophysically determined. T levels are determined by presenting a stimulus in a descending procedure where CI users are instructed to raise their hand or say "yes" when they hear the stimulus. C levels are determined using a loudness scaling method in which the clinician gradually increases the presentation level of a stimulus. The CI users are asked to indicate their loudness percept by pointing to categories on a 10-point loudness scale. C levels are set at a level that is "loud, but comfortable". C levels are generally assessed on a subset of electrodes, and the levels of intermediate electrodes are then interpolated. Subsequently, all C levels are decreased by a certain percentage of the DR. Then, the sound processor is switched to live speech mode and the clinician increases C levels while the CI user listens to speech and louder sounds to find the user's most comfortable level. Loudness balancing across electrodes is used during some fitting sessions to ensure that the CI user perceives the stimuli to be equally loud. Here, sets of four adjacent electrodes are stimulated at C level using the sweep functionality of Custom Sound®, and individual C levels are adjusted until the CI user reports equal loudness of all four electrodes.

NRT measurements are performed on all electrodes intraoperative, and on a subset of electrodes during some of the fitting sessions in the first year postoperative and at annual visits.

Study population

We retrospectively identified CI users who visited the Amsterdam UMC, location VUmc, for their annual follow-up between January 2015 and December 2017. The data of the most recent annual follow-up were used for CI users who had multiple follow-ups in this time span. Postlingually deafened Cochlear™ CI users that were unilaterally implanted at our CI centre after the age of 18 years and had more than one year CI experience were included. Of this group, CI users with strongly deviating parameters (e.g., speech coding strategy other than ACE and more than three electrodes disabled) were excluded. Our final study population consisted of 138 patients. Implant and processor details are listed in Table 1.

The final study population was split into two groups; one group with postlingually deaf adult patients with early onset of severe hearing impairment (EO group, n = 41). CI users in this group include CI users who were fitted with hearing aids before the age of seven years, or went to a school for the hearing-impaired. The other group consisted of postlingually deaf adult patients with late onset of severe hearing impairment (LO group, n = 97). The mean age at the time of the annual follow-up was 67.9 years (SD = 13.4) and 49.6 years (SD = 13.4) for the LO and EO group, respectively. The mean age at implantation was 62.0 years (SD = 13.4) and 44.0 years (SD = 13.8) years for the LO and EO groups, respectively.

Table 1. Type of implant and sound processor parameters.

		Number of CI users (percentage)
Type of implant	*CI24RECA*	101 (73.2%)
	CI422SRA	5 (3.6%)
	CI24RCS	4 (2.9%)
	CI24RCA	10 (7.2%)
	CI512	18 (13.0%)
Stimulation mode	*Default: MP1+2*	134 (97.1%)
	Else	4 (2.9%)
Channel rate*	*Default: 900 Hz*	130 (94.2%)
	Else	8 (5.8%)
Pulse width*	*Default: 25 µs*	134 (97.1%)
	Else	3 (3.1%)
Number of maxima	*Default: 8*	131 (94.9%)
	Else	7 (5.1%)
Number of disabled electrodes	*0 or 1*	120 (87.0%)
	2 or 3	18 (13.0%)
C-SPL	*Default: 65 dB SPL*	115 (83.3%)
	Else	20 (14.5%)
T-SPL	*Default: 25 dB SPL*	111 (80.4%)
	Else	24 (17.4%)
Volume*	*Default: 10*	81 (58.7%)
	Else	52 (37.7%)
Sensitivity†	*Default: 12*	115 (83.3%)
	Else	20 (14.5%)

* Stimulation levels of CI users with settings other than default for channel rate, pulse width and volume were corrected. † T- and C-SPL of CI users with sensitivity other than default were corrected.

6

Outcome measures

Speech recognition in quiet and in noise (procedures described in the paragraphs below) was assessed in a sound treated booth, where CI users are seated in front of a loudspeaker at a distance of approximately 70 cm. For the purpose of this study, speech recognition in quiet and in noise scores that were assessed with the CI alone were used.

Speech recognition in quiet

Speech recognition in quiet was assessed with monosyllabic words with a consonant-vowel-consonant (CVC) structure, pronounced by a female Dutch speaker (Bosman and Smoorenburg, 1995). CVC words were presented in quiet at 65 dB SPL. Each CVC word consisted of three phonemes, and the score of the CVC test in quiet was calculated as the percentage of phonemes correct. Typically, three lists of 12 words were presented, but occasionally, CI users were presented with less than three lists of CVC words. The mean percentage of phonemes correct of the presented lists (i.e., two or three lists) was calculated, omitting the first CVC word of each list.

Speech recognition in noise

Speech recognition in noise was assessed with the digits-in-noise test (Kaandorp et al., 2015; Smits, Goverts and Festen, 2013). Twenty-four digit-triplets were presented in steady-state speech-shaped noise using an adaptive procedure, with the overall presentation level of target speech and masking noise fixed at 65 dBA. The digits-in-noise test assesses the speech reception threshold (SRT), which is defined as the signal-to-noise ratio (SNR) in dB at which a listener correctly recognizes 50% of the digit-triplets. Typically, two lists of 24 digit-triplets were presented to assess the mean SRT, but occasionally, the SRT was assessed with only one list of digit-triplets. In that case, the SRT assessed with one list was used for the analyses. The SRT was not assessed in four CI users. These CI users were only included in the analyses for speech recognition in quiet.

Independent variables

Speech recognition scores, as well as volume and sensitivity settings, program, and MAP number were registered on a special form with a checklist for the clinician. If the form was not filled in, data were retrieved using electronic patient files, database from the audiometer, and the fitting software Custom Sound®. Independent variables were analyzed as continuous variables, unless stated otherwise.

Fitting parameters

T and C levels depend on stimulation rate, pulse width and volume settings. It is therefore important to correct the T and C levels if the stimulation rate, pulse width and volume settings were not at default at the time of speech recognition assessment. The stimulation levels were converted before the analyses, using formulas acquired through Cochlear™. The correction was applied for 59 out of the 138 CI users. The actual T and C levels, stimulation rate, pulse width and volume setting were used to calculate the corresponding stimulation levels with a volume setting of 10, stimulation rate of 900 Hz and pulse width of 25 µs. If volume settings that were used during speech recognition assessment were not available, C levels could not be converted and were therefore considered as missing data (n = 5). The DR was calculated by subtracting the corrected T levels from the corrected C levels. The corrected T and C levels were used to calculate the DR, because the DR would be different if, for instance, the volume was shifted from the default value to a lower value of 6. The mean, standard deviation (SD) and range (highest minus lowest) of T and C levels and DR were calculated to describe the profile of stimulation levels over 22 electrodes (i.e., profile and variation). In addition, across-site variation (ASV) of T and C levels and DR was determined by calculating the mean absolute values of the differences between the levels at an electrode and the more apical electrode (Pfingst, Xu and Thompson, 2004). T and C levels and DR were not always available for each electrode because of disabled electrodes. In that case, the next available electrode was used to calculate the ASV. All parameters are listed in Table 2.

Aided thresholds, NRT thresholds and impedances

Sound-field aided thresholds at octave frequencies from 125 Hz to 8 kHz were measured with narrow band (1/3 octave) noise stimuli and averaged to obtain the mean aided sound-field threshold (Table 2, audiometry). T-SPL relates the minimum intensity input level to the electrical stimulation at T level. C-SPL relates the maximum intensity input level to the electrical stimulation at C level. Both T- and C-SPL depend on the sensitivity setting, and were therefore corrected for the sensitivity setting before the analyses. The difference between the mean aided sound-field threshold and corrected T-SPL was calculated by subtracting the corrected T-SPL from the aided threshold. Data were considered as missing if either the aided thresholds were not assessed (n = 2), or if T-SPL could not be converted (n = 3).

NRT thresholds were measured intraoperatively and during annual visits with the autoNRT functionality of Custom Sound®. The NRT thresholds measured directly after implantation were generally assessed at all electrodes, while NRT thresholds measured at annual visits were assessed at a selection of electrodes (i.e., electrodes 1, 2, 11, 16 and 22). NRT thresholds cannot be assessed with certain types of implants (CI24RCS and CI24RCA) or when patients indicate that stimuli are too loud. If NRT thresholds were not assessed during the annual visit, the most recently measured thresholds were used, but only if these thresholds were measured more than one year after implantation (n = 28). Otherwise, data were considered as missing (see Table 2 for the number of missing values). The mean, SD, range, and ASV were calculated for the NRT thresholds at implantation (Table 2, NRT intraoperative) and annual visit (Table 2, NRT postoperative). These measures were included to describe the profile of NRT thresholds. The mean and mean absolute differences were calculated between NRT thresholds at implantation and NRT thresholds measured during the annual visit, using the electrodes that were assessed during the annual visit. In addition, absolute and mean differences between the current C levels and NRT thresholds measured intraoperative and during the annual visit were calculated at electrodes that were used for the measurements at implantation and annual visit, respectively. These differences were included because in our CI centre, NRT thresholds and profiles are often used as a guide for setting C levels in children. Finally, the NRT threshold of the most frequently assessed electrode (electrode 22) during the annual visit was included.

There are four different measures of impedances available; monopolar 1 (MP1), monopolar 2 (MP2), monopolar 1+2 (MP1+2) and common ground (CG). Because the correlation between the different impedance measures was very strong ($r > 0.9$) we opted to use impedances measured in MP1+2 mode (i.e., corresponding to the commonly used MP1+2 stimulation mode). Here, an intracochlear electrode is chosen as the active electrode and both extracochlear electrodes (MP1+2) are chosen as return electrodes. The mean and SD were calculated, in addition to the mean and absolute differences in impedance between adjacent electrodes (impedances in Table 2 and Appendix C). Disabled electrodes were not included in the calculation of the different impedance measures. If impedances were not measured during the annual visit ($n = 5$), the most recently measured impedances were used, but only if these were measured more than one year after implantation and, if possible, at the time of other objective measures (i.e., NRT measurements).

Statistical analyses

All statistical analyses were performed with SPSS, version 22.0. Speech recognition in quiet scores were transformed to rationalized arcsine units (RAU) (Sherbecoe and Studebaker, 2004) to normalize variance across the range of scores. A log-transformation was performed on the SRT data to achieve a normal distribution [ln(SRT+7)].

Overall, NRT measures had a considerable amount of missing data (see Table 2 for the amount of missing data), which were assumed to be missing at random. Multiple imputation was used to handle this type of missing data (Netten et al., 2017; Sterne et al., 2009). First, the distributions of the different NRT variables were visually inspected. Not-normally distributed variables were transformed using a log-transformation. Second, missing data was imputed using linear regression, and the imputation was repeated 10 times. Finally, variables that were log-transformed before imputation were back transformed. Some of the data could not be imputed, because they were not missing at random. As mentioned before in the 'Aided-thresholds, NRT thresholds and impedances' paragraph, NRT measurements are not possible with certain (i.e., older) implants. Therefore, the imputed data for users of these implants were deleted, but data of these CI users were included in the remaining analyses. The statistical analyses described below were performed on the imputed dataset. Pooled results over the 10 imputation databases are reported, unless no missing data was present.

Study population

First, effect modification for the onset of severe hearing impairment variable (LO group versus EO group) was investigated. The effect modification was investigated by adding an interaction term to the univariate linear regression analyses. Effect modification was found for many of the independent variables. Therefore, the statistical analyses were done separately for the two groups and stratified models are presented.

Descriptive statistics

Univariate associations between outcome measures and independent variables were tested using Pearson correlations, and median and range were calculated (Table 2).

Prediction models – all parameters

The models were built separately for the EO and LO groups and separately for speech recognition in quiet and in noise, resulting in four different models. Independent variables with a univariate *p*-value < 0.2 were considered as candidate predictors and were selected for the multivariable linear regression model. First, the linear relationship between the candidate predictors and outcome measure was examined. This was tested by dividing the continuous variable in quartiles and by plotting the quartile mean against the regression coefficient. Candidate predictors with a nonlinear relationship with the outcome measures were categorized in four groups with approximately the same sample size. This categorization was done separately for the LO and EO groups. The category with the lowest value was considered as the reference category. A forward selection procedure then was applied to select predictor variables (*p*-entry was set at 0.05), due to the high number of candidate predictors. A *p*-value < 0.05 was considered statistically significant. Regression coefficients, 95% confidence intervals (CIs) and *p*-values are reported.

Prediction models – fitting parameters, aided thresholds and mean (absolute) difference between NRT threshold and C level

Again, four different models were built. A forward selection procedure was used in which all fitting parameters (T level measures, C level measures, DR measures), together with the mean aided thresholds and mean (absolute) differences between NRT thresholds and C level. The models were built separately to identify important parameters which can be adjusted by clinicians to optimize speech recognition of CI users. Contrary to the procedure described in the ‘prediction models – all parameters’ paragraph, all parameters denoted with ‡ in Table 2 were included in the forward selection procedure. The remaining procedure was similar to the procedure described above (i.e., examination of linear relationship between predictor and outcome measure).

Results

The mean speech recognition in quiet was 82.9% (SD = 12.5%) for the LO group and 79.2% (SD = 11.8%) for the EO group. Speech recognition in noise was -0.8 dB SNR (SD = 3.4 dB SNR) for the LO group and 1.3 dB SNR (SD = 3.6 dB SNR) for the EO group. Independent samples *t* tests were conducted to test for significant differences between groups for speech recognition in quiet and in noise. Speech recognition in quiet was not significantly different between groups. Speech recognition in noise was significantly higher (i.e., worse) for the EO group compared to the LO group. Speech recognition scores in quiet and in noise are shown in Figure 1.

Table 2 shows the univariate correlations with speech recognition in quiet and speech recognition in noise for the LO and EO groups. Note that the number of CI users included in the prediction models differs, because of missing data.

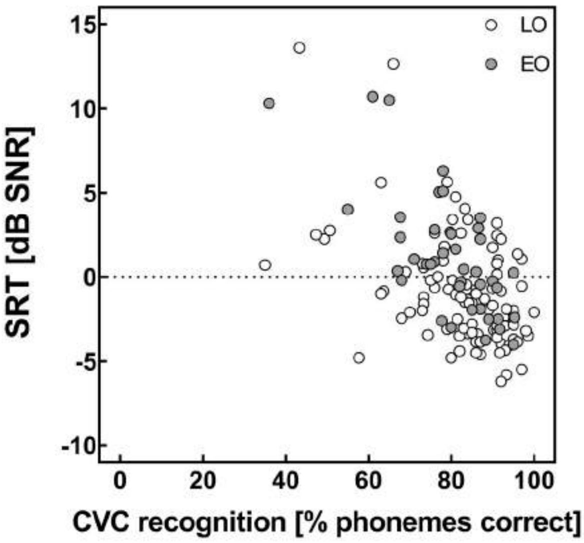


Figure 1. Speech recognition in noise (with digit-triplets) versus speech recognition in quiet (CVC words). The white symbols represent CI users from the LO group (n = 97), grey symbols represent CI users from the EO group (n = 41).

Table 2. Univariate correlations between candidate predictors and speech recognition in quiet and in noise for the late onset of severe hearing impairment (LO) and early onset of severe hearing impairment (EO) groups. Candidate predictors with a p-value < 0.2 (in bold) were included in the forward selection procedure.

	Late onset of severe hearing impairment (n = 97)						Early onset of severe hearing impairment (n = 41)					
	N	Median (range)	Quiet		Noise		N	Median (range)	Quiet		Noise	
			r	*p*	r	*p*			r	*p*	r	*p*
□ □eve□□ □C□□												
Mean□	96	120.1 (66.1 – 188.8)	0.018	.861	0.046	.662	40	127.4 (93.8 – 155.3)	**-0.482**	**□001**	**0□402**	**□010**
□□□	96	9.3 (1.8 – 26.2)	0.043	.681	-0.048	.645	40	9.2 (3.8 – 23.9)	0.085	.606	-0.026	.875
Range□	96	30.0 (6.0 – 90.0)	0.028	.790	-0.001	.993	40	32.0 (17.0 – 64.0)	0.142	.386	-0.048	.771
□□□□	96	2.7 (1.0 – 10.0)	-0.016	.875	0.059	.574	40	2.6 (1.1 – 6.2)	**0□243**	**□132**	-0.019	.906
C-level* (CL)												
Mean□	93	171.0 (115.3 – 222.2)	**0□139**	**□183**	-0.069	.517	40	172.2 (141.4 – 211.6)	**-0.446**	**□004**	**0□296**	**□063**
□□□	93	9.0 (3.0 – 25.3)	-0.001	.996	-0.028	.791	40	8.6 (2.6 – 25.9)	-0.151	.355	0.083	.612
Range□	93	30.9 (9.0 – 85.4)	0.006	.958	-0.015	.890	40	30.8 (10.0 – 96.0)	-0.087	.594	0.022	.895
□□□□	93	2.8 (0.7 – 8.9)	-0.053	.612	0.073	.497	40	2.4 (1.1 – 6.8)	0.092	.574	0.011	.947

Abs, absolute; Ann, annual; ASV, across-site-variation; Diff, difference; DR, electrical dynamic range; NRT, neural response telemetry; SD, standard deviation.
* Corrected for stimulation rate, pulse width and volume; † Median (range) and p values reported for original data; ‡ Variables included in prediction model with fitting parameters only.

Table 2. Continued

	Late onset of severe hearing impairment (n = 97)						Early onset of severe hearing impairment (n = 41)					
	N	Median (range)	Quiet		Noise		N	Median (range)	Quiet		Noise	
			r	p	r	p			r	p	r	p
DR ⍰C⍰⍰												
Mean⍰	93	49.3 (26.1 – 75.3)	**0⍰148**	**⍰157**	**-0.184**	**⍰082**	40	45.9 (31.7 – 90.1)	0.031	.853	-0.133	.414
⍰⍰⍰	93	5.9 (1.6 – 17.0)	0.126	.231	-0.050	.642	40	6.2 (1.1 – 16.5)	0.182	.264	-0.131	.423
Range⍰	93	20.7 (5.5 – 59.0)	0.109	.300	-0.001	.989	40	20.5 (4.0 – 47.8)	0.197	.223	-0.177	.277
⍰⍰⍰⍰	93	2.1 (0.6 – 6.8)	0.003	.980	0.004	.974	40	2.1 (0.6 – 4.9)	0.169	.300	-0.173	.288
Audiometry ⍰d⍰ S⍰⍰⍰												
Mean aided threshold⍰	95	26.7 (16.0 – 45.0)	**-0.478**	**⍰000**	**0⍰261**	**⍰011**	**41**	27.0 (20.0 – 36.0)	**-0.330**	**⍰035**	**0⍰238**	**⍰135**
Mean aided threshold - T-SPL⍰	93	1.0 (-12.0 – 17.5)	**-0.420**	**⍰000**	**0⍰231**	**⍰027**	**40**	1.0 (-5.0 – 15.0)	**-0.313**	**⍰049**	**0⍰266**	**⍰098**
NRT thresholds postoperative† (CL)												
Electrode 22	70	166.0 (128.0 – 189.0)	-0.126	.358	0.136	.314	27	164.0 (125.0 – 209.0)	**-0.473**	**⍰009**	**0⍰446**	**⍰012**
Mean	75	171.6 (122.0 – 192.5)	-0.067	.567	0.033	.782	31	168.0 (149.0 – 209.0)	**-0.318**	**⍰082**	0.176	.348

Abs, absolute; Ann, annual; ASV, across-site-variation; Diff, difference; DR, electrical dynamic range; NRT, neural response telemetry; SD, standard deviation.

* Corrected for stimulation rate, pulse width and volume; † Median (range) and p values reported for original data; ‡ Variables included in prediction model with fitting parameters only.

