Aus der Hals-, Nasen-, Ohren-Klinik und Poliklinik - Plastische Operationen
der Universitätsmedizin der Johannes Gutenberg-Universität Mainz

Implementation and evaluation of a psychoacoustic application for the adjustment of electrical hearing thresholds in cochlear implant patients

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Implementierung und Evaluation einer psychoakustischen Anwendung zur Anpassung der elektrischen Hörschwelle bei Cochlea Implantat Patienten

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Abstract

Objective

Fitting cochlear implants, especially the precise determination of electrical hearing thresholds, is a time-consuming and complex task for patients as well as audiologists. Aim of the research project was to develop an application that enables cochlear implant (CI) patients to determine their electrical hearing thresholds precisely and independently. Applicability and impact of this method on speech perception in noise at soft speech levels were evaluated.

Method

An adaptive psychoacoustic procedure for precise hearing threshold determination (precT) was implemented in MatLab (MathWorks) and a graphical user interface was created. Sound signals were calibrated with a CIC4-Implant-Decoder. *Study design:* A prospective study including 15 experienced adult cochlear implant users was conducted. Electrical hearing thresholds were determined with the automated precT procedure (auto-precT application). Speech perception in noise at 50 dB SPL presentation level was measured for three conditions: (P1) T-levels kept at the previously established T-levels; (P2) T-levels set to the hearing thresholds determined using the auto-precT application; (P3) T-levels set 10 cu below the values determined with the auto-precT application.

Results

All subjects were able to perform the auto-precT application independently. T-levels were altered on average by an absolute value of 10.5 cu compared to the established T-levels. Median speech reception thresholds were significantly improved from 2.5 dB SNR (P1) to 1.6 dB SNR (P2, $p = 0.02$). Speech perception was lowest using the globally lowered T-levels, median 2.9 dB SNR (P3, not significant compared to P1 and P2).

Conclusion

An application that allows patients to precisely and independently determine their electrical hearing thresholds, without an attending audiologist, was developed. The applicability of the developed application was confirmed in a clinical study. Patients benefited from adjusting the T-levels to the threshold levels determined with the auto-precT application. The integration of the application in the clinical fitting routine as well as a remote fitting software is recommended. Furthermore, future possibilities of auto-precT include the implementation of the application on tablets or smart phones.

Abbreviations

3-AFC	three alternative forced choice procedure
AD-DA	analog/digital and digital/analog conversions
auto-precT	automated procedure for precise hearing threshold determination
BKB	Bench-Kowal-Bamford sentences
CI	cochlear Implant
C-level	‘comfortable level’, maximum stimulation level
CNC	Consonant nucleus (vowel) consonant words
CUNY	City University of New York sentences
cu	clinical unit, unit used for fitting cochlear implants
dB	decibel
DIET	Direct Implant Emulator
ECAP	electrically evoked compound action potential
EDR	electrical dynamic range
FMS	Freiburger Monosyllable Test
IQR	interquartile range
MCL	most comfortable loudness
OLSA	Oldenburger Satztest - german speech perception test
PCA	personal audio cable
SNR	signal to noise ratio
SPL	sound pressure level
SRT	speech reception threshold
T-level	‘electrical threshold’, minimum stimulation level

Tables

Figures

1. A brief Introduction to cochlear implantology

This chapter is intended to provide a basic understanding of the structure and function of cochlear implants. In Germany, the first patients were supplied with cochlear implants in 1984 by Lehnhart and Laszig in Hannover (BVMed, 2015). The cochlear implant is a neuro prosthesis. Considering the term ‘prosthesis’, people usually think of motor prostheses - a lower leg prosthesis, for example, replaces a lost leg and enables a patient to walk. The cochlear implant is a sensory prosthesis. It offers patients with no or very low residual hearing the possibility to regain communicative skills. In order to understand the functionality of cochlear implants, one has to refer to basics in ear anatomy and physiology of hearing.

Anatomy of the Ear and Physiology of hearing

The ear can be divided into three regions – the external, the middle and the inner ear (Figure 1). The external and the middle ear are separated by the tympanic membrane, the middle and the inner ear are separated by the oval window. Sound waves from the environment are collected by the external ear and pass through the outer ear canal to the tympanic membrane allowing mechanical sound conduction by vibrations. Directly connected to the tympanic membrane are the ossicles of the middle ear. These are the hammer, the anvil and the stirrup. The vibrations of the tympanic membrane are reinforced by the leverage effect of the ossicles and transferred to the stirrup (Figure 2).

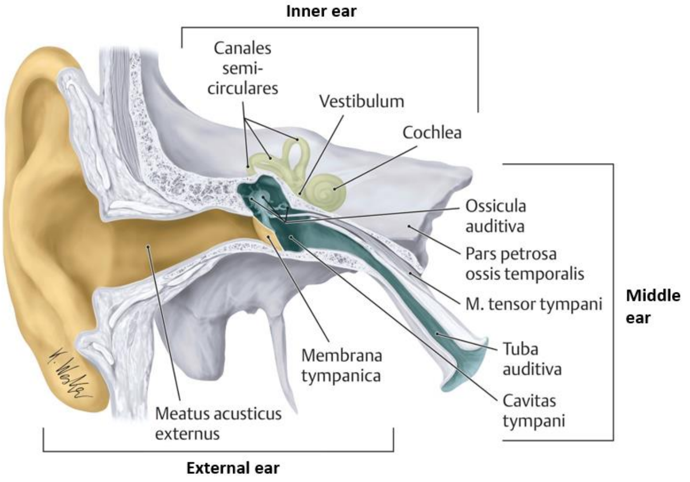


Figure 1. Overview of the anatomy of the ear. Adapted from Schünke et al. (2009)

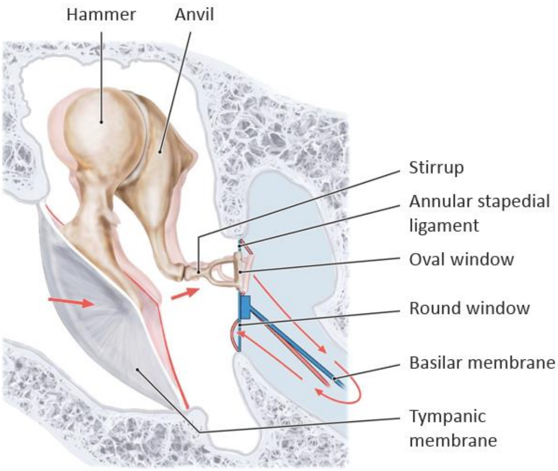


Figure 2. Sound transmission through the middle ear. Adapted from Schünke et al. (2009)

The stirrup is directly connected to the oval window (‘fenestra vestibuli’), which is the junction to the inner ear. The inner ear is a complex channel system consisting of three components (Figure 3): the vestibular duct and tympanic duct, which are filled with perilymph and the cochlear duct, which is filled with endolymph.

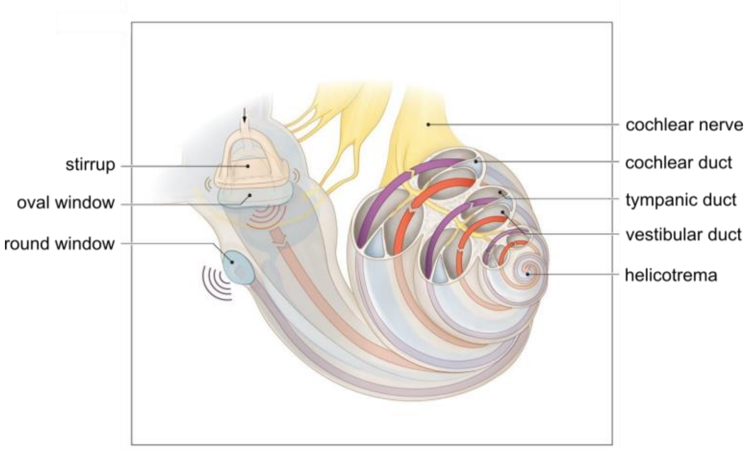


Figure 3. Sound wave transmission in the cochlea. Adapted from © AMBOSS GmbH (2018)

Oscillations of the stirrup are transmitted to the perilymph via the oval window. Thereby a travelling wave is evoked in the cochlear duct. The wave has its maximal amplitude at a specific location, which is dependent on the frequency of the sound that evoked the wave. This effect is attributed to the varying stiffness of the basilar membrane, whereby the membrane is stiffer in the basal region than in the apical region. As a result, a ‘mechanical frequency analysis’ is conducted - higher frequencies evoke waves with a maximum in the basal region and lower frequencies evoke waves with a maximum in the apical region (Figure 4). This mechanical separation of frequencies is the crucial for of the *tonotopy* of the cochlea. In the functioning ear each frequency evokes a nerve impulse at a specific location in the cochlea. This is of great importance for the development and application of cochlear implants.

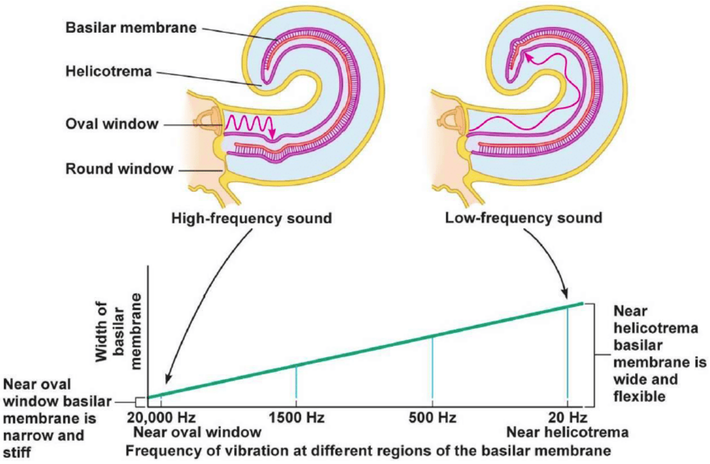


Figure 4. Tonotopy of the cochlea - http://ihearingaids.co/ (2014)

The transformation of the mechanical sound wave to an electrical signal happens in the organ of Corti. It is located at the bottom of the cochlear duct and consists of the tectorial membrane, outer and inner hair cells and the basilar membrane (Figure 5).
The oscillations of the basilar and tectorial membrane lead to a deflection of the hair cells. Thereby, the hair cells are depolarized and the firing of the afferent fibers is increased. This is the mechanical-electrical transformation. The afferent fibers of the Corti organ unite in the auditory nerve and their firing leads to a hearing sensation in the brain.

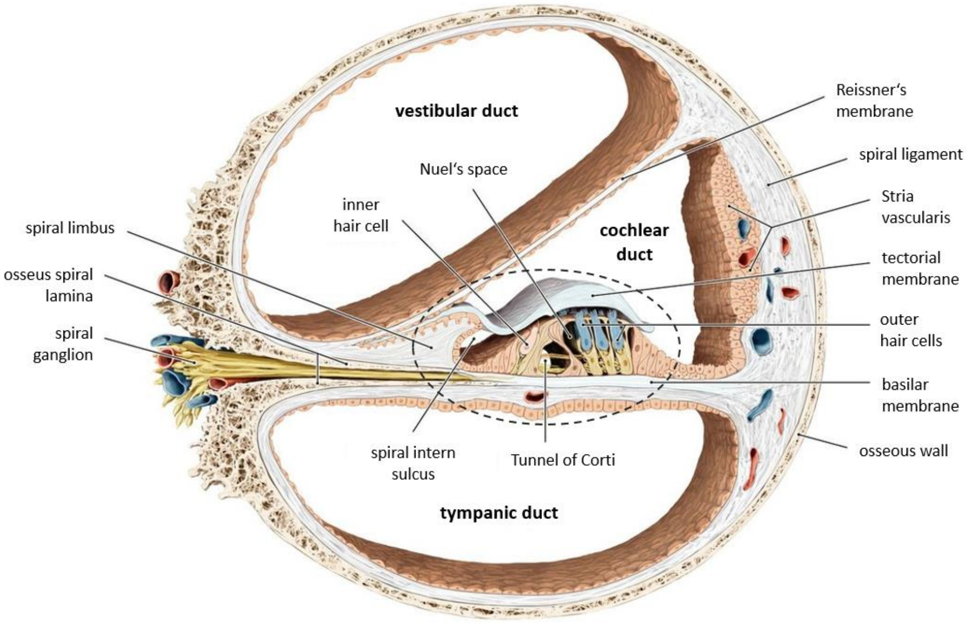


Figure 5. Organ of Corti (circled with dashed lines). Adapted from Schünke et al. (2015)

There are many different factors that lead to a reduced or lost functionality of the hearing system. Several causes are related to a loss or dysfunction of hair cells. In such cases, a rehabilitation with cochlear implants is taken into consideration.

1.1. Indication for cochlear implantation

Hearing loss has several etiologies and affects people at all ages. The prevalence of bilateral hearing loss that requires treatment in newborn is 2.1 of 1000 (Neumann et al., 2006). Etiologies of prelingual hearing loss are mainly genetic, infectious or neuropathic. Postlingual hearing loss is often caused by infections, especially meningitis, trauma and ototoxic medications (Ptok, 2011). Another widespread kind of hearing loss is noise-induced hearing loss. The outer ear cells, which are crucial for the hearing process, are most vulnerable to noise trauma.
The hearing loss of elderly people is called presbyacusis and usually affects the perception of sounds with higher frequencies.

Generally speaking, cochlear implants come into play, if patients with sensorineural hearing loss do not reach sufficient speech perception with hearing aids and an improvement with a CI is expected. With the further development of the cochlear implant technology the indications for CI implantation increased significantly in the last years. While the implantation of a CI used to be only indicated for patients with severe bilateral hearing loss, it is now common to also provide patients with unilateral deafness with a CI (Sampaio et al., 2011). ‘The diagnosis of hearing loss, its classification by grade and type, requires special expertise and the use of appropriate methods. In addition to an ENT-medical or pedaudiological examination, these include subjective and objective audiometric procedures, diagnostic imaging as well as a pedagogical, logopedic and psychological assessment of rehabilitation capacity including the psychosocial situation. Special features of children and adults are considered separately’ (German S2k Guideline, AWMF 2012). The most important subjective tests are the pure tone audiometry as well as speech perception tests. The assessment of speech perception must be performed with and without hearing aid so that the benefit of the hearing aid can be clearly determined. Objective tests include otoacoustic emissions (OAE) and electric response audiometry (ERA), which in contrast to the subjective tests, do not require active participation of the patients. A summary of indications, including audiometric and speech perception test results, for cochlear implantation is shown in Table 1. Guidelines for the indication vary slightly from country to country.

Table 1. Indications for Cochlear Implantation according to Egilmez and Kacioglu (2015)

Indications of the Cochlear Implantation		
Audimetric candidacy		
12-24 months	*2-17 years*	*>18 years*
Profound SNHL - PTA ≥90 dB	Severe to profound SNHL- PTA ≥70 dB	Severe to profound SNHL- PTA ≥70 dB
Speech recognition tests		
12-24 months	*2-17 years*	*>18 years*
Limited benefit from binaural amplification	Limited benefit from binaural amplification with scores ≤30%	Limited benefit from binaural amplification with scores lower than 50% in ear to be implanted and 60% bilaterally
Prelingual SNHL		
Perlingual SNHL		
Postlingual SNHL		
New concepts in indications		
Younger age, Hearing preservation, Unilateral deafness, Tinnitus, Bilateral implantation		

SNHL: sensorineural hearing loss; PTA: pure tone average;

Guidelines vary between countries

Besides the measurable data of hearing loss another very important criterion for cochlear implantation is a high level of motivation for the rehabilitation and learning process after the implantation which varies among patients.

1.2. Architecture and function of cochlear implants

Cochlear implants are basically a functional bypass between the acoustic environment and the auditory nerve fibers. They take over the tasks usually performed by the outer, the middle and especially the inner ear: transforming sound signals from the environment into electrical signals and analyzing them in terms of time and frequency. The implants have internal and external components (Figure 6). Outside, similar to a hearing aid, there is a microphone receiving the sounds of the environment. Usually in the same box as the microphone is the sound processor, which converts the sounds into digital signals. Connected to that is the transmitter that forwards the signals from the processor by induction to a receiver placed under the skin. The transmitter and the receiver are connected through a magnet. (That is how the transmitter stays in place. Magnets are available in different strengths and are chosen depending on the patient's head anatomy and hair.) The receiver translates the signals into current pulses and sends them to the corresponding electrodes that are placed within the cochlea turns. A wire that is leading from the receiver through the mastoid bone, the middle ear and then through a cochleostomy to the tympanic duct is the link from the receiver to the inner ear. The front part of the wire is an electrode array that evokes electrical stimuli in the cochlea and is thereby stimulating the auditory nerve cells enabling a hearing sensation.

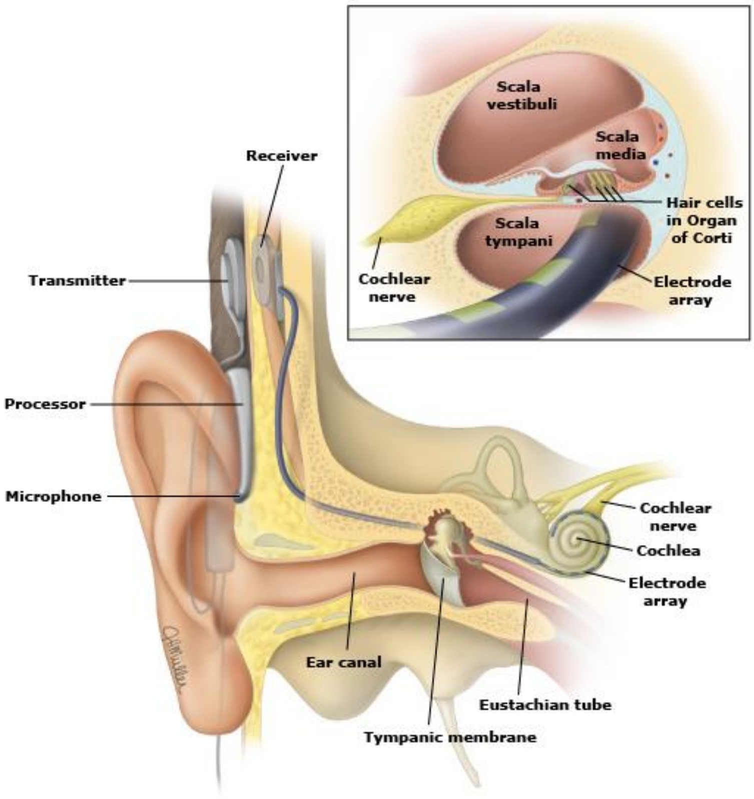


Figure 6. The internal and external components of a cochlear implant

UpToDate Inc. and/or its affiliates, (2018)

A fundamental demand for the function of cochlear implants is the frequency analysis by the processor. With normal hearing the frequency analysis of the sounds happens mechanically in the cochlea. Developers of cochlear implants aim to imitate the tonotopy of the cochlea as good as possible. Therefore, a modern cochlea implant has a number of electrodes with each one corresponding to a specific frequency band, depending on its location within the cochlea (Figure 7).

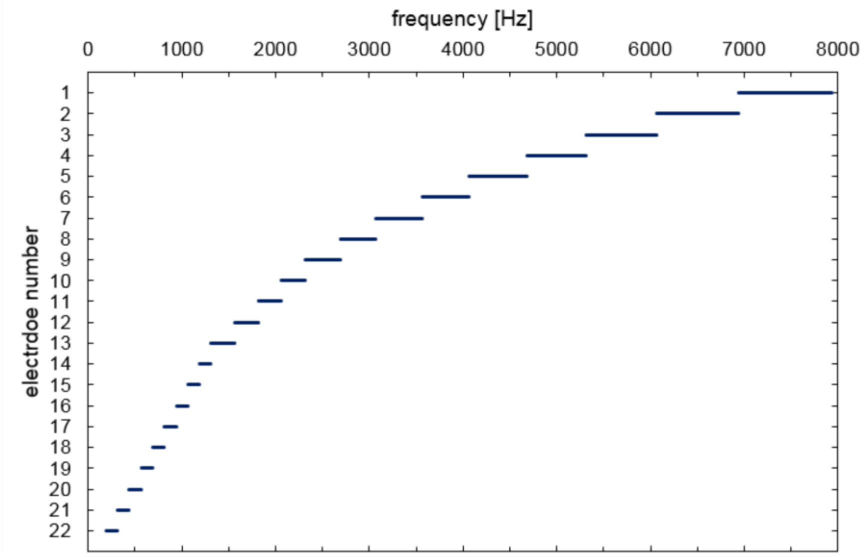


Figure 7. Frequency bands covered by the individual electrodes. Number of electrodes and width of frequency bands vary between manufacturers. The displayed frequency bands are those from the Cochlear CP910 processor that was used in the research project.

One of the technical challenges with cochlear implants is that the concept of digitally analyzing frequencies and evoking stimuli at the corresponding electrode has its limitations, because the number of electrodes that can be used is limited. This is due to superposition effects that occur if electrodes are too close to each other. Figure 8 visualizes the time and frequency analysis of a sound signal performed by the speech processor – here it is the resulting sound when pronouncing 'sa'. First, the processor filters the sound signal into the corresponding frequency bands of the electrodes. For clarity, only 4 electrodes are shown in the figure. Today most commercially available cochlear implants have 12 to 22 electrodes. After the band-pass filtering the processor detects the envelope (the course of the amplitude) for each frequency band. Then the signals are modulated into trains of square pulses. These signals are the result of a signal cascade in a cochlea implant device stimulating the auditory nerve fibers.

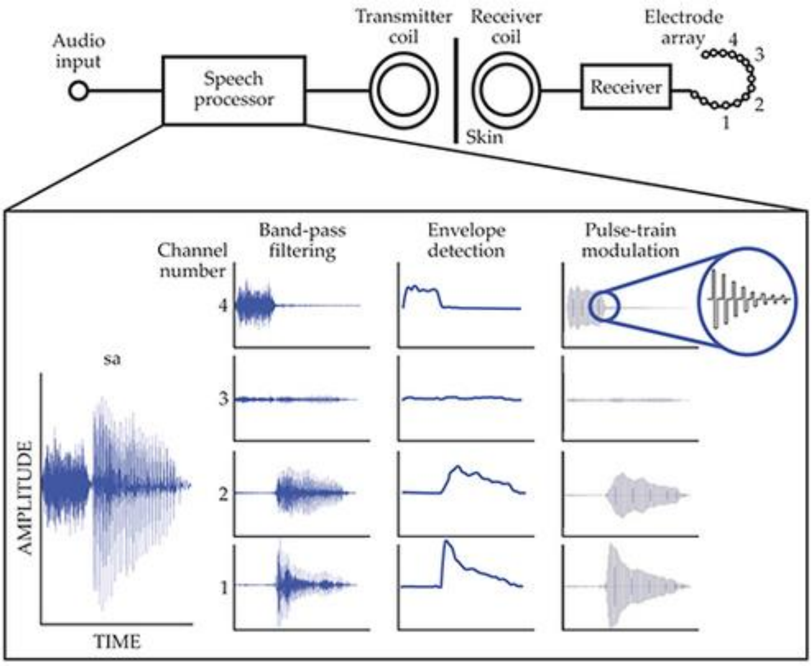


Figure 8. Transformation of a sound signal, in this case the sound ‘sa’, received by the microphone in the speech processor. Time and frequency of the signal are analyzed. The figure shows four electrodes. Currently, most commercially available cochlear implants have 12 to 22 electrodes. Figure from Svirsky (2017).

A pulse train has a small number of parameters that determine how it is stimulating the auditory nerve fibers in the surrounding of the electrode: the strength of the electric current, the width of the pulse and the stimulation rate (Figure 9). In order to prevent the electric unidimensional charge polarization of the surrounding tissue, two sequential pulses have opposite charges and equivalent current levels.

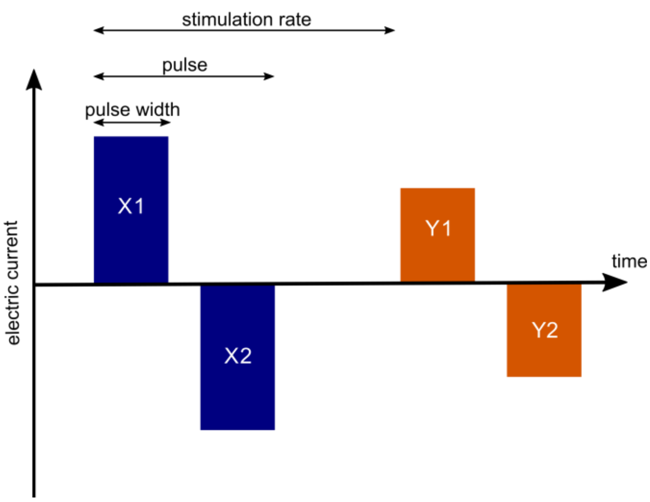


Figure 9. Parameters of the pulses evoked by an electrode.