Table 2. Continued

	Late onset of severe hearing impairment (n = 97)						Early onset of severe hearing impairment (n = 41)					
	N	Median (range)	Quiet		Noise		N	Median (range)	Quiet		Noise	
			r	*p*	r	*p*			r	*p*	r	*p*
NRT thresholds postoperative† (CL)												
□□	73	12.5 (3.3 – 26.0)	0.083	.490	-0.133	.296	28	12.6 (4.5 – 27.3)	0.090	.635	-0.197	.309
Range	73	30.0 (6.0 – 57.0)	0.146	.261	**-0.187**	**□119**	28	27.0 (9.0 – 72.0)	0.161	.401	-0.171	.387
□□□	73	12.8 (4.5 – 26.3)	-0.021	.869	-0.142	.237	28	14.1 (4.5 – 33.0)	0.074	.697	-0.227	.247
Mean diff C level□	72	0.5 (-40.2 – 33.4)	**-0.225**	**□059**	0.139	.261	30	-1.4 (-27.1 – 22.8)	-0.018	.930	-0.111	.581
Mean abs diff C level□	72	11.1 (0.2 – 40.2)	0.065	.628	-0.069	.569	30	10.1 (1.0 – 27.1	-0.258	.240	0.023	.905
NRT thresholds intraoperative† (CL)												
Mean	76	186.7 (152.1 – 240.6)	0.018	.878	0.146	.239	33	181.3 (162.1 – 226.8)	-0.118	.485	**0□262**	**□130**
□□	76	14.8 (5.5 – 38.3)	-0.067	.596	0.104	.381	33	15.5 (7.7 – 25.2)	**0□289**	**□081**	-0.227	.205
Range	76	54.5 (17.0 – 108.0)	0.016	.897	0.043	.739	33	58.0 (25.0 – 89.0)	**0□303**	**□065**	-0.238	.203
□□□	76	8.0 (1.4 – 21.2)	-0.088	.482	**0□259**	**□025**	33	7.7 (3.2 – 14.4)	**0□230**	**□177**	-0.067	.713

Abs, absolute; Ann, annual; ASV, across-site-variation; Diff, difference; DR, electrical dynamic range; NRT, neural response telemetry; SD, standard deviation.

* Corrected for stimulation rate, pulse width and volume; † Median (range) and p values reported for original data; ‡ Variables included in prediction model with fitting parameters only.

Table 2. Continued

	Late onset of severe hearing impairment (n = 97)						Early onset of severe hearing impairment (n = 41)					
	N	Median (range)	Quiet		Noise		N	Median (range)	Quiet		Noise	
			r	*p*	r	*p*			r	*p*	r	*p*
NRT thresholds intraoperative† (CL)												
Mean diff C level	72	16.1 (-37.7 – 53.6)	-0.101	.434	**0⍰316**	**⍰011**	32	11.6 (-28.9 – 44.7)	**0⍰259**	**⍰132**	-0.072	.681
Mean abs diff C level	72	18.1 (4.2 – 53.6)	-0.038	.765	**0⍰258**	**⍰031**	32	17.3 (5.9 – 44.7)	0.111	.520	0.168	.361
Mean diff ann NRT	65	-17.3 (-75.0 – 12.8)	-0.062	.642	-0.102	.441	27	-17.2 (-51.0 – 16.4)	-0.028	.899	-0.167	.369
Mean abs diff ann NRT	65	19.0 (6.0 – 75.0)	-0.004	.979	**0⍰223**	**⍰078**	27	18.0 (5.4 – 51.0)	-0.018	.931	0.178	.379
Impedances (kΩ)												
Mean	97	7.6 (1.8 – 12.4)	0.029	.778	-0.010	.922	41	7.4 (5.0 – 10.6)	0.016	.921	0.009	.958
⍰⍰	97	1.5 (0.5 – 3.4)	**-0.226**	**⍰026**	0.112	.285	41	1.5 (0.6 – 2.9)	0.030	.852	0.071	.659
Mean diff	97	0.2 (-0.1 – 0.5)	**-0.268**	**⍰008**	**0⍰158**	**⍰129**	41	0.1 (-0.1 – 0.4)	-0.153	.338	0.090	.580
Mean abs diff	97	0.7 (0.3 – 1.4)	**-0.174**	**⍰087**	**0⍰242**	**⍰018**	41	0.7 (0.5 – 1.3)	0.117	.467	0.130	.422

Abs, absolute; Ann, annual; ASV, across-site-variation; Diff, difference; DR, electrical dynamic range; NRT, neural response telemetry; SD, standard deviation.
* Corrected for stimulation rate, pulse width and volume; † Median (range) and p values reported for original data; ‡ Variables included in prediction model with fitting parameters only.

Table 3. Final multivariable prediction models with fitting parameters, aided thresholds, NRT thresholds, and impedances for the LO group (left column) and EO group (right column) for speech recognition in quiet. In total, 78 CI users were included in the model for the LO group, and 36 CI users were included in the model for the EO group. The B values and 95% confidence intervals are presented for RAU scores.

	LO Group (n = 78)				**EO Group** (n = 36)		
	B	95% CI	*p*		B	95% CI	*p*
Mean aided thresholds				**Mean C level**			
< 24.0 dB HL	Reference			< 120.0 CL	Reference		
24.0 – 27.0 dB HL	-2.29	-10.02 to 5.43	.556	120.0 – 127.5 CL	-6.01	-16.66 to 4.64	.260
27.0 – 30.0 dB HL	-10.63	-18.51 to -2.75	**.009**	127.5 – 135.0 CL	-7.06	-17.16 to 3.04	.165
> 30.0 dB HL	-12.11	-19.94 to -4.29	**.003**	≥ 135.0 CL	-18.61	-29.26 to -7.97	**.001**
Mean absolute difference in impedance							
< 0.55 kΩ	Reference						
0.55 – 0.725 kΩ	-7.86	-16.12 to 0.40	.062				
0.725 – 0.90 kΩ	-9.88	-17.80 to -1.96	**.015**				
> 0.90 kΩ	-12.23	-21.34 to -3.11	**.009**				

DR, electrical dynamic range.

6

Table 3. Continued

	▯O ▯roup (n = 78)				**▯O ▯roup** (n = 36)		
	B	95% CI	*p*		B	95% CI	*p*
Mean DR							
< 40.0 CL	Reference						
40.0 – 50.0 CL	9.78	2.26 to 17.30	**▯012**				
50.0 – 60.0 CL	11.55	3.34 to 19.76	**▯006**				
> 60.0 CL	6.67	-1.27 to 14.61	.098				
Standard deviation of impedances							
< 1.12 kΩ	Reference						
1.12 – 1.53 kΩ	10.88	2.38 to 19.37	**▯013**				
1.53 – 2.00 kΩ	4.85	-3.65 to 13.35	.259				
> 2.00 kΩ	0.30	-8.63 to 9.23	.947				
Adjusted R^2		0.26		Adjusted R^2		0.20	

DR, electrical dynamic range.

Table 4. Multivariable prediction model for speech recognition in quiet with fitting parameters, aided thresholds and mean (absolute) differences between NRT thresholds and C levels for the LO (left column) and EO (right column) groups. In total, 85 CI users were included in the model for the LO group, and 36 CI users were included in the model for the EO group. The B values and 95% confidence intervals are presented in RAU scores.

	LO Group (n = 85)				**EO Group** (n = 36)		
	B	95% CI	*p*		B	95% CI	*p*
Mean aided thresholds				**Mean [illegible] level**			
< 24.0 dB HL	Reference			< 120.0 CL	Reference		
24.0 – 27.0 dB HL	-3.29	-11.35 to 4.77	.419	120.0 – 127.5 CL	-6.01	-16.66 to 4.64	.260
27.0 – 30.0 dB HL	-10.44	-18.39 to -2.48	**.011**	127.5 – 135.0 CL	-7.06	-17.16 to 3.04	.165
> 30.0 dB HL	-11.29	-19.38 to -3.19	**.007**	≥ 135.0 CL	-18.61	-29.26 to -7.97	**.001**
Mean DR							
< 40.0 CL	Reference						
40.0 – 50.0 CL	5.55	-2.09 to 13.19	.152				
50.0 – 60.0 CL	10.18	1.98 to 18.39	**.016**				
> 60.0 CL	4.34	-3.98 to 12.65	.303				
Adjusted R^2		0.13		Adjusted R^2		0.20	

DR, electrical dynamic range.

6

Speech recognition in quiet

[illegible]r[illegible] eters

Table 3 shows the final multivariable prediction models of speech recognition in quiet for the LO and EO groups. All parameters were considered: fitting parameters, aided thresholds, NRT thresholds and impedances. Candidate predictors with $p < 0.2$ were entered in the prediction models (in bold in Table 2). There were four significant predictors of speech recognition in quiet in the LO group: mean aided thresholds, mean absolute difference in impedances, mean DR, and the standard deviation of impedances. Poorer speech recognition in quiet was found for CI users with mean aided thresholds higher than 27 dB HL compared to those with mean aided thresholds better than 24 dB HL. The mean absolute difference in impedances, which describes the profile of impedances across the electrode array, significantly predicted speech recognition in quiet when the mean absolute difference in impedances was large (i.e., > 0.725 kΩ). More specific, speech recognition in quiet decreases with a larger mean absolute difference in impedances. Furthermore, a DR between 40-50 and 50-60 CL yielded better speech recognition in quiet than a smaller DR of less than 40 CL. Finally, one other aspect of impedance measurements predicted speech recognition in quiet, which is the standard deviation of impedances. This parameter reflects the variation in impedances across the electrode array. Users with a standard deviation of impedances between 1.12 and 1.53 kΩ had better speech recognition compared to CI users with a standard deviation of impedances less than 1.12 kΩ. The total variance in speech recognition in quiet explained by the model was 26%.

For the EO group, results were different; only one significant predictor of speech recognition in quiet was found. In this group, CI users with the highest mean T levels (i.e., above 135 CL) had worse speech recognition in quiet than CI users with the lowest mean T levels (i.e., lower than 120 CL). The total variance explained by the model was 20%.

Fitting parameters, aided thresholds and mean (absolute) difference between NRT threshold and C level

Table 4 shows the final multivariable prediction models of speech recognition in quiet explained with fitting parameters (i.e., T and C levels, DR), mean aided thresholds, and mean (absolute) difference between NRT thresholds and C levels. The results of these prediction models were similar to the results of the prediction models described above, with the exception of the impedance measures that were not included in this model.

Two parameters were significant predictors of speech recognition in quiet in the LO group: mean aided thresholds and mean DR. CI users had worse speech recognition in quiet if they had mean aided thresholds higher than 27 dB HL compared to CI users with mean aided thresholds better than 24 dB HL. CI users with a mean DR of 50-60 CL had better speech recognition in quiet than CI users with a smaller DR of less than 40 CL. The variance explained by the total model was 13%.

The prediction model with fitting parameters in the EO group gave the same result as with 'Fitting parameters, aided thresholds, NRT thresholds and impedances' (see above).

Table 5. Final multivariable prediction models with fitting parameters, aided thresholds, NRT thresholds, and impedances for the LO group (left column) and EO group (right column) for speech recognition in noise. In total, 80 CI users were included in the model for the LO group, and 38 CI users were included in the model for the EO group. The B values and 95% confidence intervals are presented in transformed scores.

	LO Group (n = 80)				**EO Group** (n = 38)		
	B	95% CI	*p*		B	95% CI	*p*
Mean aided thresholds				**Mean [illegible] level[illegible]**	0.01	0.00 to 0.02	**.010**
< 24.0 dB HL	Reference						
24.0 – 27.0 dB HL	0.21	-0.10 to 0.52	.178				
27.0 – 30.0 dB HL	0.48	0.17 to 0.79	**.003**				
> 30.0 dB HL	0.34	0.02 to 0.66	**.041**				
Mean absolute difference in impedance							
< 0.55 kΩ	Reference						
0.55 – 0.725 kΩ	0.09	-0.22 to 0.40	.568				
0.725 – 0.90 kΩ	0.31	0.01 to 0.62	**.043**				
> 0.90 kΩ	0.48	0.16 to 0.79	**.003**				

DR, electrical dynamic range

* Note that these variables had a linear relationship with the outcome measure and were therefore not divided into four categories.

Table 5. Continued

		?O ?roup (n = 80)				**?O ?roup** (n = 38)		
		B	95% CI	*p*		B	95% CI	*p*
Mean DR								
	< 40.0 CL	Reference						
	40.0 – 50.0 CL	-0.36	-0.66 to -0.06	**?020**				
	50.0 – 60.0 CL	-0.27	-0.59 to 0.04	.090				
	> 60.0 CL	-0.24	-0.56 to 0.08	.139				
Adjusted R^2			0.14		Adjusted R^2		0.14	

DR, electrical dynamic range

* Note that these variables had a linear relationship with the outcome measure and were therefore not divided into four categories.

Table 6. Multivariable prediction model for speech recognition in noise with fitting parameters, aided thresholds and mean (absolute) differences between NRT thresholds and C levels for the LO (left column) and EO (right column) groups. In total, 89 CI users were included in the model for the LO group, and 34 CI users were included in the model for the EO group. The B values and 95% confidence intervals are presented in transformed scores.

	LO Group (n = 89)				**EO Group** (n = 34)		
	B	95% CI	*p*		B	95% CI	*p*
Mean aided thresholds				**Mean C levels***	0.01	0.00 to 0.02	**.012**
< 24.0 dB HL	Reference			**Range of DR**			
24.0 – 27.0 dB HL	0.23	-0.08 to 0.54	.139	< 12.0 CL	Reference		
27.0 – 30.0 dB HL	0.44	0.13 to 0.75	**.007**	12.0 – 21.0 CL	0.41	0.10 to 0.72	**.011**
> 30.0 dB HL	0.29	-0.03 to 0.61	.071	21.0 – 30.0 CL	0.27	-0.05 to 0.60	.095
				> 30.0 CL	-0.03	-0.33 to 0.29	.871
				Mean aided thresholds*	0.03	0.01 to 0.06	**.022**
Adjusted R^2		0.05		Adjusted R^2		0.34	

DR, electrical dynamic range

* Note that these variables had a linear relationship with the outcome measure and were therefore not divided into four categories.

Speech recognition in noise

All parameters

Table 5 shows the final multivariable prediction models of speech recognition in noise for the LO and EO groups. All parameters were considered: fitting parameters, aided thresholds, NRT thresholds and impedances. Candidate predictors with $p < 0.2$ were entered in the prediction models (in bold in Table 2). The prediction model for speech recognition in noise showed much overlap with the prediction model for speech recognition in quiet. Significant predictors of speech recognition in noise in the LO group were mean aided thresholds, mean absolute difference in impedances, and mean DR. Poorer speech recognition in noise was found for CI users with mean aided thresholds higher than 27 dB HL compared to CI users with mean aided thresholds better than 24 dB HL. Furthermore, speech recognition decreased when the mean absolute difference in impedances was large (i.e., > 0.725 kΩ). Finally, CI users with a mean DR of 40-50 CL had better speech recognition in noise compared to CI users with a mean DR of less than 40 CL. The total variance in speech recognition in noise explained by the model was 14%.

The prediction model for speech recognition in noise in the EO group was similar to the prediction model for speech recognition in quiet in the EO group. There was only one significant predictor of speech recognition in noise: mean T level. Similar to speech recognition in quiet, CI users with higher mean T levels had worse speech recognition in noise. The variance in speech recognition explained by the mean T level was 14%.

6

Fitting parameters, aided thresholds and mean (absolute) difference between NRT threshold and C level

The multivariable models with fitting parameters, aided thresholds and mean (absolute) difference between NRT thresholds and C levels as predictors for speech recognition in noise in the LO and EO groups are presented in Table 6. These models were built with all fitting parameters (i.e., T and C levels, DR), mean aided thresholds and mean (absolute) difference between NRT thresholds and C levels.

For the LO group, there was only one fitting parameter that predicted speech recognition in noise: mean aided thresholds. CI users with higher mean aided thresholds, between 27 and 30 dB HL, had worse speech recognition in noise compared to CI users with mean aided thresholds better than 24 dB HL. The multivariable model explained only 5% of the variance in speech recognition in noise in the LO group.

There were three fitting parameters that were significant predictors of speech recognition in noise in the EO group: mean T level, range of DR and mean aided thresholds. The mean T level appeared to be a significant predictor in the model for speech recognition in noise, such that with higher T levels, speech recognition in noise worsened. Furthermore, CI users with a range of DR between 12 and 21 CL had worse speech recognition in noise compared to CI users with a range in DR less than 12 CL. Finally, CI users with higher mean aided thresholds had worse speech recognition in noise. The multivariable model resulted in an explained variance in speech recognition in noise of 34%.

Discussion

The objective of this study was to identify parameters which may improve current fitting practices and, thus, optimize speech recognition of CI users. Prediction models were built with parameters that are available to an audiologist during a fitting session. The prediction models were built separately for speech recognition in quiet and in noise, and separate analyses were performed for postlingually deafened CI users with early onset of severe hearing impairment and CI users with late onset of severe hearing impairment.

For the LO group, elevated mean aided thresholds were found to have a negative relation with speech recognition in quiet and in noise. As an example, mean speech recognition in quiet was 90.5% for CI users with aided thresholds better than 24 dB HL compared to 77.9% for CI users with aided thresholds of 27-30 dB HL. Speech recognition in noise was -2.0 dB SNR for CI users with aided thresholds better than 24 dB HL versus -0.3 dB SNR for CI users with aided thresholds between 27-30 dB HL. CI users with a larger mean DR (i.e., between 40-60 CL) had better speech recognition both in quiet and in noise than CI users with a mean DR of less than 40 CL. Furthermore, the mean absolute difference between adjacent electrodes and the standard deviation of impedances across the electrode array were found to be associated with speech recognition in quiet and in noise. For the EO group, higher mean T levels were associated with worse speech recognition in quiet and in noise. To illustrate, speech recognition in quiet for CI users with T levels < 120 CL was 86.0% versus 68.8% for CI users with T levels > 135 CL. C levels and NRT thresholds were not found to be predictors of speech recognition in this study.

T levels

T levels represent the minimum electrical current which yields an acoustic percept. Previous studies (Baudhuin et al., 2012; Busby and Arora, 2016; Van der Beek et al., 2015) have shown the importance of setting T levels correctly for understanding speech in quiet and in noise. Busby and Arora (2016) found poorer speech recognition in quiet when T levels were set at lower levels (i.e., 60% and 90% expansion of the DR), and poorer speech recognition in noise for T levels that were set at higher levels (i.e., 60% compression of the DR). Van der Beek et al. (2015) found a significant correlation (r = 0.34) between T levels and speech recognition in quiet in users of Advanced Bionics devices, whilst Pfingst et al. (2004) did not find a correlation between mean T level and speech recognition in quiet and in noise. Several mechanisms may play a role in the relation between T levels and speech recognition performance. Studies have shown a relation between the stimulation levels and radial distance of the electrode from the modiolus, and found higher stimulation levels (i.e., higher T levels) for greater distances (DeVries, Scheperle and Bierer, 2016; Long et al., 2014; Saunders et al., 2002). Current spread due to higher stimulation levels is known to increase the risk of channel-channel interactions which also inhibits speech recognition (Jones et al., 2013). Also, less surviving spiral ganglion cells along the cochlea are expected in CI users with longer durations of deafness, thus CI users in the EO group, and this may also result in higher T levels. This could explain our findings in the EO group, and might also explain why we did not find such a relation in the LO group. Thus, it might be better to lower T levels to provide optimal speech recognition, but the above mentioned explanations might inhibit the lowering of T levels in CI users in the EO group. It might also be difficult for this group to provide reliable feedback when thresholds are assessed with soft stimuli, which will most likely result in T levels that are set too high. In that case, aided thresholds (i.e., with narrow band noise) might be a better indicator and could be used to verify T levels. If T levels are verified and appear to be set correctly, lowering T levels has limited value.

Electrical dynamic range

Fitting of the sound processor often starts by maximizing the use of the dynamic range of the auditory nerve, with T levels set at threshold and increasing C levels. Previous studies have shown that the magnitude of the dynamic ranges has an effect on speech recognition (Blamey et al., 1992; Loizou et al., 2000; Pfingst and Xu, 2005; Van der Beek et al., 2015), which are in line with the findings of the current study. The DR was found to be associated with speech recognition, only for users in the LO group. The same findings were expected in the EO group. Possibly, the dissimilarity in findings between groups is caused by the smaller sample size of the EO group compared to the LO group and because the EO group might be more heterogeneous in some aspects (i.e., age at onset of severe hearing impairment, education for the hearing impaired) than the LO group.

In our CI centre, C levels are increased during the first few weeks after the initial fitting to let CI users acclimatize to the increasing loudness while the DR increases. Based on the findings of the current study, the expansion of the DR seems to be a good approach.

C levels

C levels were not found to be a predictor of speech recognition in this study in any of our prediction models. The correlation between T and C levels is moderate to strong (r = 0.78 and r = 0.75 for the LO and EO groups, respectively). Because candidate predictors with a strong correlation generally do not end up in the same prediction model, we built a new prediction model for speech recognition in noise for the EO group, but excluded T levels from the forward selection procedure. For speech recognition in noise in the EO group, C levels indeed became significant predictors, explaining the same amount of variance as T levels did in the original prediction model. Now worse speech recognition in quiet was found for CI users with higher mean C levels. This might again be related to greater distances between the electrode and modiolus (DeVries et al., 2016; Long et al., 2014; Saunders et al., 2002) or due to current spread and the accompanying risk of channel-channel interactions (Jones et al., 2013), both negatively impacting speech recognition.

Mean aided thresholds

In line with the current study, several studies have shown a significant relationship between aided sound-field thresholds and speech recognition (Busby and Arora, 2016; Davidson et al., 2009; Firszt et al., 2004; Holden et al., 2013). Multiple studies have investigated the effect of setting T levels above or below hearing thresholds in people using the Cochlear™ Nucleus system (Busby and Arora, 2016; Dawson et al., 2007; Franck, Xu and Pfingst, 2003; Skinner et al., 1999; Zeng and Galvin III, 1999; Zhou and Pfingst, 2014). Elevated aided thresholds are the result of T levels that are set too low (Busby and Arora, 2016; Vaerenberg et al., 2014). If T levels are set too low, stimuli will be presented below the hearing thresholds, whilst T levels that are properly set, will result in aided thresholds at 25 dB SPL (i.e., at T-SPL) when the sensitivity is set at 12. Sometimes, T levels are intentionally set below the psychophysically determined threshold, because CI users complain about soft ambient sounds that are perceived too loud. CI users can also opt to lower the microphone sensitivity, which will also result in higher aided thresholds. Both may result in poorer speech recognition. Furthermore, aided thresholds might be elevated in CI users with a so called T-tail. A T-tail refers to regions with very slow loudness growth near threshold levels (Donaldson and Allen, 2003). In case of a T-tail, there is limited change in loudness percept in the lower part of the DR, across a wide range of stimulation levels. To eliminate these regions with slow loudness growth, T levels should be raised to the point at which the loudness begins to grow with increases in stimulus level (Wolfe and Schafer, 2015).

The findings of the current study and previous studies indicate the importance of measuring aided thresholds and ensure that they are at the correct level for optimal speech recognition. This is assumed to be important for CI users in the EO group as well. In case of elevated aided thresholds, clinicians should emphasize to CI users that lower aided thresholds are better for speech recognition and should encourage them to attempt to acclimatize to louder ambient sounds after raising T levels. Changing the sensitivity to a lower, less sensitive, setting should be discouraged for the same reason. The results also suggest that T levels should be determined precisely to prevent stimulation below hearing thresholds. The current procedure for setting T levels requires feedback from the CI user which might be difficult, especially when stimulation is near threshold levels. Recently, Rader et al. (2018) proposed an alternative way to precisely determine T levels, which resulted in an improvement in aided thresholds. In short, the method included the sequential presentation of two stimuli with different levels in a single channel. The number of stimuli perceived is used to determine the actual T level: if two stimuli are perceived, the stimuli were presented above hearing threshold; if no stimuli are perceived, the stimuli were presented below hearing threshold. If one stimulus is perceived, the T level is assumed to be located between the two presented stimuli. The method starts with a rough approximation, followed by two iterations with smaller level differences between the two stimuli.

It should be noted that T levels that are set too high, will not have an effect on the aided threshold. Then, a stimulus below T-SPL will still not lead to electrical stimulation. Thus, if T levels are set too high and both T-SPL and sensitivity are set at default (i.e., 25 dB SPL and 12, respectively), then aided thresholds should still be around the target level of 25 dB SPL. However, T levels that are set too high may be reflected in the feedback of CI users who complain about soft environmental sounds that are perceived too loud.

NRT thresholds

The use of NRT thresholds as an alternative for behavioral fitting has been widely studied (see He, Teagle and Buchman (2017) and de Vos et al. (2018) for an overview). In our CI centre, NRT thresholds are often used as a guide to set C levels in children. Based on our clinical experiences, we expected that smaller differences between C levels and NRT thresholds would be associated with better speech recognition. We did not, however, find such a relationship in the current study. Other studies have shown that there is only a weak to moderate (r = 0.58) correlation between NRT thresholds and C levels (de Vos et al., 2018). We explored the data and found correlations of 0.38-0.67 between NRT thresholds and C levels on the most frequently assessed electrodes (i.e., electrodes 1, 2, 6, 11, 16, 22). The correlation is strongest for electrode 16 and weakest for electrode 1 (Figure 2). For electrode 16, the mean difference between NRT thresholds and C levels is 3 CL, and around 50% of the C levels are within 10 CL of the NRT thresholds. It is important to note that the C levels were corrected for volume, pulse width, and stimulation rate if deviating from the reference settings (i.e., volume = 10, pulse width = 25 µs, and stimulation rate = 900 Hz). This correction is important to retain the original relation between C levels and NRT thresholds. For instance, if C levels are shifted because of changes in the volume setting (i.e., from the default setting of 10 to a lower value of 6), the relation between C levels and NRT thresholds will be shifted as well.

The results of the current study suggest that fitting based on NRT thresholds may not be considered a complete alternative for behavioral fitting in adult CI users. However, NRT thresholds might give a good first indication of stimulation levels if behavioral fitting proves to be difficult, for instance in children. In addition to the application of NRT thresholds in fitting, there are numerous other applications of NRT thresholds that are currently being studied and might be of value for clinical practice, for instance to estimate the neural survival of auditory nerve fibers (see He et al. (2017) for an overview).

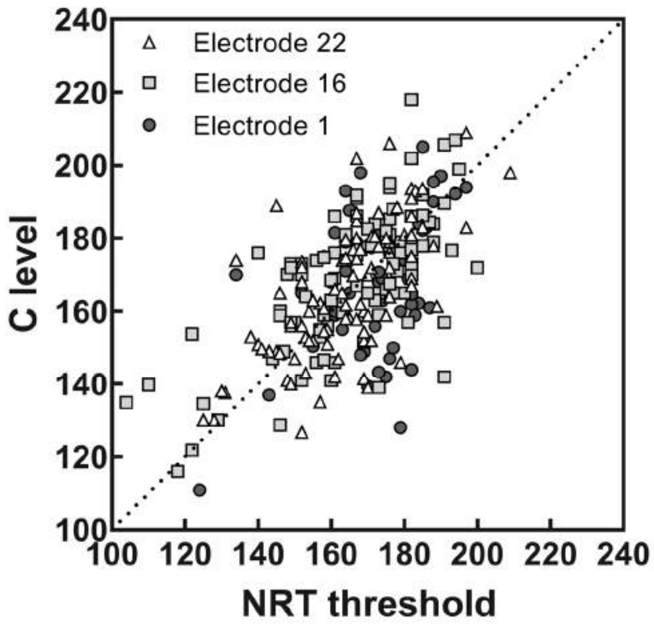


Figure 2. Scatterplot of NRT Thresholds versus C levels. Scatterplots are shown in different colors for electrodes 1, 16 and 22. The dotted line represents equal NRT thresholds and C levels.

Impedances

Impedances are frequently assessed as a measure of electrode functioning. With the different impedance measures, we aimed to describe the profile of impedances across the electrode array. The general assumption is that impedance profiles should not show an erratic pattern, but should be relatively flat showing only mild variations (Wolfe and Schafer, 2015). Impedances are related to the resistive characteristics of the surroundings of the electrode.

The tissue and fluid surrounding the electrode can be influenced by the electrode position, for instance in case of proximity to the modiolus, or partial insertion in the scala vestibuli instead of the scala tympani. Variations in impedances across the electrode array, leading to the higher mean absolute differences between adjacent electrodes, result in worse speech recognition in quiet and in noise. Studies have shown that impedances can be different for basal and apical electrodes, and for different types of electrodes (Busby, Plant and Whitford, 2002; Saunders et al., 2002). Exploration of our data revealed that CI users with a mean absolute difference in impedances above 0.725 kΩ were relatively more often implanted with electrodes other than the CI24RECA implant (i.e., the CI512 implant). A subsequent analyses revealed that CI users in the LO group with the CI512 implant had significantly worse speech recognition in quiet than CI users implanted with other implants. The findings suggest that there is more difference in impedances across the electrode array with the CI512 implant. More specifically, 73% of CI users in the LO group implanted with the CI512 implant, had a mean absolute difference above 0.725 kΩ. The absence of an association between impedances and speech recognition in the EO group might be explained by the smaller sample size, but might also be related to the smaller number of CI users implanted with the CI512 implant in this group (7.3% in the EO group versus 15.5% in the LO group).

The results of this study suggest that the profile of impedances across the electrode array is more important for speech recognition performance than the mean value of impedances. Clinicians should therefore measure impedances and evaluate the impedance profile, in addition to possible short or open circuits and changes in impedances over time. If erratic profiles of impedances are found, clinicians might opt for an integrity test and should counsel CI users about their expectations in terms of speech recognition (i.e., more erratic profile might lead to poorer speech recognition).

Strengths and limitations

The strength of the current study is the use of a study population of experienced postlingually deafened adult CI users who are homogeneous with respect to CI centre, CI brand, speech processing strategy, stimulation rate, pulse width, number of maxima, and who are rehabilitated in a team with a small number of surgeons, audiologists and speech pathologists. Although Vaerenberg et al. (2014) indicated that there is considerable variation in fitting methods used by audiologists from different CI centres, the fitting practices of audiologists in our centre are considered fairly similar. The results of this study might not be directly applicable to CI users of other CI centres, because of the differences in fitting practices between CI centres.

The results of this study cannot be directly applied or generalized to users of different brands of CIs. We applied strict inclusion criteria, which resulted in a high exclusion rate. However, this led to a relatively homogeneous study population. We excluded CI users with strongly deviating parameters (e.g., BP1 stimulation mode, > 3 electrodes disabled), because in our centre, these parameters are only modified in exceptional cases. Furthermore, we excluded prelingually deaf adults and performed separate analyses for postlingually deaf adults with early onset of severe hearing impairment, with the assumption that the fitting would be different for these patients. The results showed that the predictors of speech recognition are indeed different for the EO group than for the LO group of CI users. However, the sample size of the EO group is considerably smaller than the sample size of the LO group, which might also be the cause of the difference in predictors between the two groups.

The explained variance of the prediction models described in this study is limited. For the LO group, the explained variance for speech recognition in quiet ranges from 13% to 26%, and 5% to 14% for speech recognition in noise. For the EO group, the explained variance for speech recognition in quiet is 20%, and ranges from 14% to 34% for speech recognition in noise.

The different NRT threshold variables had considerable missing data, for which we applied multiple imputation. However, some of the data were not missing at random (i.e., NRT measurements not possible with certain implants) and could therefore not be imputed. However, the presented multivariable prediction models do not include any of the NRT variables as predictors. Thus, the prediction models would have been the same if the analyses would have been done on the original data without imputation. In addition to missing data on the NRT measures, data was missing on several other parameters. All cases were included in the analyses, however, multivariate linear regression analysis only includes complete cases. This has therefore resulted in different numbers of CI users included in the different prediction models, which limited the statistical power.

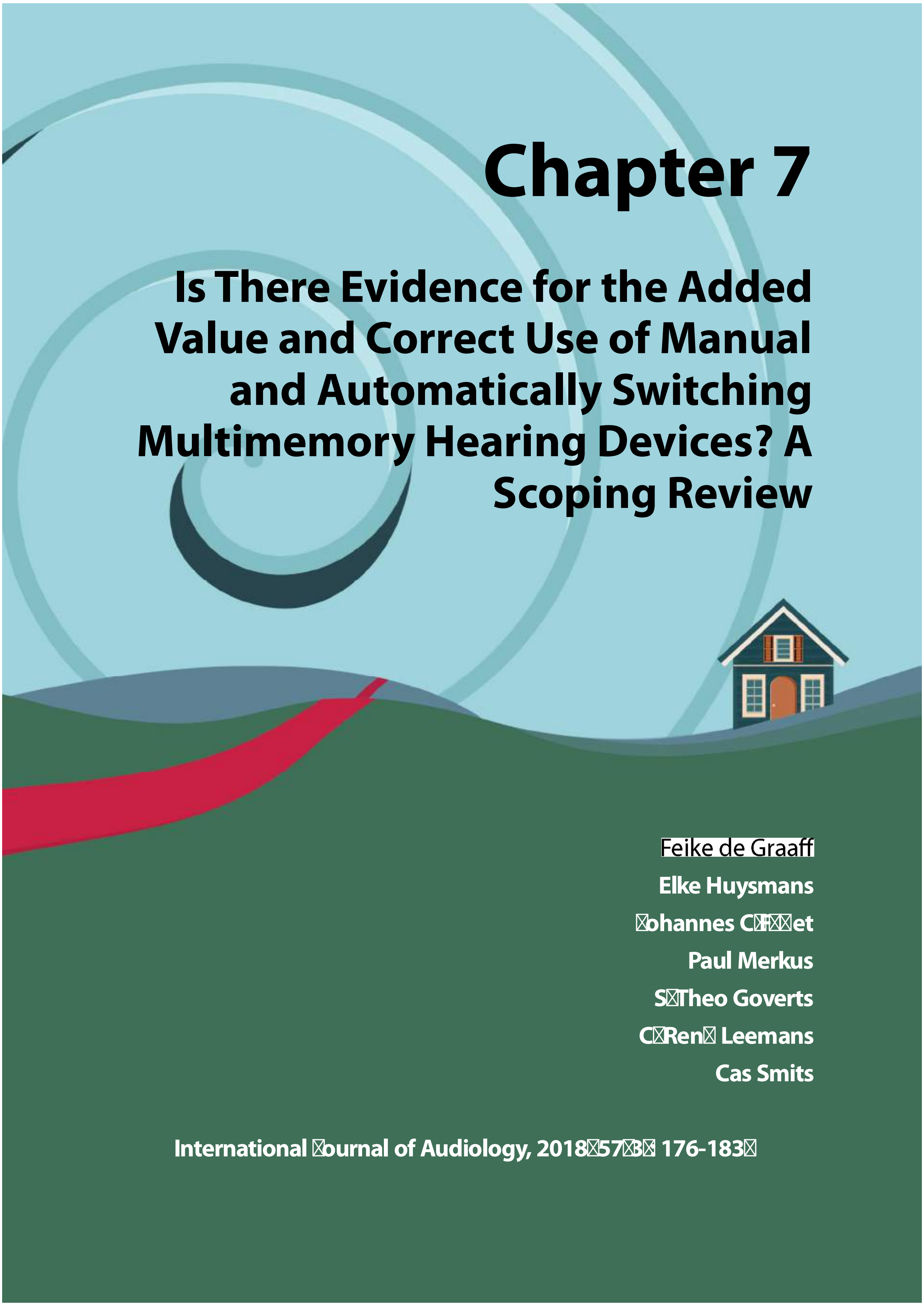
The clinical relevance of the predictors identified in the current study should be investigated. In addition, it should be examined whether adjustments to these predictors results in improved speech recognition of CI users who’s fitting deviates from the optimal fitting identified in the current.

Clinical implications

The results of this study may guide audiologists in their fitting practices and improve the performance of CI users. Clinicians should measure aided thresholds and emphasize the importance to CI users that they should be approximately at 25 dB SPL for optimal speech recognition. To reach this target, T levels should be precisely determined and should not be set below hearing thresholds. If aided thresholds are above the target level, clinicians should raise T levels and counsel CI users to get accustomed to ambient sounds and discourage them to lower the sensitivity. The DR should preferably be between 40-60 CL, by setting T levels at threshold and increasing C levels. Finally, clinicians should be aware of profiles of impedances other than a flat profile with mild variations, because this could lead to poorer speech recognition in quiet and in noise. Then, CI users should be counseled to manage their expectations about speech recognition and possibly schedule an integrity test to check for a soft failure of the implant.

Conclusions

In conclusion, we were able to identify important CI fitting parameters to predict speech recognition in quiet and in noise in two groups of CI users (i.e., early and late onset of severe hearing impairment). The predictors found in this study were very similar for speech recognition in quiet and in noise, which suggests that optimizing speech recognition in quiet will also optimize speech recognition in noise, or will at least not be at the expense of speech recognition in noise. In the group of CI users with late onset of severe hearing impairment, parameters that were found to be associated with speech recognition performance were the mean aided thresholds, DR, and measures to express the impedance profile across the electrode array. Elevated aided thresholds result in worse speech recognition in quiet and in noise. Clinical intervention is required to raise T levels in case of elevated aided thresholds. Furthermore, CI users with a larger DR were found to have better speech recognition, both in quiet and in noise. In the group of CI users with early onset of severe hearing impairment, worse speech recognition in quiet and in noise was found for CI users with higher T levels. Future research should assess the clinical relevance of the predictors identified in this study.



Chapter 7

Is There Evidence for the Added Value and Correct Use of Manual and Automatically Switching Multimemory Hearing Devices? A Scoping Review

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Abstract

Objectives: To review literature on the use of manual and automatically switching multimemory devices by hearing aid and cochlear implant (CI) recipients, and to investigate if recipients appreciate and adequately use the ability to switch between programmes in various listening environments.

Design: Literature was searched using PubMed, Embase, and ISI/Web of Science. Additional studies were identified by screening reference and citation lists, and by contacting experts.

Study sample: The search yielded 1109 records that were screened on title and abstract. This resulted in the full-text assessment of 37 articles.

Results: Sixteen articles reported on the use of multiple programmes for various listening environments, three articles reported on the use of an automatic switching mode. All studies reported on hearing aid recipients only, no study with CI recipients fulfilled the selection criteria.

Conclusions: Despite the high number of manual and automatically switching multimemory devices sold each year, there are remarkably few studies about the use of multiple programmes or automatic switching modes for various listening environments. No studies were found that examined the accuracy of the use of programmes for specific listening environments. An automatic switching device might be a solution if recipients are not able, or willing, to switch manually between programmes.

Keywords: Hearing aid, cochlear implant, multimemory, multiprogrammable, listening environment, auditory scene analysis, (omni-)directional microphones, scoping review.

Introduction

Daily communication takes place in various situations. It has been well documented that in many listening conditions, speech recognition remains a challenge for recipients of a hearing aid or cochlear implant (CI). Not surprisingly, one of the frequently mentioned situations in which hearing aid recipients seek for hearing improvement is the situation in which speech has to be recognised in noise. A device that improves hearing only in specific situations can be expected to have low overall satisfaction ratings, because of the variety in listening environments a hearing-impaired listener experiences during a day (Kochkin, 2007). Therefore, it is important for recipients of a hearing device that settings are optimally adapted, depending on the specific listening environment, to assure optimal speech recognition.

Current hearing devices allow users to either switch manually or automatically through multiple programmes for various listening environments. In clinical practice, hearing impaired listeners are regularly fitted with these manual and/or automatically switching devices. These devices often have specific programmes with names each representing a specific listening environment (e.g., “speech in noise” and “music”) and the listener is counselled to use the programmes in these specific listening environments. Despite the fact that these devices are prescribed very often, little is known about the actual need for and use of these functionalities. Therefore, a scoping review was performed to identify the available evidence on the use and appreciation of manual and automatically switching devices.