Different stimulation strategies, so-called speech coding strategies, have been developed, e.g. the continuous interleaved sampling (CIS) strategy Wilson et al., 1993, the spectral peak (SPEAK) strategy (Seligman and McDermott, 1995) and the advanced combination encoders (ACE) strategy (Arndt et al., 1999). In several studies the coding strategies were evaluated (e.g. Kiefer et al., 2001; Psarros et al., 2002; Skinner et al., 2002a; Skinner et al., 2002b). The used speech coding strategy depends on the manufacturer, the patient’s choice and the individual audiological/physiological patient’s needs. A basic and widely applied principle of the speech coding by the processor is described in the following: in order to prevent interactions between electrodes the stimulating of electrodes is temporally interleaved. That means that at each point of time only a single electrode is stimulated while the others rest. The interleaving is realized by high stimulation rates of several thousand pulses per second. Thus, even though the electrodes are not stimulated simultaneously a continuous hearing sensation is possible. This is analogous to the signal processing of vision: a film is a sequence of single pictures, but is perceived as a continuous movement, because the cognitive organ – the eye in this case - is only capable of distinguishing a limited number of frames per second.

1.3. Rehabilitation and fitting process

An emotional moment for every patient who received a cochlear implant is, when the implant is switched on for the first time. This happens on average 28 days after implantation (Vaerenberg et al., 2014) and is the start of the rehabilitation process in which many different professions play an important role, especially speech therapists and audiologists. Adjusting the parameters of the cochlear implant is not merely a technical process but goes along with the learning process of the patient. Hearing perception with a cochlea implant is different from normal hearing, because the electric stimulation of the auditory nerve fibers is not comparable. Patients need to get used to the new quality of hearing. This is an active process that may afford time and training.

In the fitting process of cochlear implants two parameters play a major role, the T-level (threshold level) and the C-level (comfortable level). The T-level is the minimum current level with which the electrodes are stimulated. It is usually set to the electrical hearing threshold, the lowest current level at which the patient perceives a hearing sensation. Determining the hearing threshold is especially challenging for newly implanted patients, as they are not used to hearing with the implant. The C-level corresponds to the current level that evokes a hearing sensation that is comfortably loud. The range in between the T- and the C-Level is called the electrical dynamic range (EDR). Each patient has a personally adjusted program, a ‘MAP’, with individual T- and C-Levels and other settings, as the speech coding strategy for example. Figure 10 shows a screenshot of an exemplary MAP in the Cochlear Fitting Software Custom Sound (COCHLEAR, Macquarie, Australia). Usually the MAP is adjusted by an audiologist after the implantation in three monthly, three quarterly and then annual sessions (Vaerenberg et al., 2014).

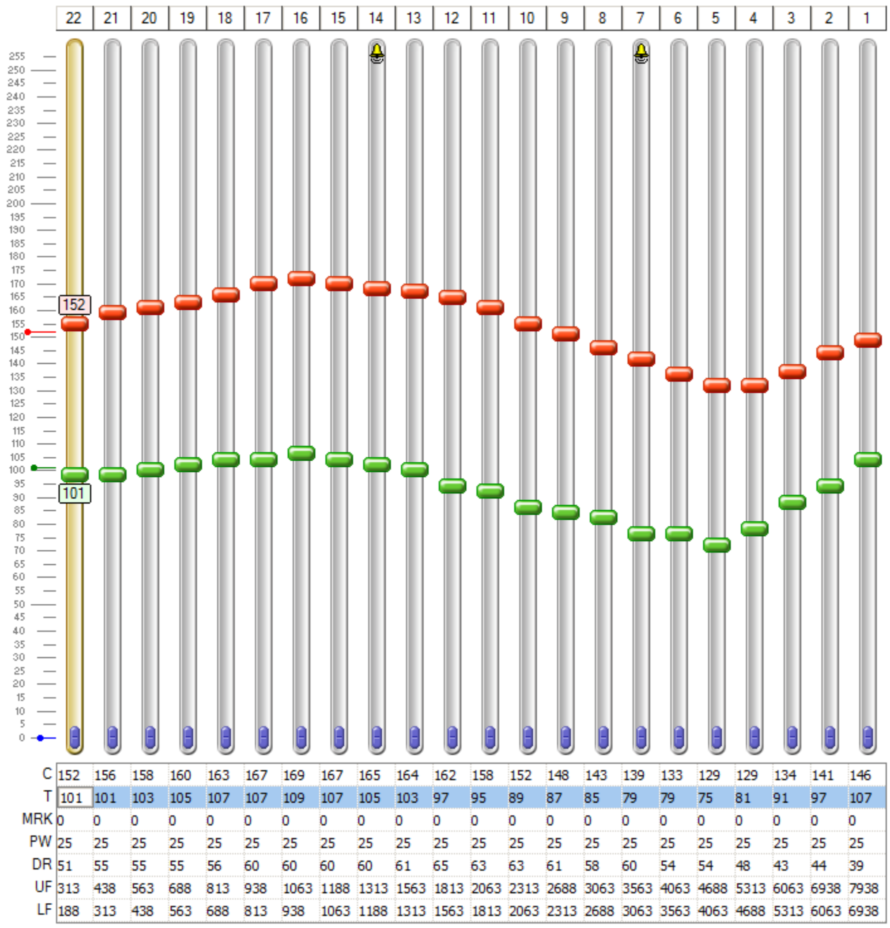


Figure 10. Screenshot of a MAP in the Cochlear Fitting Software Custom Sound. T-levels can be set with green bars and C-levels with the red bars. The numbers on top indicate the electrodes, starting with 22 as the most apical electrode. Abbreviations: PW = pulse width; DR = dynamic range; UF = upper frequency; LF = lower frequency

2. Objective

2.1. Background

A precise fitting of the cochlear implant is essential for good speech perception. Nevertheless, Vaerenberg (Vaerenberg et al., 2014) found in a global survey that it is common practice to measure T- and C-levels only for a few electrodes and then interpolate the values for the others. This reflects well-known obstacles in everyday clinical routine. C-Levels need to be adjusted well, so that all acoustic information of speech is received, all frequencies are perceived equally loud without evoking excessive loud hearing sensations. Accurate T-Levels are required, in order to perceive ‘soft’ sounds. The importance of the ability to perceive sounds for the comprehension of ‘soft’ speech has been shown in numerous studies (Skinner et al., 1999; Firszt et al., 2004; Holden et al., 2011). It was also stated by several authors that the adjustment of T-levels for individual electrodes is beneficial (Willeboer and Smoorenburg, 2006; Botros et al., 2013; Mewes and Hey, 2017).

The adjustment of precise C- and T-levels still is a problem in cochlear implant fitting. For the initial C-levels setting, it is common clinical practice that the audiologist increases the stimulation intensity, starting at zero, until a comfortable loudness is achieved. Subsequently, the loudness is compared with the adjacent electrodes and adjusted as needed. The accurate determination of the electrical hearing threshold and the precise setting of T-levels is a demanding task for the audiologists as well as patients for several reasons. During the first months after implant activation, patients need to get used to the new quality of hearing with the CI. This makes the challenging task of determining the hearing threshold even more difficult. When the audiologists present stimuli close to the hearing threshold, it is often difficult for patients to judge whether they actually heard a tone or whether it was just a ‘phantom sound’ – a hearing sensation generated by the brain without an external correlate. Furthermore, CI patients frequently suffer from tinnitus obscuring endogenous or exogenous sounds.

Many different concepts of determining the electrical hearing threshold have been discussed and there has not been an agreement on a gold standard so far. One method called ‘count the pulses’ was proposed by Skinner et al. (1995). Pulse trains were presented to the subjects at different levels and they had to count the number of pulses they heard. The hearing threshold estimate was the lowest level at which the subjects correctly counted the pulses.

This method was advanced to an adaptive procedure by van Wieringen and Wouters (2001). Common to most adaptive procedures is that the sound level is decreased until no hearing sensation is perceived anymore, the task is performed wrong, respectively. Following, the sound level is increased again until the task is performed correctly again. Subsequently, the sound level is decreased once more. The switch from decreasing to increasing the sound level and vice versa is referred to as a ‘reversal’. While Skinner et al. (1995) estimated thresholds only after one reversal or sometimes even without a reversal, van Wieringen and Wouters estimated thresholds after eight reversals. Another adaptive procedure was presented by them in the same study, the ‘choose the interval with the pulse’ task. Four intervals represented by four buttons were shown on a computer screen and highlighted one after another. The subjects had to choose during which interval they heard the pulse train. The level of the pulses was altered adaptively. Furthermore, an adjustment procedure was tested, where the subjects set the level of the pulse train to the lowest level at which they perceived the pulses.

Mewes and Hey (2017) mentioned the widely used clinical practice of behaviorally measuring hearing thresholds. Initially the stimulus level is lowered from a clearly detectable level below the hearing threshold by a set step size and then increased until a sound is perceived again. To confirm the determined hearing threshold, the stimulus level is lowered once more by a smaller step size until no hearing sensation is perceived and then increased again until the sound is detected.

With the intention to free audiologists from the time-demanding task of behaviorally measuring hearing thresholds for all electrodes (implants by COCHLEAR have 22 electrodes), so-called streamlined fitting procedures have been developed (Plant et al., 2005; Botros et al., 2013). In order to optimize the CI-programming, some audiologists use the electrically evoked whole nerve action potentials (ECAP) as a parameter. Alternatively, hearing thresholds are behaviorally measured only for some electrodes and interpolated for the rest.

Recently Rader et al. (2018) presented an innovative adaptive method for determining precise electrical hearing thresholds (precT) and evaluated the impact of the precise fitting on speech perception at soft levels. The electrical hearing thresholds were determined by applying an alternative forced choice (afc) method using the established fitting software Custom Sound (COCHLEAR, Macquarie, Australia). The results of this approach were very promising, as the concept led to a significant improvement in the perception of soft speech.

The objectives of this research project were derived from the benefit of precisely determining hearing thresholds reported by Rader et al. (2018) and the demand for improvements in clinical workflow routines.

2.2. Objectives of the research project

The goal of this research project was to advance the precT procedure, so that patients can determine their hearing thresholds precisely and more importantly *by themselves*. For this purpose, the following objectives were set:

1) The precT procedure should be automated and implemented into a MatLab (MathWorks) program. Thus, the ‘*auto-precT*’ procedure can be performed independently of the clinical fitting software and without an audiologist, just by patients themselves.
2) The new application should be evaluated in a clinical study regarding applicability and speech recognition outcome.

In chapter 3 the realization and implementation of the auto-precT application is described. Following in chapter 4, the study for the evaluation of the new method is proposed. The results of the study are shown in chapter 5 and discussed in chapter 6. In chapter 7 conclusions of the research project are presented and an outlook for further investigations and developments is given.

3. Realization of the auto-precT application

In this chapter the auto-precT application and the process of its implementation in MatLab are presented. First, the psychoacoustic procedure itself is introduced. Following, the used hardware and software settings are described, before the calibration of the setup and the software are explained. In the end of the chapter the implementation of the procedure in Matlab is visualized and described.

3.1. A two stimuli approach for precise threshold determination

The auto-precT application determines the threshold levels with an iterative adaptive three alternative choice method. The newly developed software, based on the precT method proposed by Rader et al. (2018), repeatedly presents two stimuli with the same frequency, but different current levels. After the presentation of the two stimuli the patients are asked how many sounds they heard. Given the answer is ‘two’, it can be assumed that both stimuli were above the hearing threshold and the stimuli levels are subsequently decreased by the step size set before. If no stimulus was perceived, the stimuli levels are increased by the chosen step size. Given the answer is ‘one’, the hearing threshold is presumably in between the two stimuli. In this case, the first algorithmic circle is over and a second run starts. For each electrode there are three repetitions with sequentially smaller step sizes in between the current levels of the stimuli. First, the step size is set at 10 cu, then at 6 cu and finally at 3 cu. When the patient perceived one stimulus at a time the run is finished and the step size in between the stimuli is decreased. Subsequently, the next algorithmic run begins. The starting current level is set two step sizes above the recent level. The last stimulus that was heard in the third run is saved as the hearing threshold and is later set as the T-level for that electrode. The patient works through this procedure for every electrode in a pseudo-randomized order. Figure 11 shows an exemplary run for one electrode and Figure 12 shows a process chart for the procedure from Rader et al. (2018).