In the past, analogue hearing aids were fitted with a gain-frequency response based on standard prescription rules. These standard rules were often used as a first fit, followed by individual adjustments. The resulting setting had to be used in every listening environment. Research, however, has shown that recipients can benefit from using different hearing aid settings in different listening environments. Van den Heuvel et al. (1997) and Ricketts and Bentler (1992) found better speech recognition if recipients had specific hearing aid programmes for listening in quiet and in noise.

Multimemory hearing aids were introduced that support multiple programmes for use in various listening environments (e.g., hearing in a restaurant or public place, enjoying music, or using the phone). These devices allow the hearing aid user to choose between programmes with different settings for different listening environments. Programme settings used for different listening environments could include variations in overall gain, different frequency responses, directionality of microphones, and noise suppression algorithms. In this scoping review, hearing aids that allow the user to switch between two microphone modes (directional or omnidirectional) are considered multimemory hearing aids as well. In the development of CI speech processors, a technological evolution similar as for hearing aids was observed. Directional and omnidirectional microphones were introduced, followed by multimemory speech processors. Thus, in both types of hearing devices, development resulted in the availability of various programme settings for various listening environments. In the current paper, the term 'multimemory devices' is used to refer to all the devices that allow the user to switch manually between programmes. Often terms as 'multiprogrammable' or 'programmable' devices are used for the same type of devices. With a multimemory hearing device, the user can manually switch between the various programmes as the listening environment changes, either by pressing a button on the device or by using a remote control.

After the development of multimemory hearing aids, devices were introduced that are able to switch between settings based on acoustic scene analysis of the incoming sound. In these devices, the microphone input is analysed by extracting specific signal features, followed by determining the most likely listening environment. Finally, the identified scene is used to decide if and when the current programme has to be changed (Mauger et al., 2014). The aim of this device feature is to optimise settings in all listening environments and to minimise user interaction. Recent studies have shown that automatic scene classification and programme selection in CI speech processors can benefit their recipients (see Wolfe et al. (2015) for an overview).

The current scoping review was limited to studies evaluating multimemory devices that allow the user to manually switch between discrete programmes for specific listening environments, and automatically switching devices that use a classifier to differentiate between specific listening environments and alter settings accordingly. The rationale for focusing on devices with specific programmes for specific listening environments is that patients are usually counseled to use a programme in a specific listening environment. The names of these programmes often represent a specific listening environment (e.g., "restaurant" or "music"). Based on current clinical practice, we started the literature search with several assumptions on the use of manual and automatically switching devices. First, we assume that a manual switching multimemory device has added value when the hearing device user is able to characterise listening environments adequately, select the most appropriate programme, and that he or she is capable in using the switch button or remote control. However, as listening environments may change rapidly, it seems unrealistic to expect that a user switches manually for every change in listening environment. It is assumed that a specific setting for a specific listening environment (e.g., a programme with directional microphone for conversations in a restaurant) generally provides benefit in terms of speech recognition and listening effort compared to a programme that is not optimal for the same listening environment (e.g., a programme for listening to music). Thus, the possibility to manually switch between settings does not unequivocally guarantee the most optimal setting in each listening condition.

The added value of automatically switching devices is based on the assumption that all hearing device recipients prefer a specific setting in a particular listening environment (i.e., specific settings are fixed for a particular listening environment regardless of a person's individual preferences). However, it is possible that the signal processing preferred by one individual in a specific environment might not be the preferred option for another individual in the same environment, or an individual might prefer different processing variants in the same acoustic environment, depending on his or her focus of attention in the given environment.

The research questions for the scoping review were:

1. For multimemory devices (manual switching): Do recipients of hearing aids or cochlear implants use different programmes in various listening environments? If so, do recipients value the possibility to switch between programmes, and do they use the correct programme that is designated for a specific listening environment?

2. For multimemory devices (manual switching): Which factors influence whether a recipient actually uses multiple programmes?

3. For automatically switching devices: Do hearing aid and CI recipients value a device that switches automatically between settings, depending on listening environment?

Methods

Scoping review design

The current scoping review follows the methodology for systematic scoping reviews developed by the Joanna Briggs Institute (JBI). Scoping studies are increasingly being used to review the available literature and identify possible gaps in the available evidence (Davis et al., 2009; Peters et al., 2015). The design of a scoping review differs from that of a systematic review in that it can address broader topics with many different study designs. A scoping review does generally not assess the quality of the included studies (Peters et al., 2015) and is less likely to address very specific research questions (Arksey & O'Malley, 2005). Furthermore, a scoping review adds a narrative integration of the relevant evidence. For the purpose of the current review, it was important that the design allowed inclusion of studies whose primary aim was not related to answering one of our research questions. Therefore, the descriptive style of a scoping review was used to answer the research questions. Given the limited number of studies that met the inclusion criteria, the large differences in study designs and used procedures, the frequent use of subjective measures and differences in outcome measures in the included studies, it was not possible to pool the study findings.

Data Sources

A review protocol was developed to search for evidence in various online sources. PubMed, Embase.com, and ISI/Web of Science were searched by FDG and JCFK from inception up to April 19, 2016. The following terms were used (including synonyms and closely related words) as index terms or free-text words: 'hearing aids' or 'cochlear implants', and 'multiple programmes' or 'settings' or '(omni-)directional microphone' or 'environmental or scene classification'. The full search strategy for each database can be found in Appendix B. This procedure yielded articles in English and German, which were all included for further selection.

Study Selection

From the results of the search, duplicates were removed, after which titles and abstracts were screened for possible inclusion. The first two authors (FDG and EH) independently reviewed titles and abstracts of all records. Studies that clearly did not meet the inclusion criteria were excluded. Full text articles of the remaining records were assessed for eligibility using the selection criteria described below. An additional search was conducted to identify additional records by reviewing the reference and citation lists of the records already included.

A set of criteria, composed by two investigators (FDG and EH), was used for selection of articles. Inclusion criteria were (a) patient related studies concerning adult hearing aid or CI recipients; (b) studies describing the use of multiple programmes/microphone settings, with manual or automatic selection, in various listening environments; (c) studies describing the patients' preference for and the appreciation of the use of one or more hearing device settings in various listening environments; (d) studies describing the patients preference for and appreciation of the use of a hearing device that automatically alters settings according to listening environments. Criteria (a) and (b) had to be met, combined with either (c) or (d). Studies including participants outside of the target age range (18 years and older) were excluded unless the mean age fell within the target age range, or the data could be split for separate analysis. Studies in which speech recognition measures were used to solely evaluate the effect of different settings (e.g., gain or fitting rules) on speech recognition were excluded.

Results and Discussion

Study Characteristics

After removing duplicates, the initial search yielded 1109 articles. Of these, 1072 articles were excluded based on title and abstract, leaving 37 articles for full review. Of these, 13 were included after reading the full-text article. An additional six articles were identified through reference and citation searching, resulting in a total of 19 studies that conformed to the inclusion criteria (Figure 1). Fourteen of these articles were published in peer-reviewed journals, the remaining five articles were published in non-peer-reviewed journals. The study characteristics of the finally included articles are outlined in Table 1.

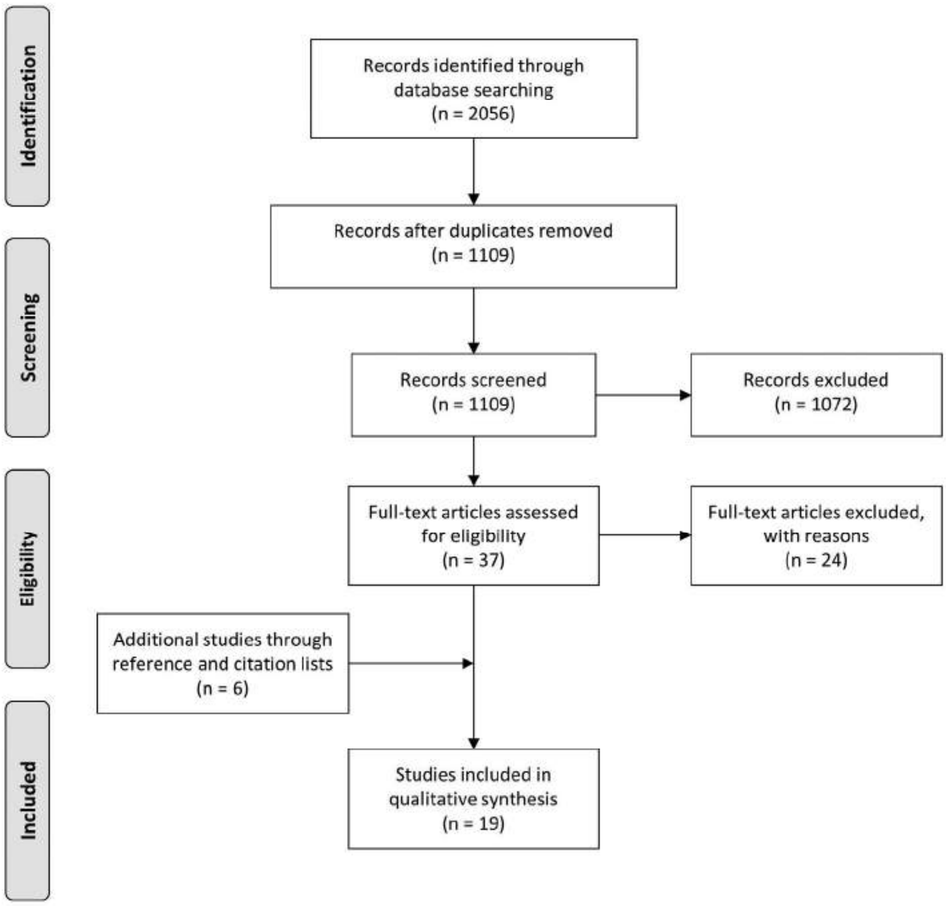

Figure 1. Study selection process.

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Table 1. Study Characteristics.

Citation	**N of participants**	**Age range** (mean/median)	**HL type and severity**	**HA experience**
Ringdahl et al. (1990)*	25	22-77 (65)	Mild-moderate SNHL	Experienced
Goldstein, Shields, and Sandlin (1991)*	10	50-78	Moderate SNHL	Experienced
Kuk (1992)*	19	58-87	NR	Experienced
Stelmachowicz, Lewis, and Carney (1994)*	9	44-71 (60)	Bilateral mild-moderate SNHL	NR
Keidser (1995)*	25	26-76 (71)	Mild-moderate SNHL; conductive HL	Experienced
Keidser, Dillon, and Byrne (1996)*	10	35-77 (72)	SNHL; conductive HL	> 6 mos experience
Kuk (1996)*	95	70	Moderate-severe HL	20% first time users
Berninger and Nordstrom (1997)*	11	41-73 (58)	Bilateral symmetric mild-moderate SNHL	Experienced
Keidser et al. (1997)*	27	NR	Mild-moderate SNHL	> 1 yr experience
Van den Heuvel, Goverts, and Kapteyn (1997)*	126	NR	sloping loss, flat loss, ski-slope loss, irregular loss	NR
Buechler (2001)†	22	63	Moderate HL	NR

*Review question 1; †Review question 3;

HL, hearing loss; Mos, months; NR, not reported; SNHL, sensorineural hearing loss; Yrs, years.

Table 1. Study Characteristics.

Citation	**N of participants**	**Age range** (mean/median)	**HL type and severity**	**HA experience**
Cord et al. (2002)*	48	45-91 (73.6)	NR	6-24 mos experience
Gabriel (2002)†	20	32-81 (60)	Symmetric moderate SNHL	Many yrs experience
Ricketts, Henry, and Gnewikow (2003)*	15	51-74 (61.3)	Symmetric mild-moderately/severe SNHL	Laboratorial experience with HA
Olson, Ioannou, and Trine (2004)†	18	29-75 (64.6)	Mild-severe symmetric SNHL	9/18 new; 9/18 several mos to 10 yrs experience
Walden et al. (2004)*	17	47-81 (70.8)	Bilateral symmetric SNHL	3-72 mos experience
Keidser et al. (2005)*	48	22-92 (75)	Slight moderate-severe HL Monaural and bilateral symmetric SNHL; conductive HL	New and experienced
Palmer, Bentler, and Mueller (2006)*	49	27-85	Bilateral SNHL	18 new (< 60 days); 31 experienced (> 6 mos)
Banerjee (2011)*	9	49-78 (63.3)	Bilateral symmetric mild-moderate SNHL	Experienced

*Review question 1; †Review question 3;

HL, hearing loss; Mos, months; NR, not reported; SNHL, sensorineural hearing loss; Yrs, years.

Overall, 11 studies investigated whether patients use different programmes, depending on the listening environment (Banerjee, 2011; Berninger & Nordstrom, 1997; Goldstein et al., 1991; Keidser, 1995; Keidser et al., 1996; Keidser et al., 2005; Keidser et al., 1997; Kuk, 1992; Ringdahl et al., 1990; Stelmachowicz et al., 1994; van den Heuvel et al., 1997). Five studies investigated the use of different microphone settings for different listening environments (Cord et al., 2002; Kuk, 1996; Palmer et al., 2006; Ricketts et al., 2003; Walden et al., 2004). Three studies investigated whether hearing aid recipients value a device that switches automatically between settings according to listening environments (Buechler, 2001; Gabriel, 2002; Olson et al., 2004). Note that all studies that fulfilled our selection criteria reported on recipients of hearing aids only. No studies were found evaluating the use of multiple programmes in recipients of a CI. The results of the included studies were all based on field and laboratorial tests during which subjects reported on the use of different programmes for different listening environments. The majority of the participants in these studies were experienced recipients of hearing aids (Table 1). Several studies included participants who already had experience with multiple programmes or were successful recipients of switchable microphones. The age of the participants and the severity of hearing loss varies within and across the included studies. Some additional characteristics of the hearing-impaired participants could be identified, such as 'good speech discrimination abilities' and 'good oral communication' (Goldstein et al., 1991; Ringdahl et al., 1990), or a very motivated group of hearing-impaired participants (van den Heuvel et al., 1997).

Question 1: For multimemory devices (manual switching): Do recipients of hearing aids or cochlear implants use different programmes in various listening environments? If so, do recipients value the possibility to switch between programmes, and do they use the correct programme that is designated for a specific listening environment?

The studies showed that there was no general preference for using different programmes. Some participants used and valued the option to have manual access to multiple programmes in various listening environments, while other hearing aid recipients did not use the option to manually switch between programmes. Stelmachowicz et al. (1994) showed that recipients did not select different settings in five simulated environments. In a study by Keidser et al. (1997), only five out of 27 participants used different settings in different listening environments. They also reported that no participant experienced benefit from more than two different settings. Banerjee (2011) showed that the default setting was deemed acceptable by the hearing recipients for the majority of time. Berninger and Nordstrom (1997) showed that the majority of their participants used all four programmes regularly. The main focus of their study was the repeatability of the recipients' preference for a specific setting, i.e., the percentage of participants who selected the same programme as the best one (on individual basis).

In demanding listening environments, this repeatability was high (75%), while it dropped to 40% in less demanding situations.

Studies showed mixed results on the use of different microphone modes (omnidirectional or directional) for specific listening environments (Cord et al., 2002; Kuk, 1996; Palmer et al., 2006; Ricketts et al., 2003; Walden et al., 2004). The majority of participants reported switching between microphone modes based on different listening environments. In contrast, however, subgroups of participants left the hearing aids in the default microphone mode (Cord et al., 2002; Kuk, 1996), were not able to differentiate between different microphone modes (Palmer et al., 2006), or were equally divided in terms of preference between the omnidirectional and directional microphone mode (Palmer et al., 2006).

In conclusion, studies showed that some hearing aid recipients value the option of multiple programmes for various listening environments. However, little is known about the correct use of programmes designated for a specific listening environment. None of the studies examined whether participants used a certain programme in the correct listening environment (e.g., whether hearing aid recipients selected the noise programme in noisy environments).

Question 2: For multimemory devices (manual switching): Which factors influence whether a recipient actually uses multiple programmes?

Several factors seem to be important for the successful use of multimemory hearing aids. First, the hearing aid user needs to meet specific criteria. A set of guidelines was devised (Keidser et al., 2005) to specify which individuals are likely to actually use multimemory hearing aids. These guidelines indicate that subjects (1) must demonstrate motivation for better hearing in listening environments experienced on a regular basis, (2) must be able to understand, manage, and accept a multimemory hearing device, and (3) must be fitted with hearing aid settings in the different programmes that are sufficiently different for the hearing aid user to tell them apart. These guidelines were evaluated in 48 hearing recipients and appeared to correctly identify the majority of the participants who actually used the multimemory hearing aid.

In addition to the characteristics of potential multimemory hearing aid recipients mentioned in the guidelines of Keidser et al. (2005), several other factors could be identified through the included studies. First, recipients need to be able to classify the listening environment, must be willing to subsequently switch to different settings, and recipients need to be physically capable of switching between programmes. In addition, the user needs to experience a clear benefit in switching between programmes (Cord et al., 2002; Kuk, 1996). Recipients who perceive an advantage in switching between different microphone modes are more likely to switch than recipients who do not (Kuk, 1996). No findings were reported about the ability of listeners to assess the listening environment correctly, except for the studies by Walden et al. (2004) and Cord et al. (2002). The data of these studies suggest that patients are able to assess the listening environment correctly and change the microphone mode accordingly. However, it is complicated to conclude about the listeners ability to specifically assess listening environments, due to a difference between listening environment (i.e., acoustic environment) and listening situation (i.e., focus of a hearing device user in an acoustic environment). For instance, a recipient might want to switch between listening to a distant speaker, or focus attention to someone sitting next to him. In this case, the listening environment has not changed, however, a change in settings might be required due to the change in listening situation.

Studies showed that patients tend to leave hearing aids in the default setting (Banerjee, 2011; Cord et al., 2002; van den Heuvel et al., 1997). This is often prompted by the fact that hearing health care professionals may instruct recipients to keep the hearing aids in the default setting, except when speech recognition is affected to such a degree that the settings of the hearing aid should be changed (e.g., in a listening environment with background noise). This may prevent recipients from switching to the most optimal setting when the environment changes. We think that counselling of hearing aid recipients is very important for the appreciation of a multimemory hearing aid use. Attention should be paid to explain the characteristics of the different programmes for specific listening environments. It is shown that patients do not understand, are not aware of, or have forgotten what the specific listening characteristics of an environment are in which programmed hearing aid settings provide substantial benefit (Cord et al., 2002). Therefore, a session could be added to the counselling process in which the recipient experiences examples of listening environments and is trained to choose the designated programme. An additional session might be needed after some time to determine whether the recipient is still using the features of the hearing device appropriately. Although this might be considered necessary for an optimal fitting procedure, it is questionable whether this is a realistic scenario for clinical practice. The possible negative effect of using a multimemory hearing aid while it is in a programme that is not designated for the specific listening environment was not discussed in any of the studies included in this review.

An important factor, also highlighted in the aforementioned guidelines, is that recipients have to find themselves in varying listening environments on a regular basis to experience benefit from a multimemory hearing aid (Keidser et al., 2005; van den Heuvel et al., 1997). However, listening environments vary between individuals. Therefore, it might be important to fit a hearing aid with settings for those listening environments that are pointed out to be important by the individual wearer. Some patients might benefit from volume changes only and do not necessarily need multiple programmes (Banerjee, 2011; Keidser et al., 1997;

Stelmachowicz et al., 1994). In a study in which recipients could alter both the volume and the settings, none of the participants who frequently used the volume control preferred different programmes for different listening environments (Keidser et al., 1997). Stelmachowicz et al. (1994) have argued that it is likely that the efficacy of a multimemory device is highly dependent upon the specific hearing aid characteristics programmed into each memory and how well these characteristics match the user's needs in daily listening environments. Adapting hearing aid settings on an individual basis, however, is very time consuming, and does not guarantee the use of multiple memories in all hearing aid recipients (Stelmachowicz et al., 1994).