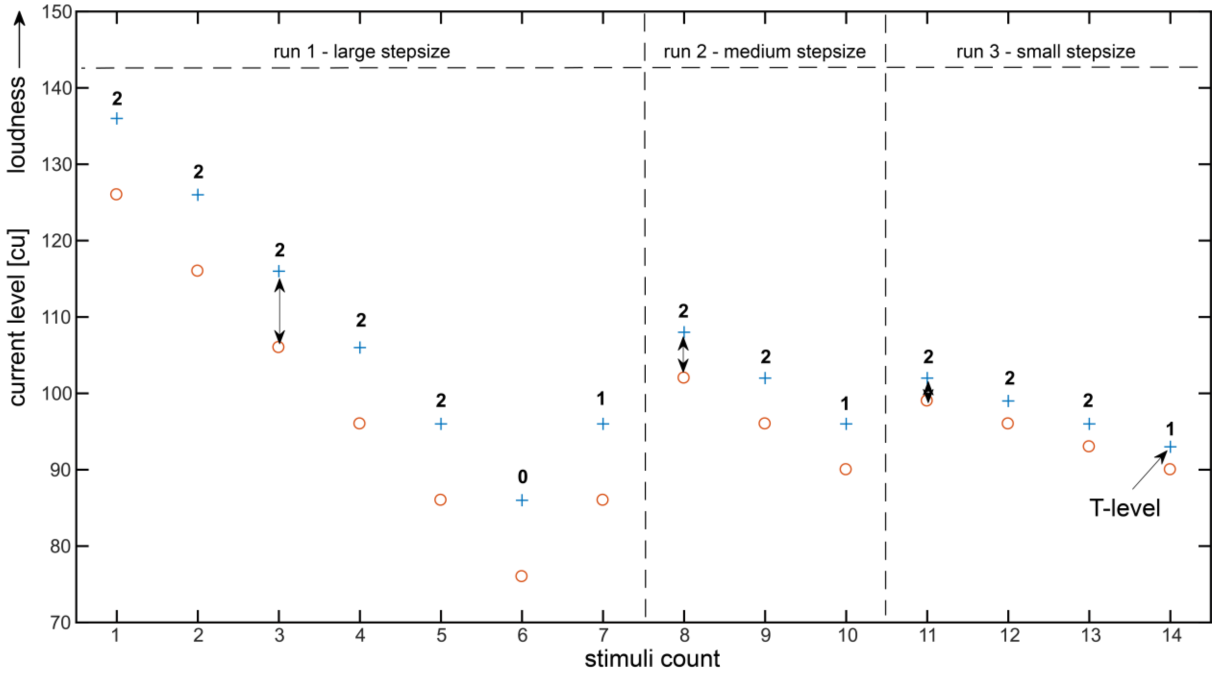


Figure 11. Exemplary iteration of the auto-precT application for one electrode. The bold numbers indicate the count of perceived sounds. The double-headed arrows indicate the applied step size. The precT method was derived from earlier reports (Rader et al., 2018).

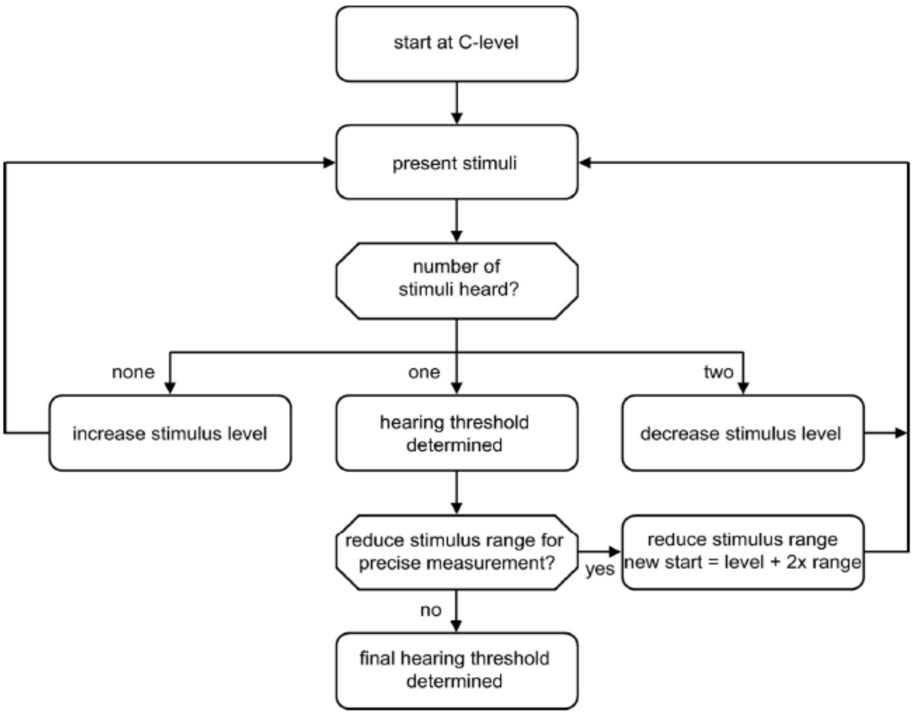


Figure 12. Process chart for electrical threshold estimation with the precT-procedure proposed by Rader et al. (2018).

3.2. Hardware setup and software settings

One aim of the study was to develop a software and a hardware setup that allows running the proposed precT procedure without the clinical fitting software. Thus, it needed to be ensured that the newly developed application can evoke specific current levels at specific electrodes. Therefore, in order to calibrate the setup, the following method was developed: A control computer with MatLab was connected to a sound card in which a so-called personal audio cable, PAC (COCHLEAR) in the following, was plugged in. This connected the sound card with a CP910 audio processor (COCHLEAR). In order to measure the current evoked by an audio signal, generated with MatLab, a Decoder Implant Emulator (DIET, COCHLEAR, Macquarie, Australia) was used. The DIET can be connected to an audio processor and measure the stimulation data. The audio signal was converted with a high-quality 24-bit, 8-channel AD-DA converter (RME Fireface UC, Haimhausen, Germany) and then transmitted to the Cochlear CP910 audio processor via the PAC. The audio processor was connected to the DIET which was linked with a second control computer in order to log the stimulation data for each electrode. Figure 13 shows the hardware setup for the calibration. Using the DIET, it was possible to correlate the generated stimuli with the electric current induced.

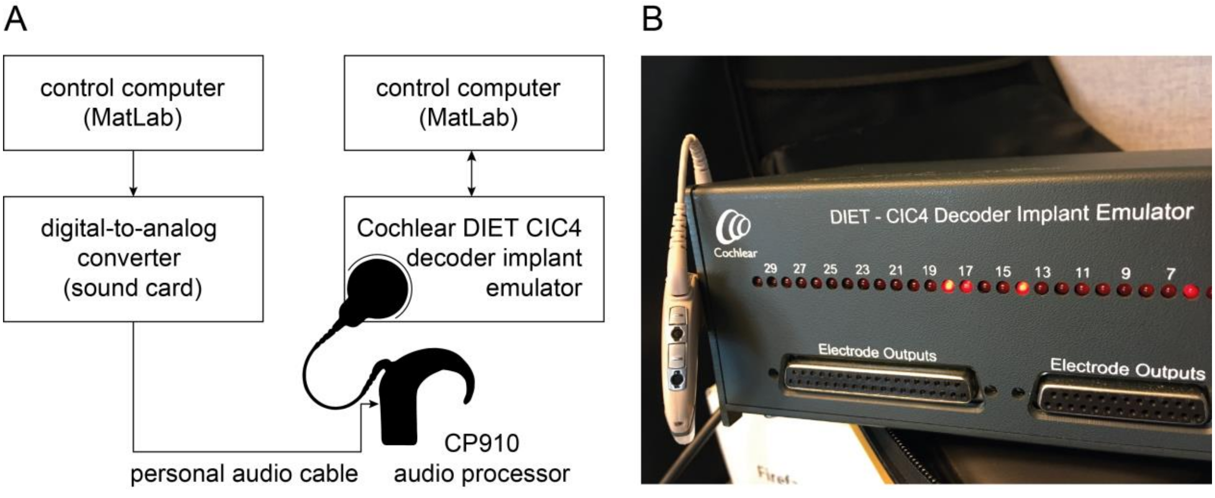


Figure 13. Hardware setup for audio processor and audio signal calibration. (A) Scheme of the setup. (B) Cochlear DIET - CIC4 decoder implant emulator with attached sound processor CP910. The DIET visualizes which electrodes are being stimulated and logs the stimulation data.

For the calibration as well as for the clinical study, a standard CP910 audio processor with a standardized ‘flat map’ was used with all T- and C-levels globally set to 82 and 166 current units. A flat map is used so that the auto-precT procedure without knowledge of the C-levels can be performed with a non-individually programmed processor. Cochlear’s ACE strategy was used as a stimulation strategy. The other settings of the flat map were as follows: The value for ‘maxima’ was set on ‘1’, in order to ensure that only one electrode was stimulated at a given time by the sinusoidal audio input via the PAC. The pulse width was set to 25 µs and the stimulation rate to 900 pps. Furthermore, the T- and C-level were globally set to 82 and 166 current units. These values are based on a clinical database of fitting parameters of Cochlear implants using Cochlear Ltd. Devices (Bewley, 2013). The whitepaper states that about 95 percent of implants have the threshold levels set within that range.

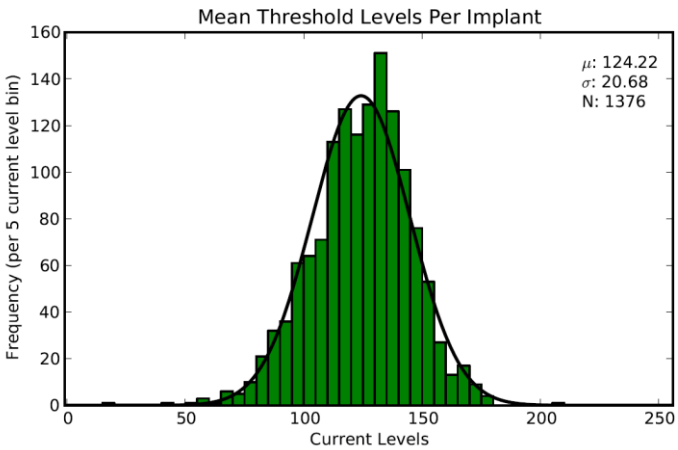


Figure 14. Histogram of Mean T-levels of CI-users using Cochlear Ltd. devices. About 95 percent of thresholds levels are within the range of 82 to 166 current. Cochlear Whitepaper (Bewley, 2013).

The other map parameters remained on factory default; especially T-SPL and C-SPL remained at 25 dB and 65 dB (Figure 15). Input sounds with a SPL below the T-SPL do not lead to a stimulation and input sounds with a SPL above the C-SPL lead to a stimulation at the set C-level.

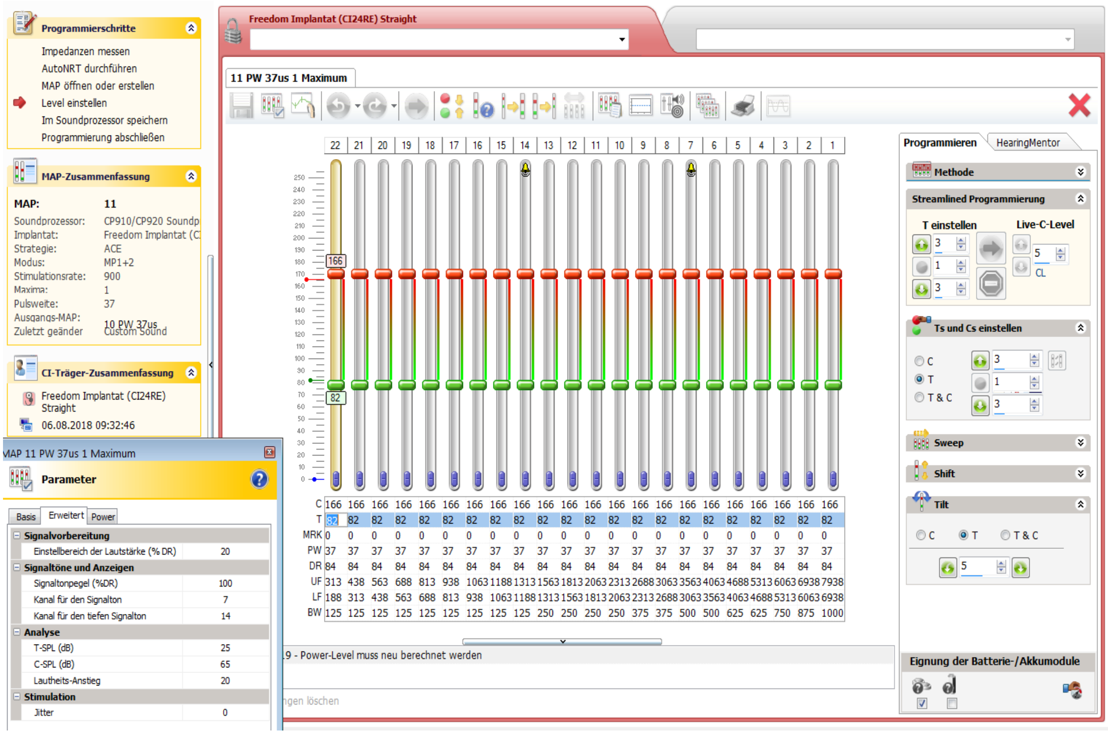


Figure 15. 'Flat Map' used for the calibration and the evaluation study. T-levels were globally set to 82 cu, C-levels to 166 cu. T-SPL and C-SPL were kept at 25 dB and 65 dB.

3.3. Calibrating electrode specific stimuli

In order to stimulate specific electrodes, sound files with the middle frequencies of the corresponding band-pass filters were created in *MatLab*. As the value for 'maxima" on the audio processor was set to 1, only the targeted electrode was stimulated. It was challenging to generate and calibrate stimuli evoking the intended specific current level, because the correlation in between the digital input in MatLab and the current level it evokes was unknown at first. In order to generate stimuli that evoke different current levels an approach with 'attenuation factors' *F(k)* was chosen (Figure 16). The sinusoidal stimuli were multiplied by the attenuation factors and the evoked current levels could be measured using the DIET. Thereby, the correlation between the applied attenuation factors and the current levels could be evaluated.

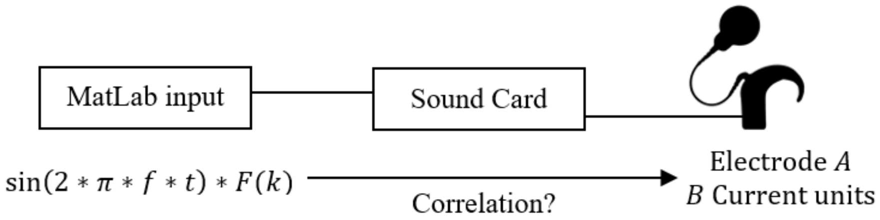


Figure 16. Calibration approach: Stimuli were sinus signals created in MatLab with the specific frequency *f* and an attenuation factor *F(k)*. The audio signals stimulated a specific electrode *A* with a current level of *B* current units. The stimulation data was logged with the DIET, so the correlation of the attenuation factors and the current level the audio signals evoked could be evaluated.

As a premise, each attenuation factor *F(k)* should lower the sound signal by 1 dB. Subsequently, the attenuation factors could be calculated with the following formula:

$$F(k) = 10^{-k/20}$$

The sound card settings were chosen so that an unattenuated stimulus evoked current levels of 166 cu (=C-level, maximum stimulation level) at every electrode. The PAC connected to the audio input of the audio processor have a frequency-specific transfer function. The frequency range of the input signal is divided into different stimulation channels. This results in different attenuation factors for the stimulation channels. In order to evaluate the correlation of the current levels evoked by the stimuli and their attenuation factors, the stimulation data was measured with the DIET. Figure 17 displays the correlation for all electrodes. Between factor 20 and 40, measurements were only performed for every fifth attenuation factor. The values in between were interpolated.

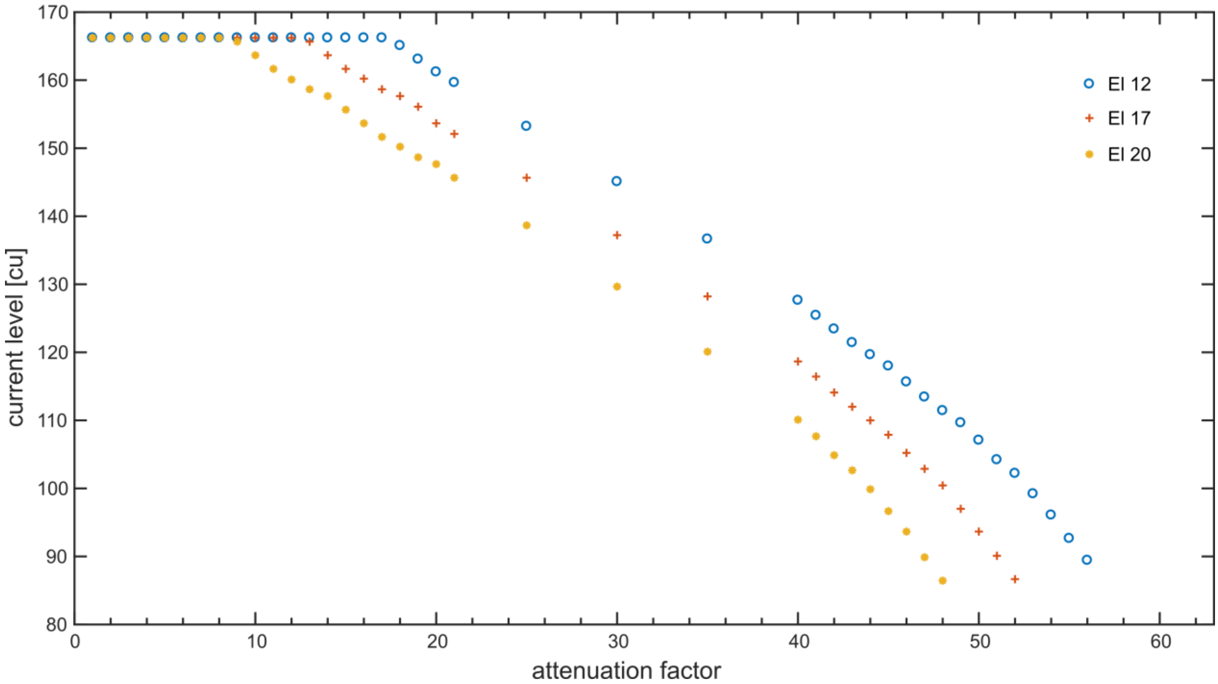


Figure 17. Correlation of the current levels evoked by the audio signals generated with MatLab and their attenuation factors. Exemplary data for three electrodes. In the linear range in the middle, measurements were only made for every fifth attenuation factor.

Figure 18 displays the correlation for all electrodes. The attenuation factors are indicated by the bold numbers.

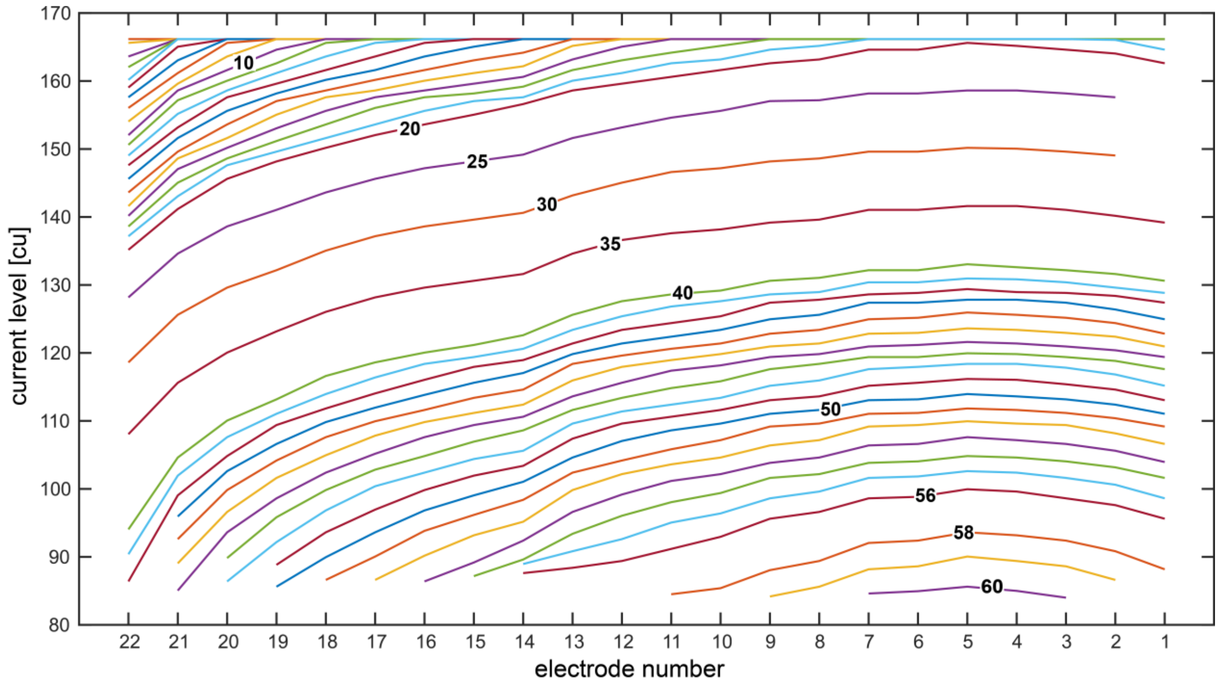


Figure 18. Correlation between current levels evoked by the stimuli generated with MatLab and their attenuation factors (bold numbers). Stimulation data was measured

with the DIET. Between factor 20 and 40, measurements were only performed for every fifth attenuation factor and for factor 57 no data was measured. The values in between were interpolated.

A calibration matrix was created with this data (Table 2). With the calibration matrix the attenuation factor that is needed to generate a stimulus that evokes a specific current level could be interpolated.

Table 2. Calibration matrix

	el = 22	…	*el = 1*
F(1)	current levels		
…			
F(65)			

Calibration matrix with the current levels that have been measured for the different attenuation factors.

Several MatLab scripts were used in the calibration process (Figure 19). Shortly before running the script that plays the sounds, the script for the data logging was started. The sounds were generated sinus tones with the middle frequencies of the band-pass filters of the corresponding electrodes and were presented for one second. After a short break the next sound was played. The stimulation data measured with the DIET was logged in a separate file for each attenuation factor. The output file was a matrix that contained the following information: the evoked current level, the stimulated electrode and a time stamp. Figure 20 shows a plot for the data in an exemplary output file. Applying another MatLab script, the median current level during stimulation was calculated for every electrode and assigned to the corresponding attenuation factor, so that an array of the 22 current levels was obtained for each attenuation factor. Put together these arrays formed the calibration matrix as described above.

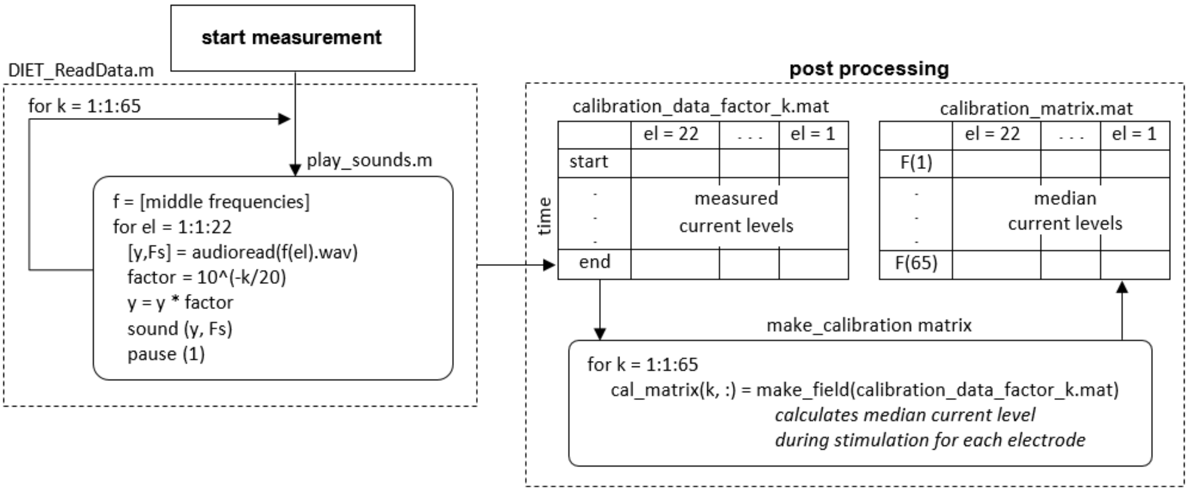


Figure 19. Process chart of the MatLab implementation for the calibration measurement and the post processing of the data in order to obtain the calibration matrix. An exemplary plot for the data logged for the stimulation of all electrodes with one attenuation factor is shown in Figure 20.