In conclusion, a multitude of factors that may influence the (non-)use of multiple programmes, related to the hearing aid recipients as well as to counselling and fitting, could be identified through the included studies. However, the evidence is weak and based on relatively little and older hearing aid literature. It is evident that, because of this vast number of factors, not all hearing aid recipients value the option to manually switch between multiple programmes in different listening environments.

Question 3: For automatically switching devices: Do hearing aid and CI recipients value a device that switches automatically between settings, depending on listening environment?

Three studies that described the use of an automatic switching mode (Buechler, 2001; Gabriel, 2002; Olson et al., 2004) concluded that this functionality was used with satisfaction by the majority (range 75% to 89%) of hearing aid recipients. Despite the usefulness, the automatic mode was shown to be unable to anticipate to individual preferences or correctly classify all acoustic environments (Buechler, 2001). This might be due to the fact that an automatic switching device selects its settings mainly based on specific acoustical characteristics of the incoming sound (e.g., presence or absence of noise). Individuals, on the other hand, may choose settings based on these acoustical characteristics, but also on the specific listening situation.

The majority of subjects in the study by Gabriel (2002) evaluated the automatic switching mode as being useful. The automatic switching mode was used for 70% of the time, a remote control to switch manually between programmes was used the rest of the time. Buechler (2001) showed that in specific situations (e.g., in a group conversation, where one user prefers to focus on the speaker, while another hearing aid user wants to hear other speakers in the group as well), the option to manually choose the appropriate programme was preferred. However, the majority of subjects found the automatic switching mode useful, and only 20% of the subjects indicated that they would rather not use it. Olson et al. (2004) reported that subjects supported the concept of an automatic switching mode, and the subjects indicated that the automatic mode switched to the preferred microphone mode (directional or omnidirectional) about 89% of the time, although the recipients had difficulty to choose between microphone modes in some environments (e.g., a restaurant environment).

The included studies have shown that an automatic switching device might be a good option for individuals who want to have a simple-to-use device, and for individuals who are not able to assess the listening environment adequately and alter the settings of hearing aids accordingly (Buechler, 2001; Gabriel, 2002; Olson et al., 2004). A manual switch would be preferable for individuals who prefer settings that deviate from the selection made by the automatic switching mode (Gabriel, 2002). However, the studies are quite old and may therefore not reflect the possibilities of newer automatic switching hearing devices.

General Discussion

Despite the high number of multimemory devices and automatically switching devices that is sold each year, this scoping review showed that the evidence for the use of these features, the appreciation of these features, and the evidence for the appropriateness of the setting selection is remarkably low. It was recently stated that the hearing aid industry has been emphasizing on development over research (Zeng, 2015). A continuous market pressure to introduce new features often results in the introduction of technologies in an absence of compelling evidence to support the use of these features by the key stakeholders, the hearing aid recipients.

First, it is important to mention that the evidence is weak and based on relatively little and older literature. The number of studies identified through this search was low and the studies have several limitations. These limitations have impeded us to draw strong and consistent conclusions concerning the use and appreciation of manual and/or automatically switching devices. The lack of evidence is surprising, since manual and automatic switching devices are the current state of the art and fitted regularly to hearing impaired patients. Some of the studies are decades old and report on technology that is different from the current technology used in hearing aids and cochlear implants. Thus, the results of these older studies may not reflect the possibilities of newer hearing devices, especially with automatic switching devices. Automatic switching devices have gone through various technical improvements over the last years, resulting in more powerful devices that can address more and more specific listening situations. The principle of manual switching between different programmes for various listening environments has not changed and is independent of technical improvements in hearing devices, since patients still have to be able to characterise the environment, select the most appropriate programme and switch between programmes.

Second, studies report that participants 'benefit' from the use of a multimemory device if they use the feature to manually switch between programmes. However, it is not clear what exactly motivates a participant to use this feature: is he or she experiencing benefit in terms of increased speech recognition abilities when altering settings, or does a change in settings lead to more comfort in listening? As no studies elaborate on this issue, it is unclear which experience underlies the actual use of a multimemory device.

Third, the data on the use of multiple programmes are often subjective. The majority of studies use diaries or questionnaires in which recipients report details of the listening environments (i.e., type of environment, and signal and noise location), the used settings for the specific environment, and whether they altered the settings. The diaries and questionnaires thereby rely fully on, for instance, the ability of participants to distinguish between programmes, the participants' analysis of the environment, judgment of sound quality, and willingness to report every single listening environment. Whether the appropriate programme is used in each listening environment cannot be derived from these data. In contrast, the availability of datalogging in a hearing device, as used in some studies (Banerjee, 2011; van den Heuvel et al., 1997), provides objective information on the hours of daily usage, relative amount of time each programme is used, and volume changes. Kießling et al. (2007) state that subjects are able to provide a reliable estimation of their daily hearing aid use, as compared to the hearing aid use acquired with datalogging. However, this study also found that the correlation between self-reported and logged device adjustments, such as programme changes, is considerably less strong. Datalogging can offer an objective view of the use of multimemory hearing aids, next to the subjective information as used in the majority of the studies.

Fourth, in several studies (Banerjee, 2011; Ringdahl et al., 1990), the participants had to select the preferred setting for a specific listening environment, also referred to as 'forced choice by direct comparison', even while the recipients did not necessarily have a preference for one setting over another. Although this method can be considered to be more reliable than retrospective interviews or questionnaires, it can lead to unreliable data since it cannot be verified that participants have tried multiple settings, or have compared their own hearing aid to the study hearing aid.

Fifth, some studies (Kuk, 1992; Ringdahl et al., 1990) used a study design in which a new hearing aid was compared to the participant's own hearing aid. This study method introduces bias, because new hearing aid technology is generally rated more positively than conventional technology (Ruth A. Bentler et al., 2003; Dawes et al., 2011). In addition, it is unclear whether subjects have followed the study instructions of Kuk (1992) and Ringdahl et al. (1990) and duly have tried multiple programmes in various listening environments, or have only compared their own hearing aid to the default setting of the study hearing aid.

A sixth limitation is that most studies pay little attention to continuation of the use of the features after completion of the study. It is plausible that subjects use multiple programmes during the study period, but stop using them once the study finishes. Kuk (1992) used a follow-up period of one year to investigate the long-term use in 19 participants. He showed that the number of participants using multiple programmes increased from 9 to 11 during the follow-up period. This might indicate that participants need to acclimatise to the use of a multimemory hearing aid. The results of a study by Gatehouse (1992) showed that it takes time, ideally 6 to 12 weeks, to experience the advantages of a new frequency response setting. In the studies discussed in the current article, the hearing aid experience of included participants ranged from no experience with hearing aids to extensive experience with hearing aids and even extensive experience with multimemory hearing aids. It goes without saying that new hearing aid recipients need time to familiarise themselves with hearing aids in general, let alone a multimemory hearing aid that requires them to analyze the listening environments and change the hearing aid settings accordingly.

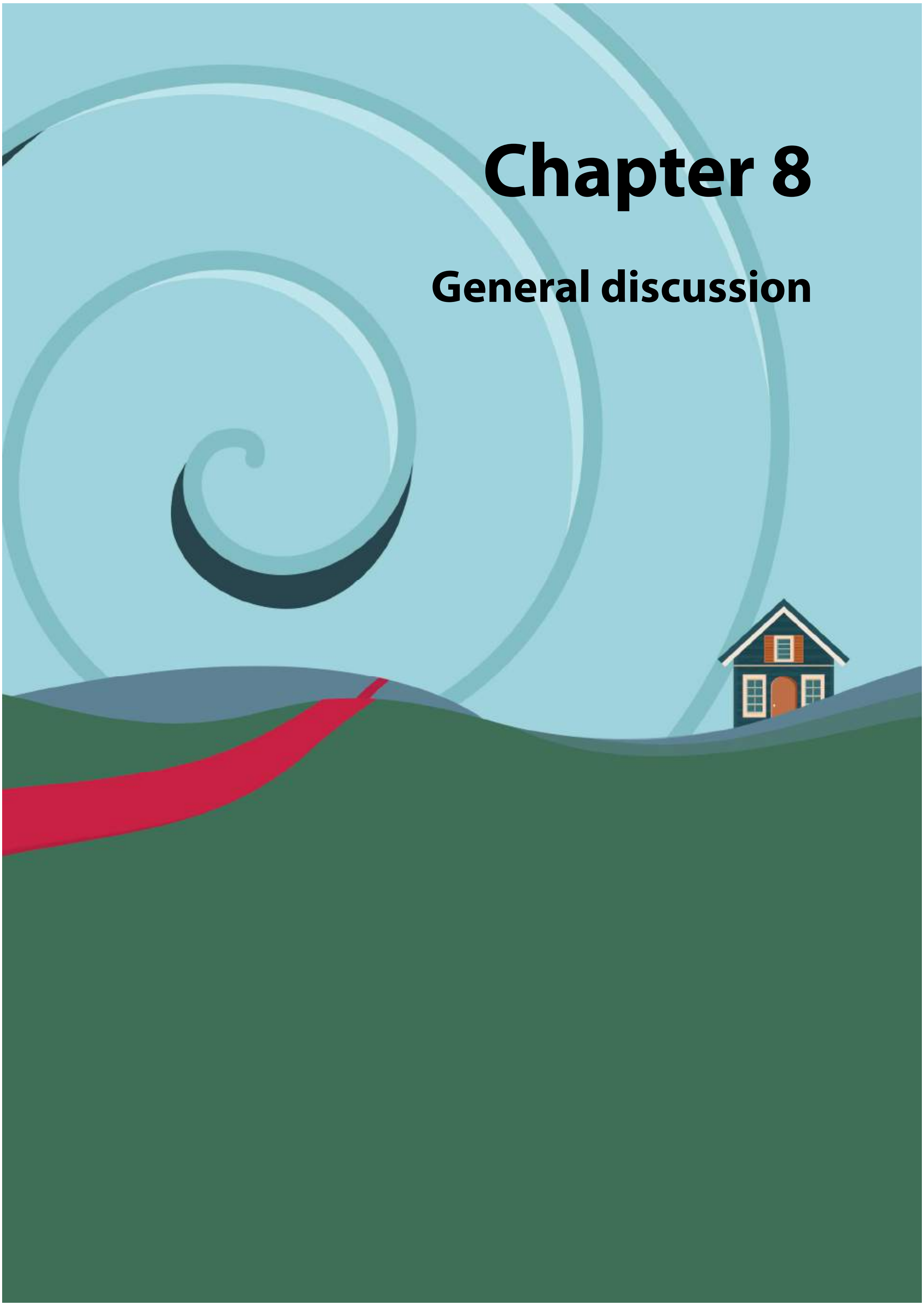
Seventh, several studies (Goldstein et al., 1991; Keidser, 1995; Stelmachowicz et al., 1994) have investigated the use of a multimemory hearing aid in a laboratorial setting. Listening environments in daily life may change rapidly and are not comparable to situations in a sound-treated test booth. In addition, participants in a laboratorial setting might be more aware of the fact that they need to change programmes than when using a hearing aid in daily life. Punch et al. (1994), for instance, have shown that preferences of hearing aid recipients in laboratory sessions were only fair predictors of preferred hearing aid settings in the real world. However, test conditions that simulate the hearing aid wearer's daily listening environments in a laboratorial setting might help to evaluate the candidacy for the use of multiple programmes for different listening environments.

Conclusions

This scoping review investigated the available evidence for the added value of manual and automatically switching devices for various listening environments by hearing aid and CI recipients. No studies were found that concerned CI recipients, so this review only reflects the results of studies in hearing aid recipients. It is unclear to what extent the results of studies with hearing aid recipients hold for CI recipients, who often have more severe hearing losses than hearing aid recipients. Nineteen studies were included in the review, from which 14 were published in peer-reviewed journals.

The scoping review shows that, despite the high number of multimemory devices and automatically switching devices that are sold each year, the evidence for use and appreciation of these features, and the evidence for the appropriateness of the setting selection is remarkably low. Although the evidence is weak, this review indicated that some hearing impaired individuals use the possibilities of a multimemory hearing aid. No studies were found that objectively examined the accuracy of the use of specific programmes for specific listening environments by hearing aid recipients. Several characteristics of hearing aid recipients could be identified that possibly influence the use of a multimemory device: the hearing aid user must indicate a clear need for better hearing in various, often encountered, listening environments, must understand the use of a multimemory device, must be willing to change the settings, and must be able to assess the listening environments. In addition, the programmes of the hearing device must be sufficiently different for the user to tell them apart, the user has to be aware of the different programmes, the user has to experience benefit from the various programs in different listening environments, and must be able to use either a switch button or remote control to change the settings. For those who are not able, or willing, to switch between programmes in different listening environments, an automatic switching device might be a good solution. For others, satisfying results with an automatic switching device can be obtained in combination with a manual switch.

The use of a scoping review allowed the inclusion of a broad range of literature to synthesise the available evidence and identify interesting topics for future research. Based on this scoping review, we recommend that future research should focus on gathering objective evidence concerning the use of multiple programmes for various listening environments by hearing aid and CI recipients. Datalogging functionalities in modern hearing devices, hearing aids and CIs, could be useful to gain further insight in the listening environments encountered (Busch et al., 2017), and in the programmes a user selects in these environments.



Chapter 8

General discussion

The studies in this thesis aim to improve the clinical care pathway of adult cochlear implant (CI) users. The development, validation and clinical use of self-administered speech recognition tests were described in the first section of this thesis. The subsequent section described the prediction of speech recognition in quiet and in noise from fitting parameters and objective measures that are available to a clinician during a fitting session. The final section described the use of multimemory and automatically switching devices by CI users. The main findings of the studies are discussed in this final chapter. Furthermore, suggestions for future research and clinical implications are presented.

Speech recognition assessment 1

Currently, speech recognition tests are performed in the clinic in a sound-treated booth with calibrated equipment to create a controlled test environment. A clinician is needed to administer the tests and judge the responses given by the CI user. In the studies described in this thesis, we developed and evaluated the use of self-administered tests for CI users to assess their speech recognition at home. The home test setup comprises a tablet computer and an audio cable to directly present calibrated stimuli from the tablet computer to the CI sound processor. The tablet computer eliminates the need for a clinician to administer the tests, because the tests can be self-administered by the CI user and the correctness of the response is judged by an algorithm. The audio cable avoids the influence of loudspeaker quality, background noise and reverberation at the CI user's home, thereby providing a controlled test environment and eliminating the need for a sound-treated booth like in the clinic. We have shown that the home tests are technically possible (*chapter 2*), that they are a valid alternative to tests in the clinic (*chapter 3*) and that CI users are generally positive about the possibility to perform self-administered speech recognition tests at home (*chapter 3 and 4*). Furthermore, we have shown that the home test setup can be used to assess other aspects of speech recognition as well, such as binaural speech recognition in noise of bimodal and bilateral CI users (*chapter 5*).

Several studies addressed the remote or self-assessment of speech recognition (Cullington et al., 2018; Cullington & Agyemang-Prempeh, 2017; Goehring et al., 2012; Hughes et al., 2012). Cullington and Agyemang-Prempeh (2017) used the remote speech recognition tests in combination with a questionnaire to identify CI users who would require intervention in the clinic. In a subsequent study, the tests were incorporated in a remote care tool (Cullington et al., 2018). In line with the studies in this thesis, the authors concluded that the majority of CI users are able and willing to assess their speech recognition at home, and that the frequent assessment of speech recognition can identify CI users who require intervention in the clinic. Goehring et al. (2012) and Hughes et al. (2012) compared speech recognition assessed at a remote location with a videoconference setup with regular testing in a soundbooth and found poorer speech recognition scores at the remote sites. Background noise and reverberation were the most important factors that had a negative influence on speech recognition at the remote setting. The use of an audio cable diminishes the possible negative effects identified in the studies by Hughes et al. (2012) and Goehring et al. (2012), which is of paramount importance to prevent variation in scores as a result of the test environment. However, the functionality of the microphones cannot be assessed when using an audio cable, which is a disadvantage. Speech recognition assessed with an audio cable might therefore not fully represent speech recognition in daily life, but it might give a better indication of the potential speech recognition abilities of CI users than speech recognition assessed with a loudspeaker.

The speech recognition in noise scores obtained with an audio cable were better than speech recognition in noise scores obtained with a loudspeaker (*chapter 3*). In addition, several CI users obtained speech recognition in noise scores with an audio cable that were similar to speech recognition in noise scores of normal-hearing listeners (*see Figure 2 in chapter 3 and Figure 1 in chapter 4*). These findings are encouraging, but also suggests that sound processing is different for stimuli presented via the microphones (i.e., when a loudspeaker is used) or audio cable. We conducted some additional pilot testing to find explanations for this difference. As a first step, we explored the sound processor settings of the experienced CI users (who participated in the study presented in *chapter 3*) to find a possible link between settings and speech recognition in noise scores. However, due to the limited number of participants and the wide variety in sound processor settings (i.e., microphone directionality and noise reduction algorithms), no clear reason for the observed difference in speech recognition in noise scores could be identified. Subsequently, a pilot study was performed with 12 adult CI users in which the SNR-NR noise reduction algorithm was consecutively switched on and off. The tests were performed with continuous and discontinuous noise and with an audio cable. The results showed that speech recognition in noise was better for tests with continuous noise than for tests with discontinuous noise, which was in contrast to the results presented in *chapter 2*. Possibly, this is because the SNR-NR algorithm is more effective for higher SNRs, thus for CI users with poorer SRTs (Dawson, Mauger, and Hersbach, 2011). The study presented in *chapter 2* included experienced CI users with relatively good (i.e., more negative) SRTs. Thus, the SNR-NR algorithm might have had no effect on the speech recognition in noise scores in these CI users.

The study with newly-implanted CI users (*chapter 4*) allowed us to further investigate the difference in speech recognition in noise scores obtained with a loudspeaker (clinic testing) or audio cable (home testing). The comparison revealed even a larger mean difference than in the study with experienced CI users (3.2 dB versus 1.6 dB previously). The tests in the clinic in this study, however, were conducted with a different type of masking noise (i.e., discontinuous) than the masking noise used in the self-administered tests (i.e., continuous) from *chapter 2*. It was concluded that the scores obtained with the self-administered tests (i.e., audio cable and continuous noise) cannot be directly compared to scores obtained in the clinic (i.e., loudspeaker and discontinuous noise). If scores are to be compared, a reference measurement in the clinic with the home test setup is required. Based on these findings, we expect that the differences in speech recognition in noise scores are caused by a complex interaction between speech recognition scores, processor settings, noise reduction algorithms, silences in the masking noise and stimuli presentation mode. It has been demonstrated that some of the advanced sound processing features incorporated in modern CI devices (e.g., noise reduction and adaptive algorithms) improve speech recognition (see Wolfe et al. (2015) for an overview). However, it is unknown to what extent these advanced sound processor settings influence speech recognition scores when obtained with an audio cable.

It is likely that many factors make some unique contribution to the difference in speech recognition in noise scores assessed with the loudspeaker and those assessed with the audio cable. The acoustics of the soundbooth may have a slight negative effect, and larger differences could arise between individuals due to head movements and differences in head diffraction. Further studies are required to investigate the causes of the differences in speech recognition in noise scores between the clinic and home tests.

Further research

Although the current self-administered test setup provides several advantages over the setup of the tests in the clinic, several aspects need to be dealt with before the tests are a viable alternative for tests in the clinic. First, the usability of the setup has to be improved to increase the number of CI users who will be able to use the technology required. The current setup of the self-administered tests is not suitable for all CI users, which was reflected by 2 out of 10 newly-implanted CI users who did not feel confident in using the technology required (*chapter 4*). Thus, it remains important to identify those CI users who are not able to use the technology required and provide them with care as usual. Whilst experienced CI users did not report any problems (*chapter 3*), one of the major complaints concerning the usability of the home tests in newly-implanted was the difficulty in connecting the audio cable to the sound processor and tablet computer. It is well known that newly-implanted CI users are less confident in handling their CI and are therefore somewhat reluctant in using accessories such as the audio cable. The usability of the home tests is expected to improve, if stimuli can be presented without the need of the audio cable that has to be connected. One of the recent developments that could lead to an improvement is the direct streaming of stimuli which is incorporated in the newest CI sound processor of Cochlear™.