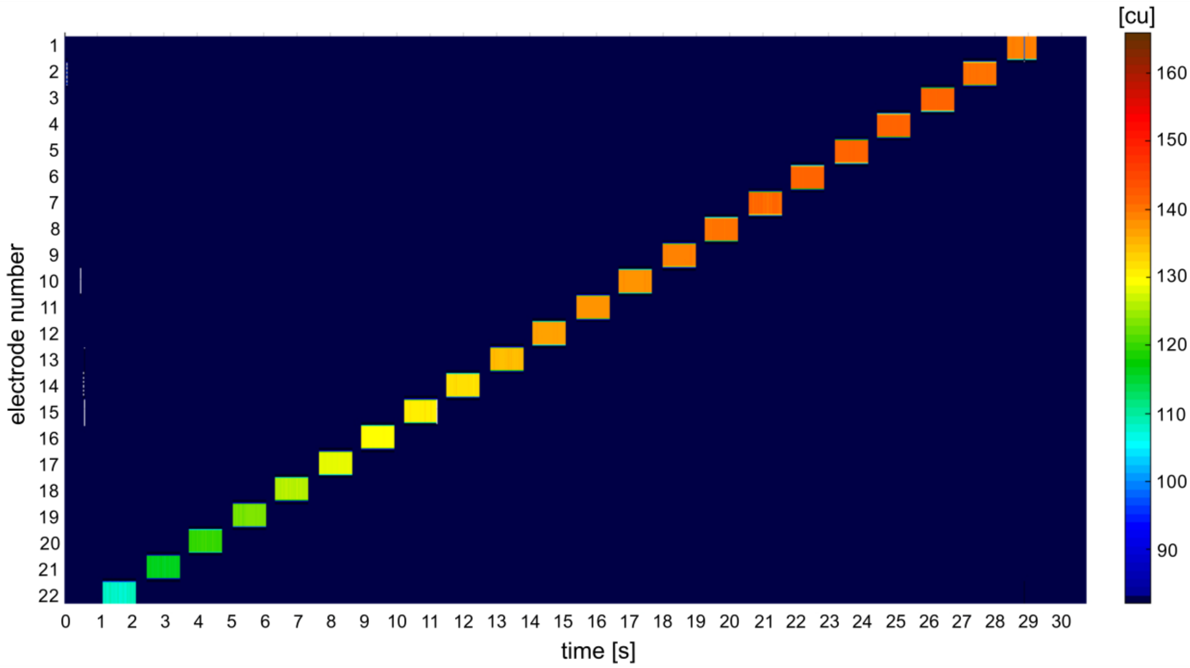


Figure 20. Exemplary calibration measurements with MatLab: stimulation data logged with the DIET over time shown for attenuation factor 35. Color coding from 82 to 166 cu.

The accuracy of the calibration that has been previously described was assessed in the following way: the hardware was set up in a soundproof room as shown in Figure 13. A MatLab script was run and generated audio signals that stimulated every electrode, one after another, with the same intended current levels. The stimulation data was logged with the DIET. Hence, the evoked current levels could be compared with the MatLab input. The assessment showed that the calibration and the hardware setup were sufficiently accurate (Figure 21). Only small deviations (standard deviation was 0.37 cu; that is less than 1 % of the average dynamic range) and minor fluctuations were observed. The reason for the deviations was the rounding of the attenuation factors. Minor fluctuations at the individual electrodes were caused by the setup itself: When using the clinical fitting software to stimulate single electrodes, the stimuli are directly generated by the sound processor. In our setup the stimuli were evoked by an audio signal. The processing of the audio signal caused the minor fluctuations.

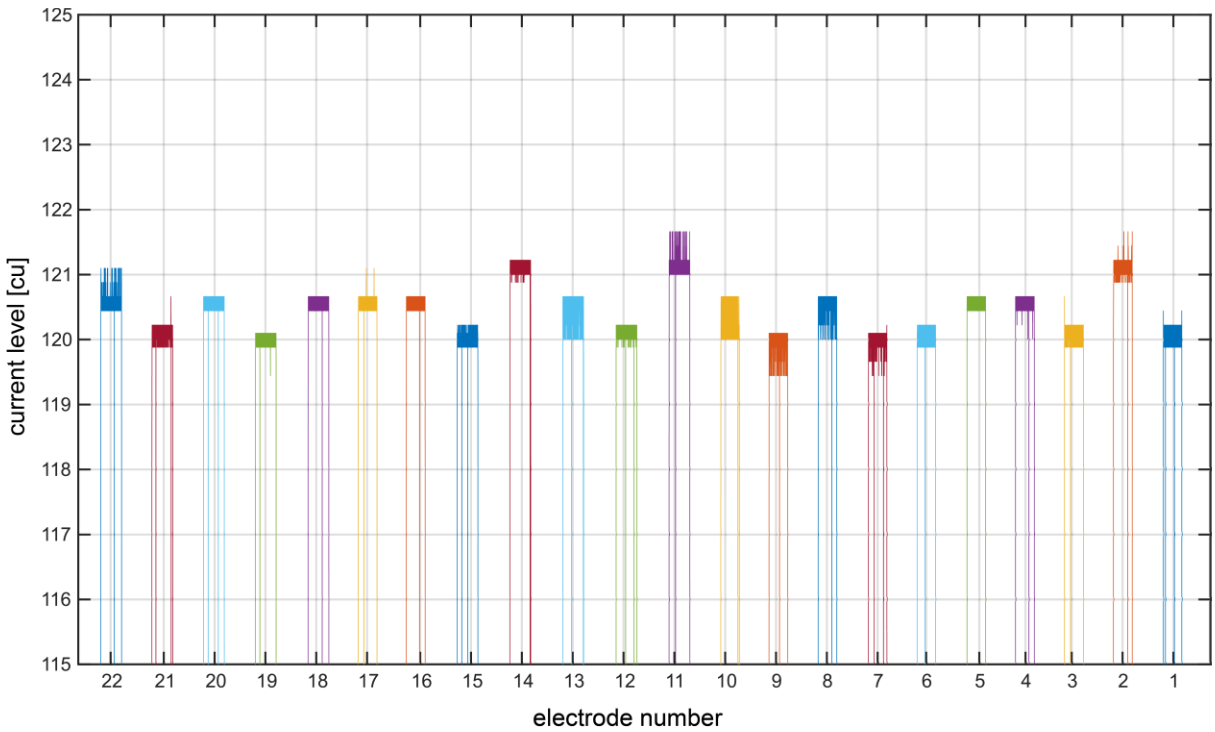


Figure 21. Exemplary calibration assessment – the audio signals created with a MatLab script should evoke a current level of 120 cu. The figure shows the stimulation data measured with the DIET. The calibration was sufficiently accurate. The small deviations were due to rounding of the attenuation factors. Minor fluctuations at the individual electrodes were caused by the processing of the audio signal.

After successful calibration of the setup, the precT-procedure was implemented in MatLab. This process is described in the following section.

3.4. MatLab implementation of the AFC procedure

For standardization of the precT procedure (see section 3.1), it was implemented in MatLab. Several scripts, functions and a graphical user interface (GUI) were developed. The following figure shows the components of the created MatLab surrounding.

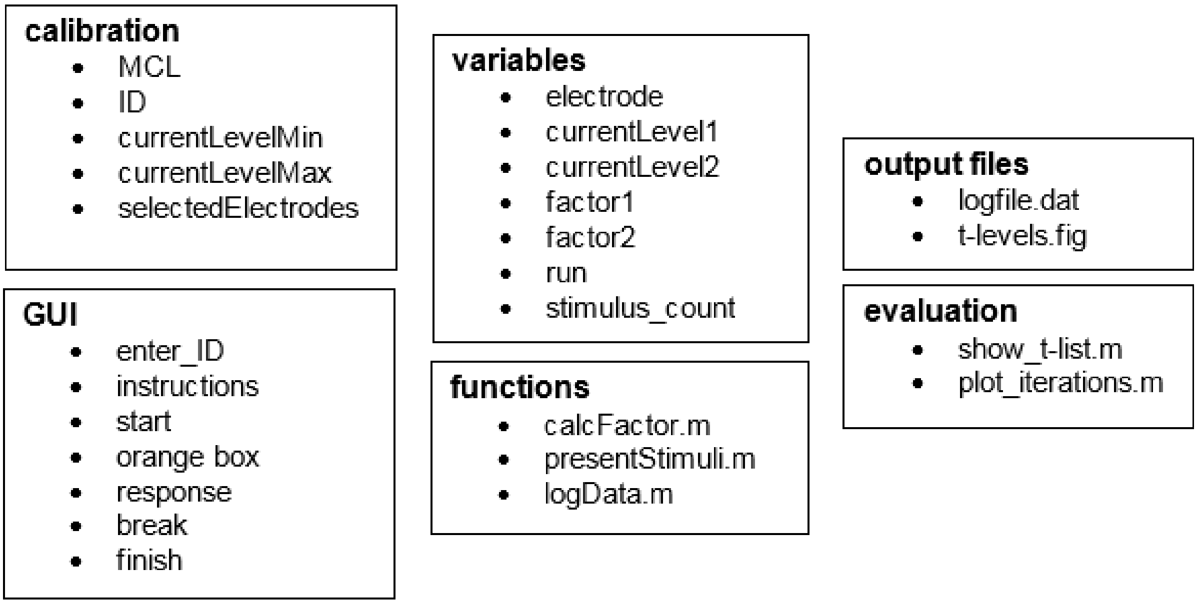


Figure 22. Overview of the auto-precT application components

The aim was to create a self-explanatory application. Hence, a GUI that is easy to understand and interact with had to be designed (Figure 23). When the auto-precT application started, the patients' ID was entered at first. Before the patient began with the AFC procedure, a window with text explaining the application was displayed. Next, an example with two stimuli was presented– one at the starting current level and one 10 cu below. Afterwards, an interactive window appeared asking the patient whether he had any further questions and if he was ready to start. Then the AFC procedure started. A window with an orange box appeared. Only during the time, the orange box was visible, audio stimuli were to be perceived. To prevent a habituation effect, the order of the stimuli was randomized, so at times the stimulus with the higher current level is played first and at times the stimulus with the lower current level. Furthermore, the time before the first stimulus is presented is randomly varied (1 to 2 seconds) and likewise the time between the first and the second stimulus (1.4 to 2.4 seconds). After the presentation of the stimuli a response window is shown, giving the question how many sounds were heard. The subject pressed zero, one or two. Then the next iteration started. After the subject finished the three runs for an electrode, a window appeared with a button saying, ‘continue with the next electrode’. This was implemented to give the patient an opportunity to take a break if needed, as the procedure requires a relevant degree of concentration.

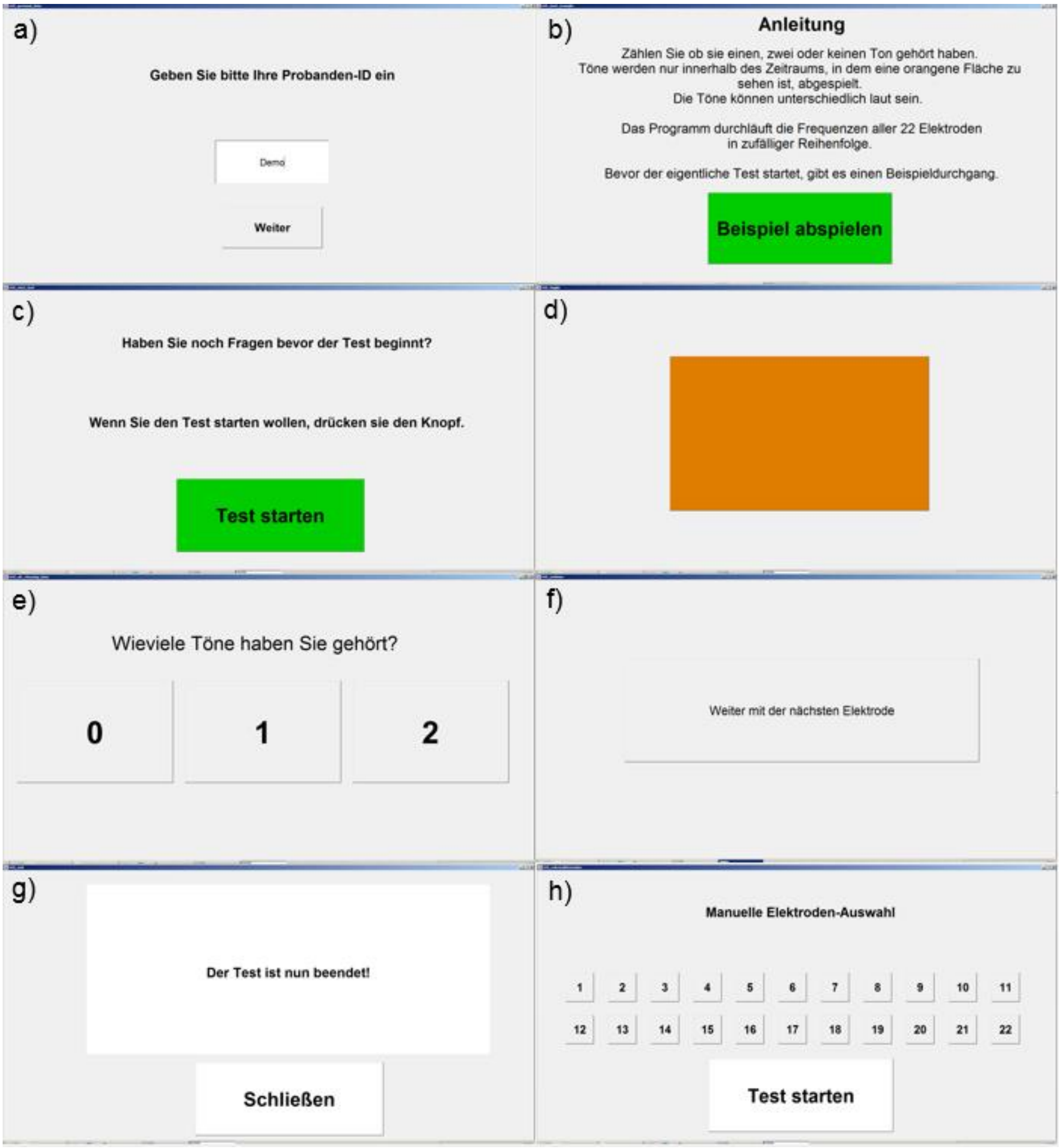


Figure 23. GUI screenshots: a) starting window, entering the patients’ ID; b) instructions and button to play an example; c) window with a button to begin the application; d) orange box, which is only visible during the time stimuli can be perceived; e) response window; f) window, with ‘continue with the next electrode’ button, that is displayed after the three consecutive runs for one electrode are finished; g) finishing window at the end; h) window that is shown at the start if the option to manually select electrodes for testing was chosen.