Digital streaming could have an additional advantage if stimuli could be presented binaurally to bilateral CIs or a CI and contralateral hearing aid to assess binaural speech recognition of bilateral and bimodal CI users.

Second, to implement the self-administered tests in clinical care of CI users, the tests should be made available to users of other CI brands as well. Currently, the self-administered tests are limited to users of specific Cochlear™ sound processors, because of the need for an accessory socket to connect the audio cable. Likewise, the self-administered tests only allow binaural speech recognition assessment of bimodal CI users with contralateral hearing aids that have direct audio input. Furthermore, the tests have been developed for use on a Windows tablet computer. The accessibility of the tests can be improved further if the tests are made available for other platforms as well, such as laptops or mobile phones.

Third and final, further research is necessary to identify the causes of the difference in speech recognition in noise when assessed with an audio cable or loudspeaker. The first step would be to investigate possible differences in the signal that is presented to the internal receiver if stimuli are presented via an audio cable or loudspeaker. Subsequently, the influence of different sound processor settings, such as microphone directionality or noise reduction algorithms, or individual differences, such as head movements, should be systematically investigated. Possibly, this could lead to improvements in the signal processing and will thereby hopefully result in improvements in speech recognition performance of CI users.

Future perspectives

More frequent assessments of speech recognition that is possible with the home tests will provide clinicians with a far more detailed insight in both the current performance as well as the progression in speech recognition than currently available. This insight enables early identification of CI users for whom speech recognition performance deteriorates or does not improve as expected. For these CI users, auditory training can be intensified or appointments can be scheduled to fit the sound processor. This insight could also be used to reduce the number of visits and associated time spent in the clinic for those CI users for whom speech recognition is satisfactory and who therefore do not need to bring extra visits to the clinic. Thus, the information gathered with the home tests can be used to schedule visits based on the clinical need and patient's requirements, as opposed to the current fixed schedule (i.e., 10 visits for a total of 20 hours within the first year after implantation). Individualized scheduling will most likely result in a significant reduction in the number of visits to the clinic, thereby lowering the demand per CI user on CI centres, whilst the information available to clinicians is more detailed than before.

The self-administered tests do have more potential than solely a reduction in the number of visits. For instance, clinicians can provide CI users with multiple sound processor settings. Subsequently, CI users can perform speech recognition tests with the different settings to find out which settings provide optimal speech recognition. The self-administered tests can also be used to assess speech recognition more often than the current annual assessment in the clinic. For instance if the CI users or their friends and family have doubts about their own speech recognition performance or do want to track their own performance over time. This might also be of use for clinicians, because the current annual assessment of speech recognition in the clinic only provides snapshots of performance.

From a CI user perspective, the self-administered tests allow CI users to assess their speech recognition, from the comfort of their home, more frequently and independently of visits to the clinic. The home tests provide insight in their progression in speech recognition, which has not been available to CI users so far and might motivate them to improve their speech recognition further. If appointments are scheduled based on their individual's needs, the possible reduction in the number of visits could lead to a significant reduction in time spent away from work or family. Not only because they spent less time in clinic, but also in time spent travelling to their CI centre. This might not be as big of a problem for CI users in The Netherlands, however, when considering other countries such as Australia, where clinics might be located hours away from the CI users' home, this can have a significant impact on the CI user's life.

The interest in the use of telehealth technologies within the hearing healthcare field is growing (Paglialonga et al., 2018). Recently, Bush et al. (2016) identified 12 studies that present on the remote delivery of CI care, including intraoperative testing (Shapiro et al., 2008) and programming (Botros, Banna, and Maruthurkkara, 2013; Eikelboom et al., 2014; McElveen et al., 2010; Ramos et al., 2009; Wesarg et al., 2010). Although the interest in the use of these technologies is growing, they are only applied to a limited extent world-wide and are currently not part of care as usual for CI users in the Netherlands. Further research is needed to increase the penetration and efficacy of telehealth technologies in clinical practice. Telehealth applications, such as the MyHearingApp (*chapter 4*) in which the self-administered tests were incorporated, have the potential to facilitate self-care of CI users and thereby increase their involvement in their care. If parts of the CI care can be relocated to the CI user's home, this can further decrease the demand per CI user placed on the CI centre. The functionalities currently implemented in the MyHearingApp (Philips et al., 2018) could be combined with other aspects of CI care, such as exercises for auditory training, remote fitting, and assessments of device and electrode functioning (e.g., impedances and audiometry).

Considerations for clinical practice

The development of the self-administered speech recognition tests provided several useful insights to be considered for current clinical practice. The first consideration is related to the masking noise used in speech recognition in noise testing. The standard clinic tests to asses speech recognition in noise use discontinuous noise, with quiet periods between stimuli. Modern CIs contain advanced sound processing features (i.e., noise reduction and adaptive algorithms), that are relatively slow-acting; with time constants in the order of seconds. The quiet periods between stimuli in the discontinuous noise prevent these features to become fully active, thereby possibly affecting speech recognition in noise scores. Therefore, the effect of another type of masking noise (i.e., continuous noise) for speech in noise testing was investigated in this thesis. Here, the noise is presented continuously throughout the test. Although we expected to find better speech recognition in noise scores with continuous noise, no significant differences in speech recognition between continuous and discontinuous noise were found for both normal-hearing individuals and CI users (*chapter 2*). In a different group of CI users (*chapter 4*), however, we found a mean difference of 3.2 dB between clinic tests with a loudspeaker and discontinuous noise and the self-administered tests with an audio cable and continuous noise in a group of newly-implanted CI users. This suggests that discontinuous noise does influence speech recognition in noise scores, because the difference is larger than the difference between tests with a loudspeaker and audio cable reported before (1.6 dB in experienced CI users). However, the tests in experienced users were conducted with continuous noise, and, as previously mentioned in this chapter, it is currently unknown what the exact underlying cause of the difference in speech recognition in noise scores assessed with the loudspeaker and audio cable is.

The second consideration is related to the word lists (NVA lists) used for the assessment of speech recognition in quiet with monosyllable words. The speech recognition in quiet tests use lists of 12 words. In total, there are 45 lists available, but only 15 lists contain unique words. The remaining lists contain the same words as the first 15 lists, but in a different order. The results of *chapter 3* revealed that there are large differences in scores obtained with different word lists, and therefore suggest that the lists are not equally intelligible for CI users. Based on these results, the lists used for the speech recognition in quiet tests in newly-implanted CI users (*chapter 4*) were chosen automatically and at random. In clinical practice however, word lists are chosen by the clinician. If, by incidence, speech recognition is assessed with less intelligible lists of words, this might result in lower scores. This would suggest a deterioration in speech recognition in quiet, while it might actually be caused by the differences in word lists. It is therefore recommended to only use lists with equal intelligibility, or to normalize the lists for speech recognition assessment in CI users.

Fitting of the sound processor

The fitting and fine-tuning of the CI sound processor is important to achieve optimal speech recognition performance for CI users. Several important parameters that predict speech recognition in quiet and in noise were identified (*chapter 6*). The findings of the prediction models led to the following clinical recommendations for CI users with late onset of severe hearing impairment (i.e., onset after the age of seven years): (1) assess sound-field aided thresholds and adjust T levels if the mean aided thresholds are higher than the target of 25 dB SPL, (2) set T levels at threshold and increase C levels to ensure a large dynamic range, preferably as large as 40-60 CL, and (3) be aware of impedance profiles across the array with high variation. For CI users with early onset of severe hearing impairment (i.e., onset before the age of seven years) higher T levels were associated with worse speech recognition in quiet and in noise. However, it is not recommended to lower T levels in this group of CI users, because the higher T levels are most likely related to the duration of deafness (i.e., less surviving ganglion cells along the cochlea) in this group of CI users. The predictors of speech recognition in quiet and in noise were largely similar. Thus adjusting fitting parameters to optimize speech recognition in quiet will most likely also result in an improvement of, or will at least not be at the expense of, speech recognition in noise.

Further research

The generalizability of the results to other populations of CI users is limited. For instance, CI users with prelingual onset of hearing impairment or deviating MAP parameters were excluded. Vaerenberg et al. (2014) have shown that fitting practices across different CI centres vary widely. Therefore, the results of the current study (*chapter 6*) are not only limited to the CI population studied in this thesis, but may also be different for CI users fitted in other CI centres. Therefore, it is recommended to investigate the importance of the identified predictors in CI users of other CI centres, as well as CI users outside the homogeneous group that was studied in this thesis. Two suggestions for setting up such a study are: (1) the prediction model described in this thesis can be externally validated in a dataset with other CI populations or CI users from other CI centres, or (2) the procedure to build the prediction models as described in this thesis can be used to identify predictors of speech recognition in CI users that were excluded from the current study or CI users from other CI centres.

In addition to the generalizability of the results, it is important to assess the clinical relevance of the findings of this thesis. The predictors that were identified can be used to select a group of CI users in which an improvement in speech recognition could be expected when certain fitting parameters are changed. The group has to be selected based on their fitting parameters, specifically if these parameters deviate from the fitting parameters identified through the prediction model. Subsequently, a cross-over study design (e.g., A-B-A) can be used to assess speech recognition with the old fitting (i.e., A), and new fitting (i.e., B) based on the prediction model. Such a study is needed to proof that the predictors that were identified can be used to improve speech recognition of individual CI users.

Future perspectives

Vaerenberg et al. (2014) identified many different fitting practices across CI centres. In addition to the difference in fitting practices across CI centres, there are no fitting rules available to clinicians, comparable to the ones used in hearing aid fitting (e.g., NAL and DSL prescription rules). The identification of important predictors of speech recognition may guide audiologists in their fitting practices and improve the performance of CI users. However, the combination of the differences in fitting practices and the lack of targets make it difficult, or even impossible, to compare outcome measures and to judge which settings yield the best results. Defining targets and outcome measures might be a next step to optimize fitting practices and improve outcomes. The Fitting to Outcomes eXpert (FOX) system (Govaerts et al., 2010; Vaerenberg et al., 2014; Vaerenberg et al., 2011) is an example of a software application that suggests adjustments to the fitting, based on target outcomes. In that case, target outcomes can be set by a clinician and, if targets are not met, changes to the fitting can be suggested. Systems like FOX may provide effective tools to set targets and optimize fittings, which can help clinicians to improve outcomes.

The use of manually and automatically switching programs

It is well known that speech recognition remains a challenge for CI users in many of the daily encountered listening situations. Therefore, CI users are often fitted with manual and/or automatic selection programs for various listening situations. A review of the literature (*chap-[illegible]r [illegible]*) revealed that there is remarkably little evidence available on the use and appreciation of these features in users of hearing aids, whilst no studies were identified that included CI users. The review indicated that some hearing impaired individuals use the possibilities of manual switching hearing devices, and that an automatic switching device might be a good solution for those who are not able or willing to manually switch between programs for various listening environments. For others, satisfying results can be obtained with an automatically switching program in combination with manual programs. Through the scoping review (*chapter 7*), several characteristics of CI users who would potentially benefit of multimemory devices could be identified: (1) users must indicate a clear need for better hearing in various, often encountered, listening environments, (2) users must understand the use of a multimemory device and be able to use either a switch button or remote control to change settings (3) users must be able to assess the listening environment and change settings accordingly, and (4) users must be aware of the different programs.

The findings of the scoping review were used to design an experimental study to investigate the use of manually and automatically switching programs for various listening environments by 15 adult CI users. A cross-over study design was used in which Cochlear™ CI users alternatively used two manual programs or one automatic program for three weeks. Datalog information is stored on the sound processor. This datalog information contains information about the listening environments classified by the automatic program, and the use of the manual programs. This information was used to investigate whether CI users select the appropriate program in specific listening environments. In addition to the datalog information, the experiences with and preferences for either manual or automatic program selection were evaluated. 1

The automatic program incorporated in modern Cochlear™ sound processors classifies six listening situations (i.e., music, wind, speech, speech in noise, noise and quiet) and alters the microphone directionality (i.e., omnidirectional, fixed or adaptive directional) based on the identified listening environment. For the experimental study, the manual programs were based on the selections of the automatic program. However, only two programs were included. One program was included for quiet listening environments with the omnidirectional microphone (i.e., listening environments classified as music, speech, and quiet). The second program was included for noisy listening environments with the directional microphone (i.e., listening environments classified as wind, speech in noise, and noise).

An extensive counselling session was performed at the beginning of the three week study period in which the participants used the manual programs. In this counselling session, attention was paid to several of the characteristics identified through the scoping review (see above). First, the characteristics of the different programs were explained. Subsequently, the participants were exposed to a variety of listening situations to train them in assessing the situation and select the most appropriate program. The listening situations were presented in a room where the participant was surrounded by eight loudspeakers. Examples of the simulated listening situations are: having a conversation with someone in a quiet listening environment (i.e., target speaker presented at 0°) and having a conversation with someone in a busy restaurant (i.e., target speaker presented at 0° and four interfering speakers presented at 45°, 135°, 225°, and 315°). Finally, a listening situation was presented to the participant to let them experience the difference between the speech in quiet (omnidirectional) and speech in noise (directional) programs. Thus, at the end of this session, participants were (1) expected to understand the use of the manual programs and to be able to switch between them, (2) able to assess quiet and noisy listening environments and choose the appropriate program, and (3) aware of the difference between the two programs.

An important aspect for the use of manual programs for various listening environments is that the programs provide benefit in different listening environments. Therefore, speech recognition in noise was assessed to examine the benefit of the adaptive directional microphone over the omnidirectional microphone in a listening environment with spatially separated speech and noise. As expected, all participants except for two, performed better with the adaptive microphone directionality than with the standard microphone directionality (mean SRT -6.2 dB SNR versus -4.0 dB SNR, respectively. See Figure 1).

The datalog information of the participants showed that they left their CI in the default setting (program 1) for 61% of the time when they had two manual programs available (left panel of Figure 2). This is in line with previous studies that have shown that hearing device users tend to leave their devices in the default setting (Banerjee, 2011; Cord et al., 2002; Searchfield et al., 2018; Van den Heuvel, Goverts, and Kapteyn, 1997). The omni- and directional programs were randomly assigned to either program 1 or program 2. Thus, half of the participants had the program for quiet environments in program 1, whilst the other half had the program for noisy environments in program 1. The datalog information shows that the program for quiet environments is used 60% of the time (middle panel Figure 2).

Thus, despite the counselling session, participants mainly used program 1. Several reasons were mentioned by the participants. The first and foremost mentioned reason was that programs were not sufficiently different, which was also identified as an important factor in the scoping review (*chapter 7*). Some participants mentioned that there was a noticeable difference between the programs directly after switching, but that the difference was less noticeable after some time. Another frequently mentioned reason was that participants reported that they did not find themselves in varying listening environments on a regular basis. They reported that they spent most of their time in quiet listening environments and refrain from noisy situations because of their hearing impairment. The datalog information revealed that participants indeed spent most of their time in quiet environments (70.7%), compared to noisy environments (29.3%) (right panel Figure 2).

At the end of the study, 10 out of 15 participants preferred the automatic program selection. The main reasons that were reported were mostly related to the ease of use, because the CI assesses the listening situation and changes the settings accordingly, without CI users having to worry about selecting the appropriate program and constantly being reminded of their impairment. The remaining five participants preferred the manual selection of programs. These participants reported to prefer to be in control of their settings, because they felt that the automatic switching program did not always choose the appropriate settings.

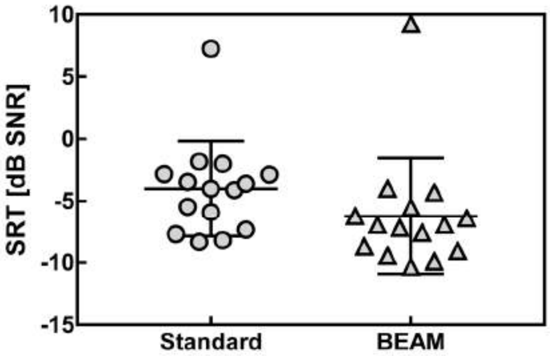


Figure 1. Speech recognition in noise with standard (i.e., omnidirectional) microphone directionality (circles) and BEAM (i.e., directional) microphone directionality (triangles).

In conclusion, as already substantiated by the results from the scoping review, the questionnaires used in the current study showed that an automatic program, either in combination with manual programs, might be a good solution for the majority of CI users. The manual programs are valued by CI users who want to have control over their own device. Further investigation of the data will be performed to investigate whether CI users are able to select the most appropriate program for specific listening situations. This investigation is of paramount importance for the final recommendations about the use of manual programs.

Further research

The listening goals of CI users in the listening environments encountered are not taken into account in the selection of settings by the automatic classifier. This was also illustrated by several participants of the experimental study, who reported to have difficulties with having a conversation while an airplane flew over. In that case, the automatic classifier registered the noise coming from the airplane and subsequently changed to the directional microphone which impaired intelligibility of the speech in that specific listening situation. Further research is necessary to optimize the classification of the automatic program. Until then, a combination between the automatic program and customized manual programs seems valuable for CI users who want to have control over their device and encounter listening environments that are not properly classified by the automatic program. However, as mentioned above, further investigation of the data is required before the final recommendations on the use of automatic versus manual switching programs can be made.

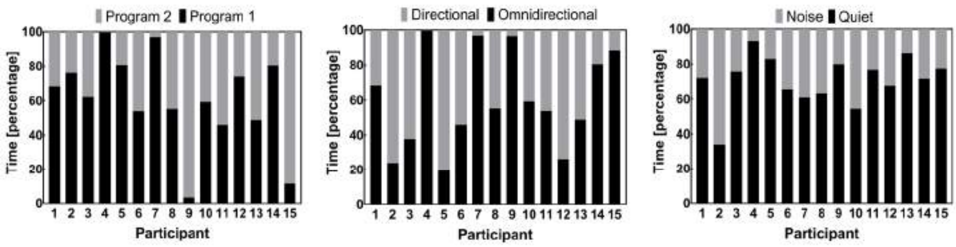


Figure 2. Overview of the program selection by participants (left panel), microphone setting used (middle panel) and encountered listening environments (right panel).

In our study, the volume and sensitivity control were disabled to prevent CI users from creating additional differences between the two manual programs. Thus, the settings evaluated in the experimental study (i.e., only two programs, and fixed volume and sensitivity and sound processing features) only represent a subset of the possibilities of modern CIs. Future research could evaluate the use of the advanced sound processing features, for instance to create additional differences between programs for various listening environments. Furthermore, the volume and sensitivity control could be enabled for some CI users, because they might benefit from volume or sensitivity changes only and do not necessarily need multiple programs. However, research in this thesis has shown that volume and sensitivity changes can influence speech recognition outcomes negatively (*chapter 6*). Therefore, the impact of adjustments to the volume and/or sensitivity should be emphasized to the CI user.

Several aspects identified to be important in the scoping review could not be addressed in the experimental study and therefore require further study. First, the experimental study comprised two periods of three weeks each, to evaluate the automatic and manual programs. Although all participants were experienced CI users, they might need time to acclimatize to new settings. Also, CI users might be aware of the different programs in the context of the study, but might forget what the different programs are for once the study is finished. Therefore, it is recommended to evaluate the long-term use of the automatic and manual programs. This was well illustrated by one of the participants who used four programs prior to the study, but left her CI in one of them when she forgot what the different programs were for.