Figure 24 shows the basic structure of the code of the MatLab application. In a configuration script different values and options for the procedure could be set, for example the limits for the stimuli levels. Accordingly, to the ‘flat map’ the maximum current level was set to 166 cu and the minimum current level to 82 cu. The start current level was set to the most comfortable level (MCL = C-Level) or to 166 cu if the MCL was higher than that. The array ‘El [el_count]’ contained the electrode numbers in a pseudo-randomized order. The function ‘CalcFactor’ calculated the attenuation factor needed to evoke the set current level at the electrode that was currently tested. ‘PresentStimuli’ plays the audio signals with the calculated attenuation factors and shows the GUI with the orange box. The application continually logged the data, so if there’s a pc problem or the subject interrupted the procedure no data is lost.

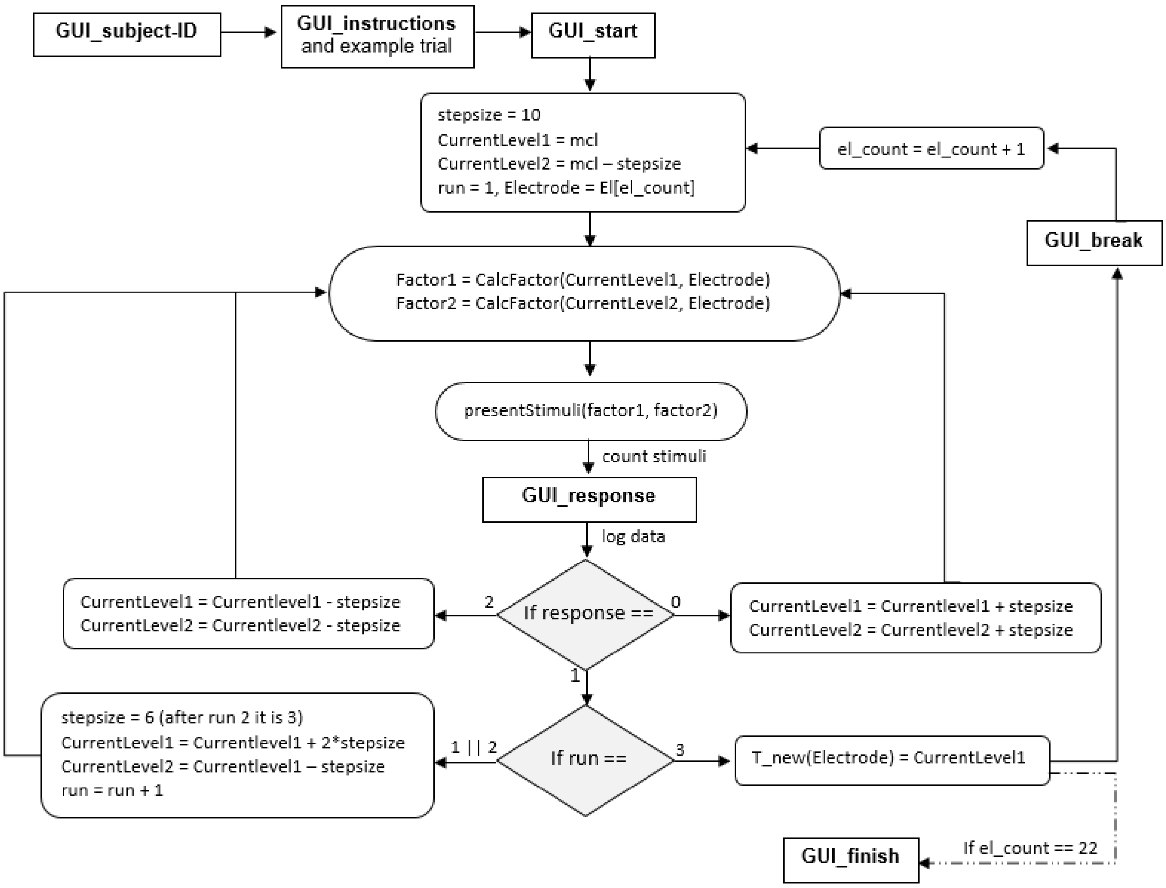


Figure 24. Process chart of the MatLab application. ‘El [el_count]’ represents an array with the electrode numbers in a pseudo-randomized order. ‘CalcFactor’ calculated the attenuation factor needed to evoke the targeted current level at the electrode that is currently tested. ‘PresentStimuli’ played the audio signals with the calculated attenuation factors and shows the GUI with the orange box.

The logfile of the MatLab application contained all relevant information: stimulation data (current levels, electrodes), the subject's responses and a time stamp. Several additional functions and options were created - for example the option to manually select which electrodes should be tested (Figure 23h). This offered the opportunity to repeat the procedure for selected electrodes or to only test specific electrodes in general. Furthermore, scripts for the evaluation of the measured data were developed, e.g. a script that plotted all iterations for each electrode in one figure thereby giving an overview of the whole procedure.

4. Evaluation of the new application

In order to evaluate the auto-precT application, regarding applicability and speech perception outcome, a clinical study was conducted. The study consisted of three parts. First, the speech perception of the subjects with their current sound processor settings was tested (P1). Then, the subjects worked through the newly developed application and determined their threshold values on their own. After that, a map with T-levels set to the determined hearing thresholds (P2) and another map with T-levels 10 cu lower than thresholds determined with the auto-precT application (P3) were created. Subsequently, the speech perception with those maps was tested.

4.1. Subjects

Fifteen experienced CI-users with a CI usage from 7 to 124 months (median: 21) participated in this prospective study. The age ranged from 20 to 71 with a median of 56 years. Criterion for inclusion in the study was good speech perception with a Freiburg Monosyllable Score (FMS) of 60 percent or higher at 65 dB SPL free field presentation level. Five participants were bilaterally implanted. In this case only the better performing ear (higher scores in speech perception) was tested. The contralateral ear of subjects with residual hearing was masked with an earplug during all tests. All subjects had an implant by COCHLEAR (Macquarie, Australia) and were using a CP810, CP910, CP950 or CP1000 sound processor and the speech coding strategy 'ACE' (Arndt et al., 1999). The other established processor settings were following: stimulation rate 900 pps; stimulation mode MP1+2; maxima 8; pulse width 25 μs (n = 3), 37 μs (n = 11), 50 μs (n = 1). The subject demographics are shown in Table 3.

Table 3. Demographical data of study participants

Subject ID	Age	Sex	Tested side	Implant use (months)	CI processor type	Implant type	Subject etiology	FMS (65dB) with CI
01	56	M	left*	14	CP 910	CI 522	Progressive	70
02	63	M	left	48	CP 910	CI 422	Progressive	70
03	51	F	left	13	CP 950	CI 522	Progressive	70
04	51	F	right*	7	CP 1000	CI 522	Progressive	100
05	62	F	left	38	CP 810	CI 422	Infectious	60
06	53	F	left	12	CP 950	CI 532	M. Meniére	60
07	58	M	right*	54	CP 910	CI 522	Progessive	70
08	61	F	right	21	CP 910	CI 522	Infecstious	100
09	37	M	left	33	CP 910	CI 522	Hereditary	80
10	69	M	left*	31	CP 910	CI 512	Hereditary	90
11	56	F	left	14	CP 950	CI 522	Ototoxic	85
12	41	M	right	124	CP 910	CI24RE	Infectious	95
13	71	M	right	18	CP 950	CI 522	Progressive	80
14	30	M	right*	18	CP 910	CI 522	Hereditary	70
15	20	F	right	65	CP 810	CI 422	Ototoxic	100

FMS values of the ear, that was tested in the study; * bilaterally implanted

The study was approved by the local ethical review board (Landesärztekammer Rheinland/Pfalz, 837.462.17(11296)) and informed consent was given by the participants.

4.2. Experimental Setup

For the assessment of speech perception, sound presentation was realized using a computer equipped with a high-quality 24-bit, 8-channel AD-DA converter (RME Fireface UC, Haimhausen, Germany) connected to an active loudspeaker (KS Digital C5, Saarbrücken, Germany) placed in front of the subject. The speaker was placed in a soundproof room at 0° azimuths and a distance of 100 cm to the subjects' ears. Free field stimuli were calibrated at listening position according to the manufacturer's instructions using an Audio XL2 sound pressure level meter (NTI, Schaan, Liechtenstein).

4.3. Speech perception in noise

In order to evaluate the speech perception in noise, the closed-set German matrix test ‘Oldenburger Satztest” (OLSA) was used. The test was conducted with a set speech level of 50 dB SPL presentation level. The noise level is adjusted after each trial according to the amount of correctly recognized words. The adaptive procedure determines the signal to noise ratio (SNR) at which 50 percent of the words were understood. The speech perception test was conducted with three different conditions that were programmed on the study sound processor:

- (P1), T_established: The established map that the patient was currently using. T-levels and other parameters were not changed. The T-levels had been adjusted by the common clinical procedure
- (P2), T_auto-precT: A map with the T-levels set to the threshold values determined using the proposed auto-precT application
- (P3), T_auto-precT-10: A map with T-levels 10 cu lower than the determined threshold levels to simulate underestimated T-levels

Before running the auto-precT application, the subjects performed two OLSA test runs with their familiar sound processor settings (P1) – the first one was a training run in order to get used to the speech test, the second run was used for evaluation (test 1). After the determination of the hearing thresholds using the auto-precT application, the participants carried out the OLSA again (tests 2 and 3) with the newly created speech processor settings (P2, P3). With each setting the OLSA was conducted twice. The better test result out of two was used for further analysis. The tests were executed in a randomized manner regarding the processor setting (P2, P3).

4.4. Hearing threshold determination with the developed application

For the autonomous hearing threshold determination, the participants used the study sound processor that was calibrated for the procedure. Processor settings were the same for all subjects (flat map settings described in section 3.2). Only the pulse width was changed in two cases: (1), if the pulse width set in the subject’s established settings was different from the default value of 25 μs in the flat map–then the pulse width in the study processor was changed to individual processor value set in the established map (for three subjects the established pulse width was 25 μs, for eleven subjects 37 μs and for one subject 50 μs); (2), if threshold levels in a subjects established map were below 82 cu or above 166 cu (T- and C-levels in the 'flat map'). When a lower pulse width is applied, a higher current level is needed to evoke the same electric charge and vice versa. So, if the subject's T-levels were out of the range of the flat map (82/166 cu), the pulse width in the study processor was altered so that the T-levels with the altered pulse width were within the range of 82 and 166 cu (pulse width in the study processor needed to be lowered for three subjects from 37 μs to 25 μs, who had T-levels below 82 cu in their established map, and to be raised for from 25 μs to 37 μs for one subject, who had T-levels above 166 cu in his established map). For those four subjects for whom a pulse width different from the established processor setting was applied in the study processor, the measured threshold levels needed to be converted for the evaluation in order to be comparable to the established T-levels, due to pulse width affecting T-levels. For the conversion the T-levels determined with the study sound processor were first transformed into microampere with the formula $I_1[\mu A] = 17.5*100*(I_1[cu]/255)$. Subsequently, the corresponding amperage (I_2) for the pulse width in the established map (PW_2) was calculated with following formula: $I_2 = Q/PW2 = I_1*PW_1/PW_2$. Last, the calculated amperage was transformed back into current units.

The audio signals were directly transmitted from the sound card to the calibrated CP910 'standard speech processor' via a standard audio cable. Feedback from the subject was collected using a touch screen monitor with a graphical user interface. Figure 25 shows the hardware setup for the study.