Second and final, it is well known that laboratory settings do not optimally reflect daily life situations. For instance, the difference between the two manual programs might be very clear in the controlled environment of a sound-treated booth, but might not be so obvious when listening in daily life. Furthermore, the counselling session was done in a room with eight loudspeakers, in which daily life situations were simulated. Examples of these simulated situations were having a conversation with one person in a quiet environment, or talking to someone in a busy restaurant. Although it is very difficult to create listening environments in a laboratory setting that are truly representative of daily life environments, they might help CI users to get familiarized with different listening environments (i.e., quiet and noisy listening environments) that are represented by the manual programs. Alternatively, recordings of daily life situations can be used.

Future perspectives

It is suggested to provide CI users with an automatic program, unless they indicate a clear need for manual programs to accommodate for specific listening environments. Then, either a combination of automatic and manual programs or only manual programs can be provided. However, CI users should be clearly instructed on the settings they are provided with, even if it only concerns the automatic program. CI users should be counselled on the use of the programs in various listening environments and should be familiarized with the difference between the programs. A counselling session can be done within 10 minutes, and is considered important for the use and appreciation of the different programs. Therefore it should be implemented in current care. Finally, the use of multiple programs should be regularly evaluated, to prevent CI users from forgetting what the different programs are for.

General conclusions

The rehabilitation after cochlear implantation is lifelong and includes the assessment of outcome measures and the fitting or fine-tuning of the sound processor. Our findings indicate that self-administration of speech recognition tests at home is a viable alternative to administration of tests in the clinic. With this improvement, CI centres can identify and spent their resources on those CI users who require intervention in the clinic, and reduce the number of visits for CI users who do not have a clinical need. Although the current self-administered test setup provides several advantages over the setup of the tests in the clinic, further research and development is needed prior to implementation in clinical care of CI users. The usability and availability have to be improved to increase the number of CI users who will be able to use the self-administered tests, and further investigation is required to investigate the differences in speech recognition in noise scores between clinic and home tests.

The mean aided thresholds, mean electrical dynamic range, mean T levels, and measures to express the impedance profile across the electrode array were identified as predictors of speech recognition in quiet and in noise. The identification of these predictors of speech recognition in quiet and in noise can be used by clinicians and CI centres to improve their fitting practices and subsequently improve the performance of CI users. Future research should assess the clinical relevance of predictors identified in this study.

The automatic program, or a combination of manual and automatic program selection, will provide satisfactory results for the majority of CI users. It is therefore suggested to provide CI users with an automatic program, unless they indicate a clear need for manual programs to accommodate for specific listening environments. Further research is required for the final recommendations on the use of automatic versus manual switching programs.

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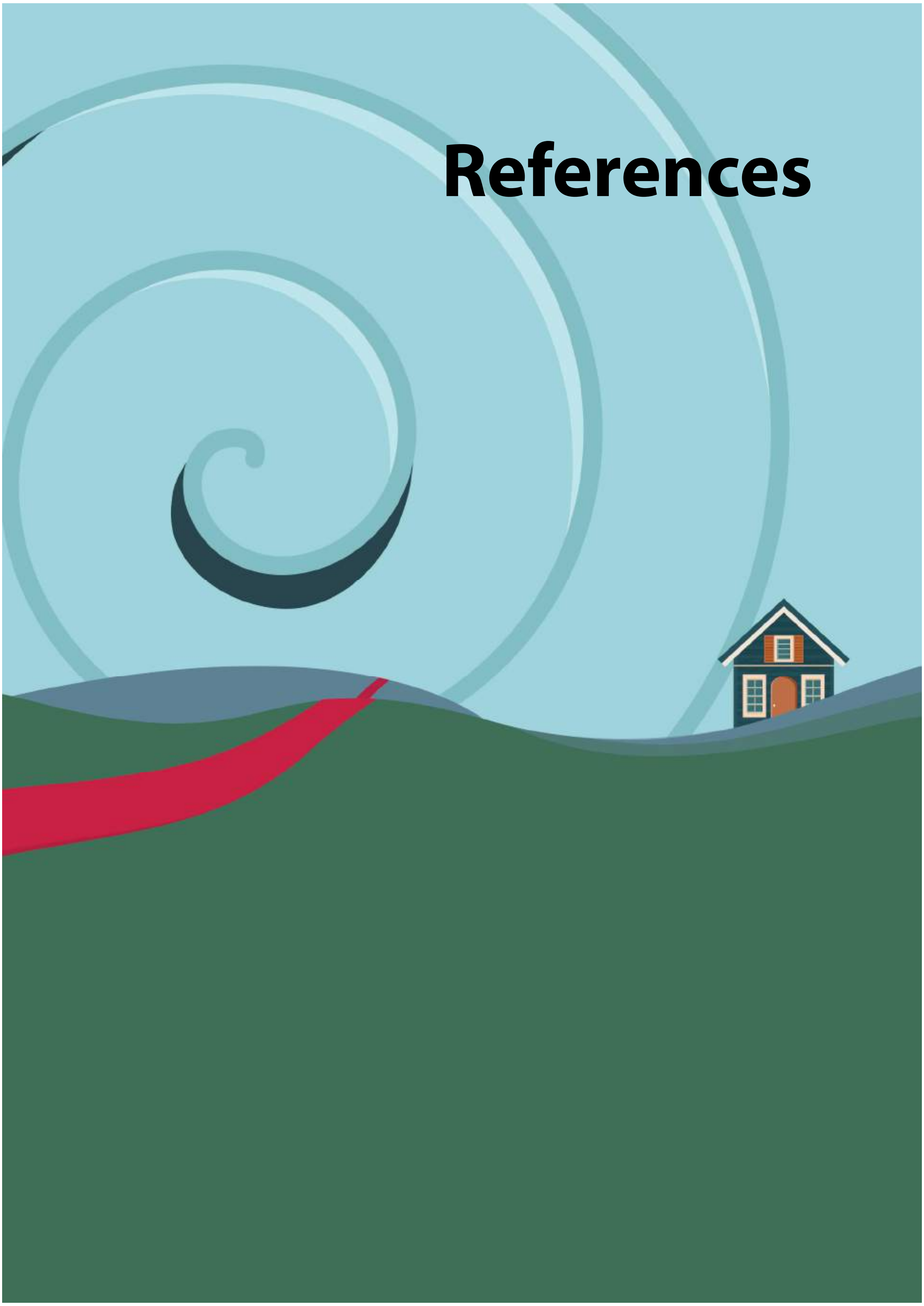

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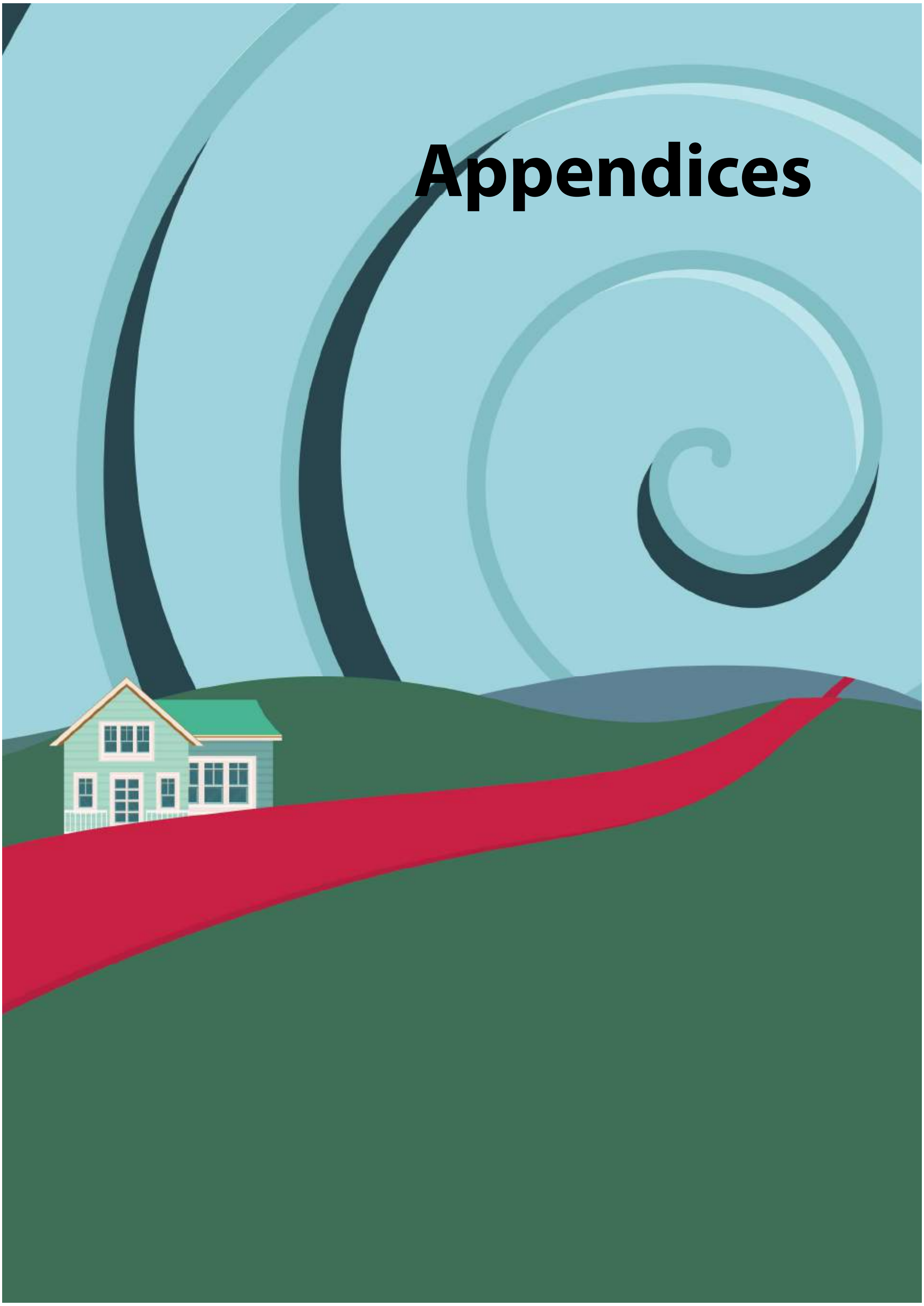
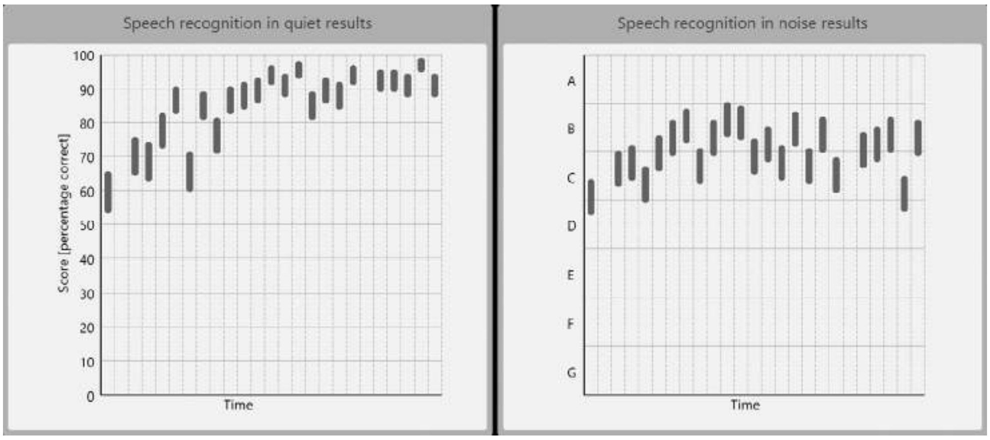
Appendices



Appendi⍰ A

Example bar graph showing the speech recognition in quiet (left panel) and speech recognition in noise scores (right panel).



Datails of the speech reception threshold (SRT) categories in dB SNR that were used to present the results of the digits-in-noise test to the partici-pants⍰

<u>**Cate⍰ory A**</u>
< -7.5 dB SNR
(reference category for NH adults; Smits et al., 2013)

<u>**Cate⍰ory ⍰**</u>
-7.5 to -4.5 dB SNR

<u>**Cate⍰ory C**</u>
-4.5 to -1.5 dB SNR

<u>**Cate⍰ory D**</u>
-1.5 to 1.5 dB SNR

<u>**Cate⍰ory ⍰**</u>
1.5 to 4.5 dB SNR

<u>**Cate⍰ory ⍰**</u>
4.5 to 7.5 dB SNR

<u>**Cate⍰ory ⍰**</u>
> 7.5 dB SNR

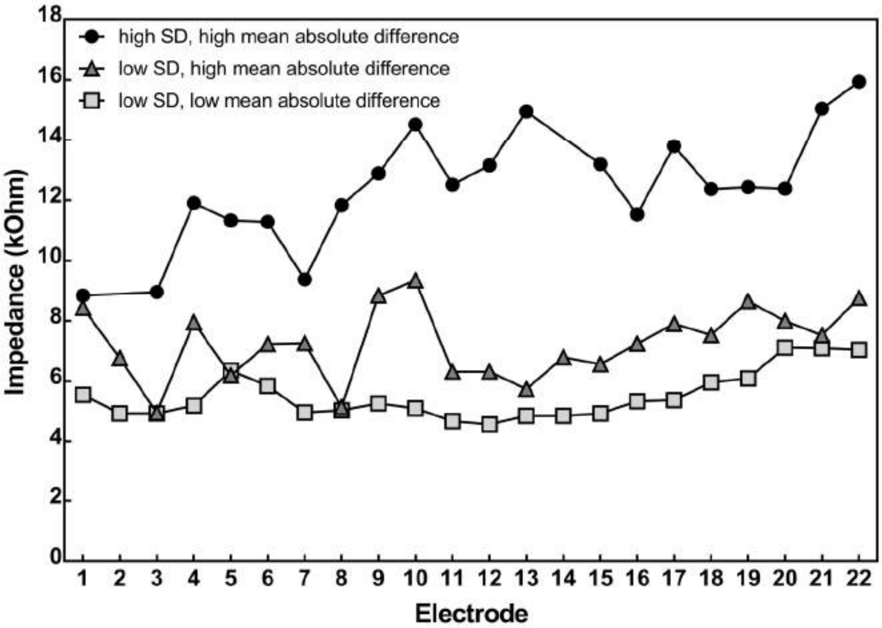
Appendix A

Search strategy

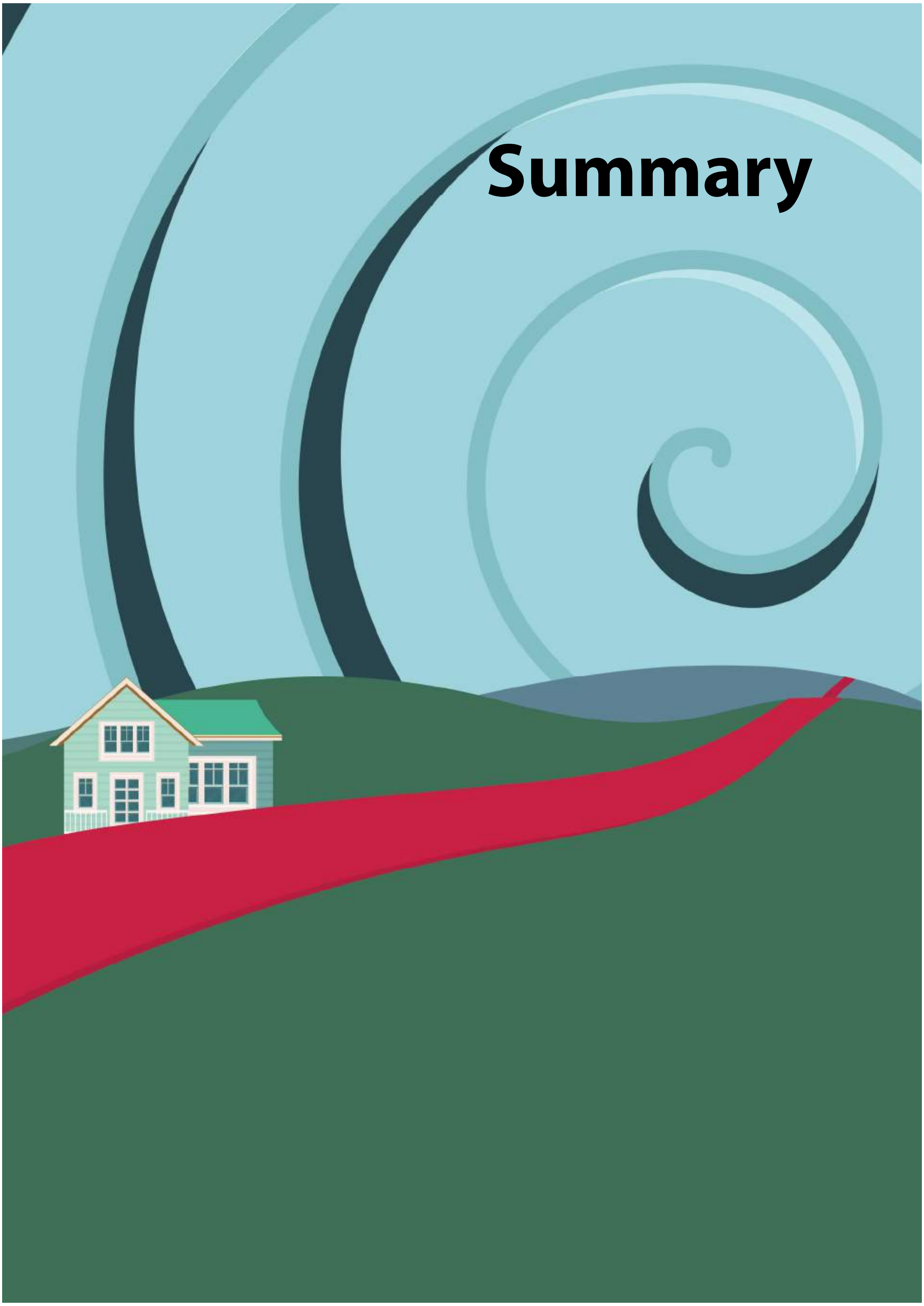
Database	Syntax	Results
PubMed	#1 Search ("Hearing Aids"[Mesh:NoExp] OR "Cochlear Implants"[Mesh] OR hearing aid*[tiab] OR Cochlear Implant*[tiab] OR Cochlear Prosthes*[tiab] OR Auditory Prosthes*[tiab] OR hearing aid*[ot] OR Cochlear Implant*[ot] OR Cochlear Prosthes*[ot] OR Auditory Prosthes*[ot]) #2 Search ((("Software"[Majr] OR (multipl*[ti] AND (memor*[ti] OR program*[ti])) OR programmable*[ti] OR map[ti] OR maps[ti] OR microphone*[ti] OR multi memor*[ti] OR setting*[ti] OR auto select*[tiab] OR autoselect*[tiab] OR microphone*[ti] OR directional*[tiab] OR omnidirectional*[tiab] OR unidirectional*[tiab] OR logging*[ti] OR classificat*[ti] OR setup*[ti] OR set up*[ti] OR log[ti] OR logs[ti]) OR (multipl*[ot] AND (memor*[ot] OR program*[ot])) OR programmable*[ot] OR map[ot] OR maps[ot] OR microphone*[ot] OR multi memor*[ot] OR setting*[ot] OR auto select*[ot] OR microphone*[ot] OR directional*[ot] OR omnidirectional*[ot] OR unidirectional*[ot] OR autoselect*[ot] OR logging*[ot] OR classificat*[ot] OR setup*[ot] OR set up*[ot] OR log[ot] OR logs[ot]) OR scene classification*[tiab] OR scene classification*[ot] OR environmental classification*[tiab] OR environmental classification*[ot])) #1 AND #2	675
EMBASE	Modified search strategy from PubMed	793
Web of science	Modified search strategy from PubMed	588

Appendix C

Examples of impedance profiles



Summary



Summary

A cochlear implant (CI) is a medical device that improves the hearing of patients with severe-to-profound hearing loss. The standard clinical care pathway for CI users in the Netherlands is intense with numerous visits to the clinic in the first year after implantation and regular follow up visits thereafter. During these visits, speech recognition is assessed and the sound processor is fitted or fine-tuned. This thesis describes studies which aim to improve different aspects of the clinical care pathway of new and experienced CI users.

Chapter 1 provides a brief overview of the history of cochlear implantation, changes in candidacy criteria over time, the clinical care pathway of new and experienced CI users, and the use of telehealth in the clinical care pathway. Finally, the outline of this thesis is presented.

Chapter 2 addresses the technical challenges that were encountered in the development of self-administered speech recognition tests for experienced adult CI users at home. It was demonstrated that speech recognition in noise is not influenced by the use of continuous steady-state masking noise (i.e., noise that is presented continuously throughout the test) instead of the standard discontinuous noise (i.e., noise that starts and stops after each stimulus) for both normal-hearing individuals and CI users. The study also showed that the direct coupling between the sound processor and tablet computer by means of an audio cable can be used as an alternative to a loudspeaker and sound booth for speech recognition testing, and that calibrated stimuli can be presented at predefined levels.

In **Chapter 3**, home self-administered speech recognition in quiet and in noise tests were compared to the standard tests in the clinic for a group of experienced CI users. Potential effects of stimuli presentation mode (loudspeaker or audio cable) and assessment (clinician in the clinic or self-assessment at home) on speech recognition scores were investigated. For the recognition of speech in quiet, no significant differences were observed between any of the conditions. In noise, speech recognition scores were significantly better with the audio cable than with the loudspeaker. Home self-assessment of speech recognition had no effect on speech recognition scores. The results demonstrated that it is feasible for experienced CI users to perform speech recognition tests in the home environment.

Chapter 4 presents a study that evaluated the use and feasibility of the self-administered test functionality as part of a telehealth application, the MyHearingApp, with newly-implanted CI users during the first three months of rehabilitation. The frequent assessment of speech recognition provided fine-grained progress details and revealed that speech recognition in quiet and in noise improved steadily during the first few weeks of rehabilitation, after which it stabilized. The fine-grained information enables clinicians to monitor their CI user's speech recognition ability over time without the need for the CI user to visit the clinic.

The home tests provide a good alternative for tests in the clinic for newly-implanted CI users who are able to use the technology required.

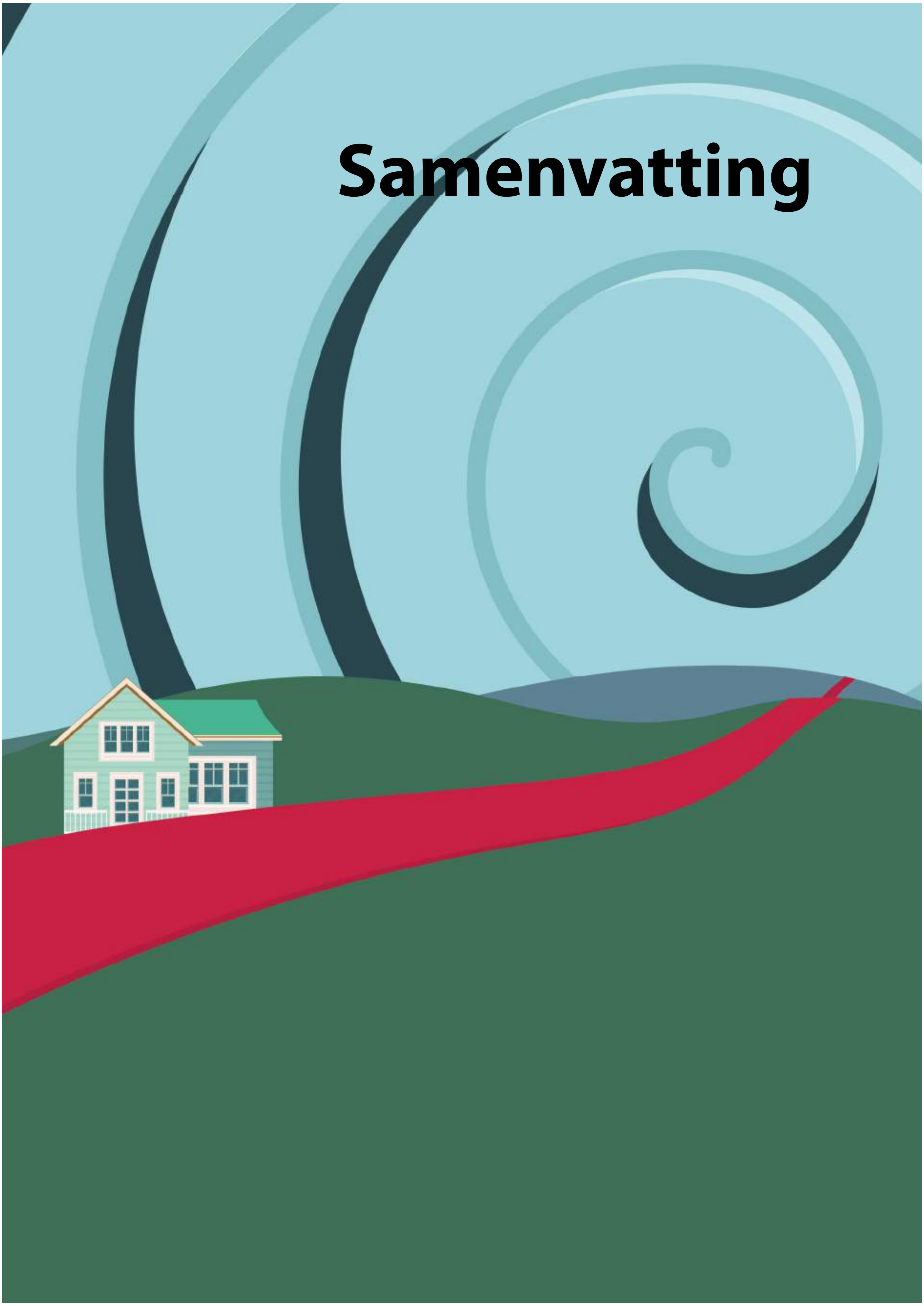
In **Chapter 5**, the self-administered speech recognition test setup was used to assess speech recognition in noise with the Australian English digits-in-noise test in bimodal and bilateral CI users. Monaural and binaural speech recognition in noise was assessed via direct audio input in different conditions to determine the binaural benefit, to investigate the presence of binaural unmasking, and to assess the fluctuating masker benefit. For both bilateral and bimodal CI users, no binaural benefit was demonstrated when comparing monaural to binaural speech recognition in noise. There was no binaural unmasking present in both bilateral and bimodal CI users when speech recognition in noise was compared for diotic (i.e., identical signals in the left and right ear) and dichotic (i.e., with an inter-aural phase difference in the speech signal) listening conditions. Both bilateral and bimodal CI users benefit from interruptions in the masking noise, which yielded a large fluctuating masker benefit when speech recognition assessed with steady-state masking noise was compared to interrupted masking noise.

In **Chapter 6**, a study is presented with the objective to predict speech recognition in quiet and in noise from fitting parameters (e.g., T and C levels), electrically evoked compound action potential thresholds, and impedances in a large group of Cochlear™ adult CI users. Important parameters to predict speech recognition in quiet and in noise were identified. The mean aided thresholds, T levels, and the electrical dynamic range were found to be associated with speech recognition performance in CI users with late onset of severe hearing impairment. Elevated aided thresholds result in worse speech recognition in quiet and in noise. CI users with a DR of 40-60 CL were found to have better speech recognition, both in quiet and in noise. For CI users with early onset of severe hearing impairment, worse speech recognition in quiet and in noise was found for CI users with higher T levels. Measures to express the impedance profile across the electrode array were also identified as predictors of speech recognition in quiet and in noise in this group.

Chapter 7 presents a scoping review on the available literature concerning the use of manual and automatically switching devices by hearing aid and CI users. The literature was synthesized to investigate if users of hearing devices appreciate and adequately use the ability to switch between programs in various listening environments. The review showed that, despite the high number of manual and automatically switching devices that are sold each year, there are remarkably few studies about the use of multiple programs or automatic switching modes for various listening environments. No studies were identified that concerned the use of manual and automatically switching devices in CI users, and no studies were found that examined the accuracy of the use of the appropriate selection of programs for specific listening environments. Although the evidence is weak, the review indicated that at least some hearing impaired individuals use the possibilities of multimemory hearing aids, and that an automatic switching device might be a good solution for those who are not able, or willing, to switch between programs in different listening environments.

The final chapter, **Chapter 8**, discusses the main findings and clinical implications. In addition, suggestions for further research are presented. The findings of the studies in this thesis indicate that self-administered speech recognition tests can be used as an alternative to tests in the clinic. The outcomes of the self-administered tests can be used by CI centres to identify those CI users who require intervention in the clinic. Then, resources can be spent on these C users and the number of visits can be reduced for C users who do not have a clinical need. Furthermore, important predictors of speech recognition have been identified which can be used by clinicians and CI centres to improve their fitting practices and subsequently improve the performance of CI users. Additionally, the fitting of automatic and manual programs for various listening environments can be improved by identifying CI users who would benefit from and prefer to have manual programs.

Samenvatting



Een cochleair implantaat (CI) is een medisch hulpmiddel dat het gehoor van mensen met een ernstig tot zeer ernstig gehoorverlies verbetert. Een CI bestaat uit een inwendig en een uitwendig deel. Het inwendige deel, het implantaat, bestaat uit een ontvanger en een elektrodenbundel die tijdens een operatie in het slakkenhuis (de cochlea) wordt geplaatst. Het uitwendige deel, dat als een hoortoestel achter het oor wordt gedragen, vangt het geluid op uit de omgeving door middel van een microfoon in de geluidsprocessor en bestaat verder uit een spoel en een batterij. Het geluid dat wordt opgevangen wordt geanalyseerd, bewerkt en via de spoel doorgegeven aan het implantaat. De elektroden in het slakkenhuis geven de elektrische signalen door aan de gehoorzenuw.

De standaard zorg voor CI gebruikers in Nederland is zeer intensief met vele bezoeken in het eerste jaar na implantatie, gevolgd door controles elke paar jaar. Tijdens deze bezoeken wordt het spraakverstaan (het verstaan van losse woorden in stilte en het verstaan van cijfers in achtergrondlawaai) gemeten en worden de instellingen van de geluidsprocessor gecontroleerd en indien nodig bijgesteld. Dit proefschrift beschrijft studies die gericht zijn op het verbeteren van verschillende aspecten van de zorg voor nieuwe en ervaren CI gebruikers.

Hoofdstuk 1 omvat een kort overzicht van de geschiedenis van cochleaire implantatie en de veranderingen in de criteria voor implantatie in de loop van de tijd. Tevens wordt er ingegaan op de inrichting van de huidige zorg voor nieuwe en ervaren CI gebruikers en het gebruik van eHealth. Tot slot worden de hoofdlijnen van dit proefschrift gepresenteerd.

We ontwierpen een test voor het thuis zelfstandig meten van het spraakverstaan, via een tablet computer, als alternatief voor de standaard spraakverstaantesten in de kliniek. Daarbij werd gebruik gemaakt van een directe koppeling tussen de geluidsprocessor en een tablet computer. In **Hoofdstuk 2** worden de ontwikkeling van deze thuistest en de technische uitdagingen die zich voordeden bij de ontwikkeling beschreven. We toonden aan dat het spraakverstaan in ruis van zowel normaalhorende proefpersonen als ervaren CI gebruikers niet werd beïnvloed door het gebruik van continue ruis (ruis die continu wordt gepresenteerd gedurende de test) in plaats van de discontinue ruis (ruis die na elke stimulus start en stopt) zoals die normaal in de kliniek wordt gebruikt. Ook lieten we zien dat een directe koppeling tussen de geluidsprocessor en tablet computer middels een audiokabel kan worden gebruikt, als goed alternatief voor het meten van het spraakverstaan in een geluidsdichte cabine met een luidspreker. Tevens werd aangetoond dat stimuli kunnen worden aangeboden op vooraf gedefinieerde niveaus zoals dat ook met een gekalibreerde opstelling in de kliniek gebeurt.

Na de ontwikkeling van de thuistest werd een studie uitgevoerd bij ervaren CI gebruikers. Deze studie is beschreven in **Hoofdstuk 3**. De resultaten van de thuistesten voor het meten van spraakverstaan in stilte en in ruis werden vergeleken met de standaard spraakverstaantesten in de kliniek. Allereerst werd onderzocht of de manier van het geluid aanbieden (luidspreker of audiokabel) effect had op het spraakverstaan. Voor het spraakverstaan in stilte werden geen verschillen gevonden, terwijl het spraakverstaan in ruis gemeten met een audiokabel significant beter bleek te zijn dan het spraakverstaan in ruis gemeten met een luidspreker. Ook werd onderzocht of de plaats en manier van testafname (professional in de kliniek of zelftest in de thuissituatie) effect hadden op het spraakverstaan. Voor zowel spraakverstaan in stilte als het spraakverstaan in ruis werden geen significante verschillen gevonden.

De resultaten van de studie met ervaren CI gebruikers toonde aan dat het haalbaar is voor CI gebruikers om zelf het spraakverstaan te meten in de thuissituatie. De thuistesten werden vervolgens opgenomen in een applicatie, de zogenaamde MyHearingApp, voor gebruik op een tablet computer. Het gebruik en de mogelijkheid van de thuistesten werd vervolgens geëvalueerd in nieuwe CI gebruikers. Zij voerden twee keer per week spraakverstaantesten uit gedurende de eerste drie maanden van de intensieve revalidatieperiode. De resultaten van deze studie worden gepresenteerd in **Hoofdstuk 4**. Het twee keer per week meten van het spraakverstaan leverde gedetailleerde informatie op en toonde een duidelijke verbetering in spraakverstaan gedurende de eerste weken van de revalidatie, waarna het spraakverstaan hetzelfde bleef. De gedetailleerde informatie die beschikbaar komt geeft professionals de mogelijkheid om het spraakverstaan van een CI gebruiker wekelijks te volgen, zonder dat de CI gebruiker de kliniek hoeft te bezoeken. Dit bespaart kosten en tijd voor zowel CI gebruikers als de betrokken professionals. De resultaten tonen aan dat thuistesten een goed alternatief kunnen vormen voor testen in de kliniek voor nieuwe CI gebruikers die in staat zijn de vereiste technologie te gebruiken.

In de studie in **Hoofdstuk 5** werden de zelftesten voor het meten van spraakverstaan gecombineerd met de recent ontwikkelde Australisch Engelse cijfers-in-ruis test. Hiermee werd het spraakverstaan in ruis gemeten bij bimodale (CI in het ene oor en een hoortoestel aan het andere oor) en bilaterale (een CI in beide oren) CI gebruikers. Het spraakverstaan in ruis werd gemeten met een audiokabel in stationaire ruis (ruis die doorlopend even sterk aanwezig is) en fluctuerende ruis (afwisselend korte stukjes ruis en stilte). Zowel het spraakverstaan met één oor (CI of hoortoestel), als het spraakverstaan met beide oren (twee CIs of CI en hoortoestel) werd gemeten. Hierbij werd hetzelfde signaal aan beide oren aangeboden of werd er een verschil in beide signalen aangebracht om de samenwerking tussen beide oren te kunnen meten. Voor zowel bilaterale als bimodale CI gebruikers werd er geen verbetering gezien bij het spraakverstaan in ruis met één CI vergeleken met twee CIs of een CI in combinatie met een hoortoestel. Tevens bleken CI gebruikers geen voordeel te hebben van het verschil dat werd geïntroduceerd tussen het linker- en rechteroor. Dit is in tegenstelling tot mensen met een normaal gehoor die door het introduceren van een verschil in spraaksignaal tussen het linker- en rechteroor juist veel betere spraakverstaanscores behalen. Zowel bilaterale als bimodale CI gebruikers bleken wel veel betere spraakverstaanscores te behalen bij het gebruik van fluctuerende ruis ten opzichte van stationaire ruis.

In **Hoofdstuk 6** wordt een studie gepresenteerd die als doel had om spraakverstaan in stilte en in ruis te voorspellen op basis van metingen en instellingen van de geluidsprocessor. Voorbeelden daarvan zijn: instelling per elektrode waarbij de drempel bereikt wordt (T niveau), instelling per elektrode die tot een duidelijk waarneembaar geluid leidt, maar wat niet te hard klinkt (C niveau), het verschil tussen het T en C niveau (dynamisch bereik), elektrisch opgewekte samengestelde actiepotentialen (ECAP) en impedanties van de elektrodes (elektrische weerstand). Gegevens van een grote groep CI gebruikers van het merk Cochlear™ werden hiervoor geanalyseerd, waarbij onderscheid werd gemaakt tussen CI gebruikers bij wie het ernstige gehoorverlies op een vroege of latere leeftijd is ontstaan. Met behulp van statistische analyses werden variabelen gevonden die het spraakverstaan in stilte en in ruis deels kunnen voorspellen. De statistische analyse toonde aan dat gehoordrempels en het dynamisch bereik invloed hebben op het spraakverstaan van CI gebruikers bij wie het ernstige gehoorverlies op latere leeftijd ontstond. Ook bleken CI gebruikers met bepaalde impedanties slechter spraak te kunnen verstaan. CI gebruikers bij wie het ernstige gehoorverlies op vroege leeftijd ontstond hadden slechter spraakverstaan met hogere T niveaus.

Hoofdstuk 7 presenteert een overzicht van de beschikbare wetenschappelijke artikelen over het gebruik van handmatig en/of automatisch schakelende toestellen door hoortoestel en CI gebruikers. De literatuur werd samengevoegd om te onderzoeken of gebruikers van hoortoestellen de mogelijkheid waarderen om te kunnen schakelen tussen programma's in verschillende luisteromgevingen en of ze deze mogelijkheid juist gebruiken. Uit het overzicht bleek dat, ondanks het grote aantal handmatige en automatisch schakelende toestellen dat elk jaar wordt verkocht, er opvallend weinig studies zijn over het gebruik hiervan in verschillende luisteromgevingen. We vonden geen enkele studie die betrekking had op het gebruik van handmatig en/of automatisch schakelende toestellen door CI gebruikers. Evenmin werden er studies gevonden waarin de nauwkeurigheid van de programmaselectie door gebruikers is onderzocht. Hoewel het bewijs zwak is, gaf het literatuuroverzicht aan dat slechts een klein deel van de slechthorenden de mogelijkheden van hoortoestellen met meerdere programma's gebruikt. Ook gaf het literatuuroverzicht aan dat een toestel met een automatisch schakelprogramma een goede oplossing kan zijn voor hen die niet kunnen of willen schakelen tussen programma's in verschillende luisteromgevingen.

Het laatste hoofdstuk, **Hoofdstuk 8**, omvat een overzicht van de belangrijkste bevindingen en klinische gevolgen. Daarnaast worden suggesties voor verder onderzoek gepresenteerd. De bevindingen van de studies in dit proefschrift geven aan dat een thuistest voor het meten van spraakverstaan een goed alternatief kan zijn voor de testen in de kliniek. De uitkomsten van de thuistest kunnen door CI centra worden gebruikt om CI gebruikers te identificeren voor wie het nodig is om een bezoek te brengen aan de kliniek. Op deze manier kunnen de middelen worden besteed aan CI gebruikers die mogelijk bijstelling van het CI of extra begeleiding nodig hebben. Ook kan het aantal afspraken worden verminderd voor CI gebruikers die geen afspraak in de kliniek nodig hebben. Verder zijn er belangrijke voorspellers van het spraakverstaan in stilte en in ruis geïdentificeerd die door professionals en CI centra kunnen worden gebruikt om de afstelling van CIs en de prestaties van CI gebruikers te verbeteren. Daarnaast kan het aanmeten van handmatige en automatische schakelprogramma's voor verschillende luisteromgevingen worden verbeterd door CI gebruikers te identificeren die baat hebben bij en voorkeur hebben voor handmatige programma's.