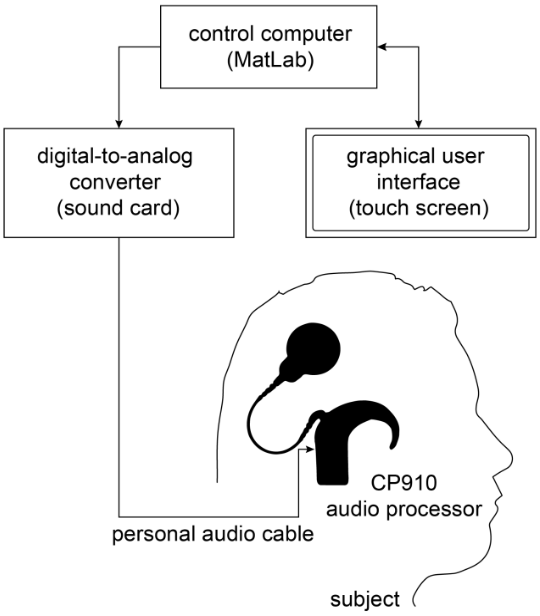


Figure 25. Hardware setup for the study. Stimuli generated with the MatLab application were transmitted to the audio processor with a personal audio cable. Subjects' responses were collected with a touch screen on which the GUI of the application was shown.

4.5. Subjective preference

After programming the three different map conditions (programs), they were varied during a five-minute conversation with the subjects. Programs were varied every 30 to 60 seconds. The patients were asked which map condition they subjectively would prefer for everyday use.

4.6. Statistics

Statistical analyses were performed with the software IBM SPSS Statistics 23. For all test variables, a Shapiro-Wilk test confirmed normal distribution. Differences in between OLSA scores were tested with a t-test for paired samples. P-values below 0.05 were considered as significant. Unless stated otherwise, the analyses are based on the data for all n = 15 subjects.

5. Results

5.1. Feasibility and duration of the auto-precT application

All patients were able to perform the auto-precT application independently. 11 subjects performed the application taking only smaller breaks (< 5 minutes). The remaining four subjects took longer breaks, mostly because they needed to use the bathroom. Three of those four subjects needed an overall time of 53 to 56 minutes to complete application including the breaks. One subject, who took multiple breaks due to strong concentration problems, needed 69 minutes. Test duration was statistically evaluated only for the 11 subjects who took short breaks, in order to be able to compare the duration to our previous study (Rader et al. [11]). In that study the precT procedure was manually performed using the clinical fitting software Custom Sound and no breaks were reported. In the current study the 11 subjects performed the auto-precT application, executing three consecutive runs for all 22 electrodes, in an average time of 39 min (approximately 107 s/electrode), including the smaller breaks they took - that is 7 minutes less than in the previous study by Rader et al. (2018). The average time needed for the first run was 14:12 (min : s), for the second run (step size 6 cu) it was 11:24 and for the third run (step size 3 cu) 13:24. No strong correlation between age and time needed was observed.

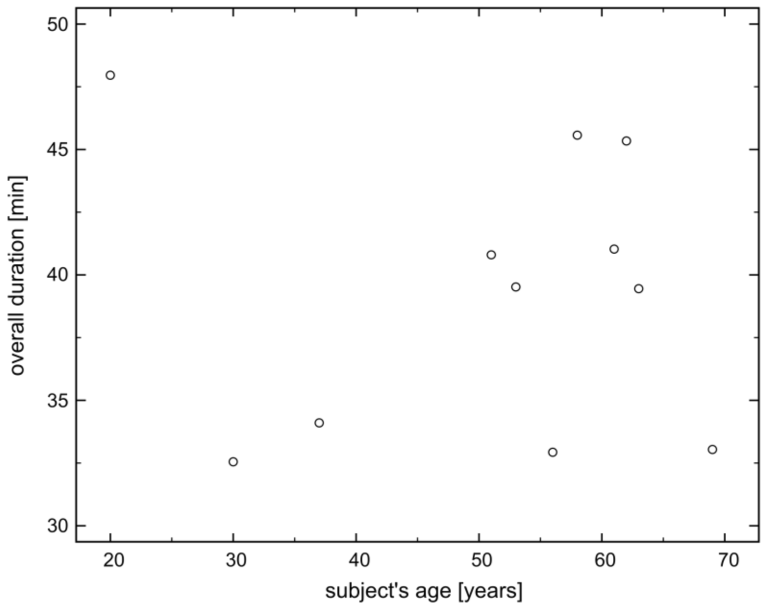


Figure 26. Time subjects needed to determine their hearing thresholds with the auto-precT application. Duration of the application was evaluated for the n = 11 subjects who performed the application without longer interruptions (breaks longer than 5 min).

5.2. Threshold values determined with the auto-precT application

The mean difference of the threshold values determined with the auto-precT application and the established T-levels is shown in Figure 27-A. The median mean difference is -0.7 cu, but the values are broadly spread. Threshold values determined with the auto-precT application were higher than the established ones for some subjects, for some they were lower. Figure 27-B shows the mean difference relative to the dynamic range (median: -2.2 %). This plot has been included, because some authors, e.g. Busby and Arora (2016), altered the T-levels in order to reach set percentages of compression or expansion of the EDR. The analysis of the distribution of the mean absolute values for the difference of T-levels revealed a median of 10.5 cu, indicating that T-levels were shifted in both directions (Figure 28).

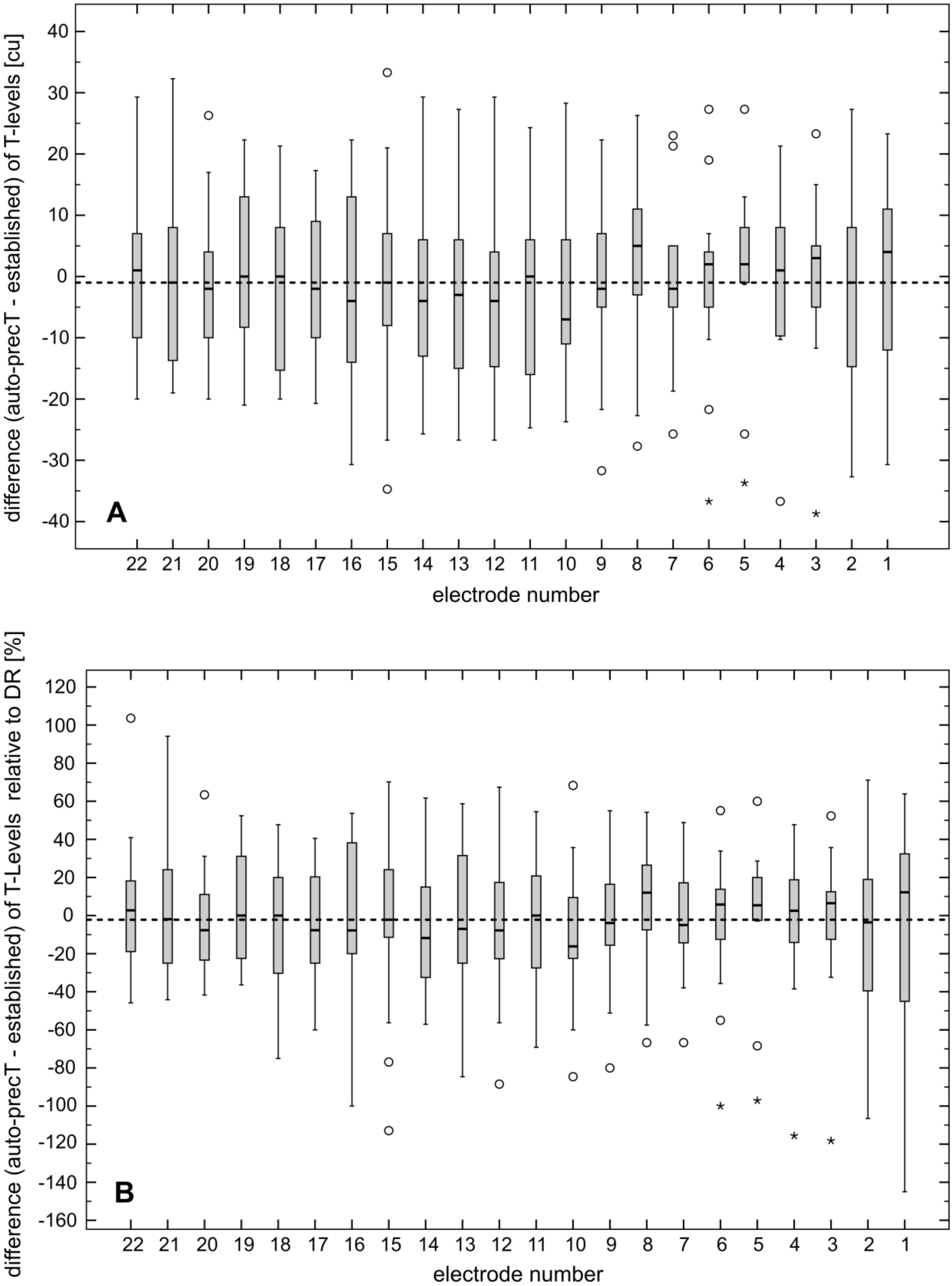


Figure 27. A: **Difference between the established T-levels and those determined using the auto-precT application.** B: Difference between the established T-levels and those determined using the auto-precT application relatively to the dynamic range of each electrode (median: -2.2 %). The box plot contains median, 1st and 3rd quartiles, minimum and

maximum values. The circles indicate mild outliers (>1,5*IQR from the first or third quartile), the asterisks indicate extreme outliers (>3*IQR from the first or third quartile). Dashed line indicates median over all electrodes.

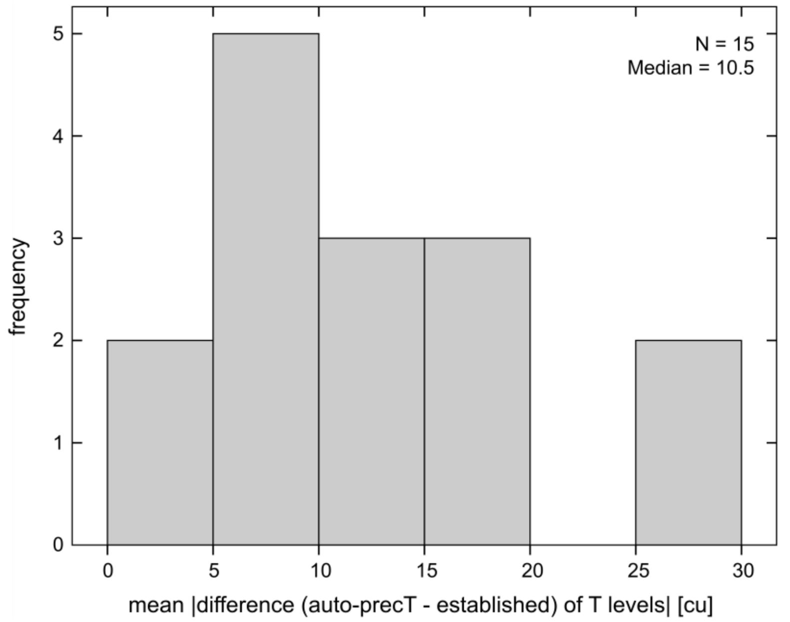


Figure 28. Distribution of the values for the mean absolute differences between the T-levels determined using the auto-precT application and the established ones.

5.3. Subjective Preference

Asked for the preferred map conditions for everyday use, 13 of 15 subjects chose the auto-precT condition. One participant chose the established settings and one the auto-precT$_{-10}$ condition.

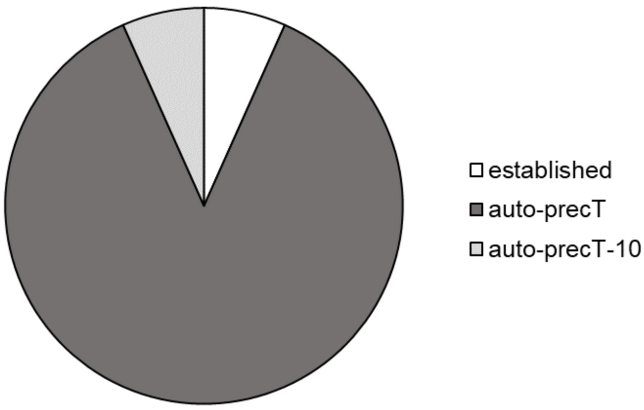


Figure 29. Subjectively preferred map condition. While talking to the subject the map conditions were varied and the subjects were asked which condition they would prefer for everyday use.

5.4. Speech perception in noise at 50 dB SPL speech level

Figure 30 displays the results of the speech perception tests in noise at 50 dB SPL speech level. Median speech reception thresholds were significantly improved (p = 0.02) with T-levels set to those determined with the auto-precT application (P2) compared to the $T_{established}$ (P1) condition from 2.5 dB SNR to 1.6 dB SNR. Speech perception was lowest with the globally lowered T-levels (P3), (median: 2.9 dB SNR).

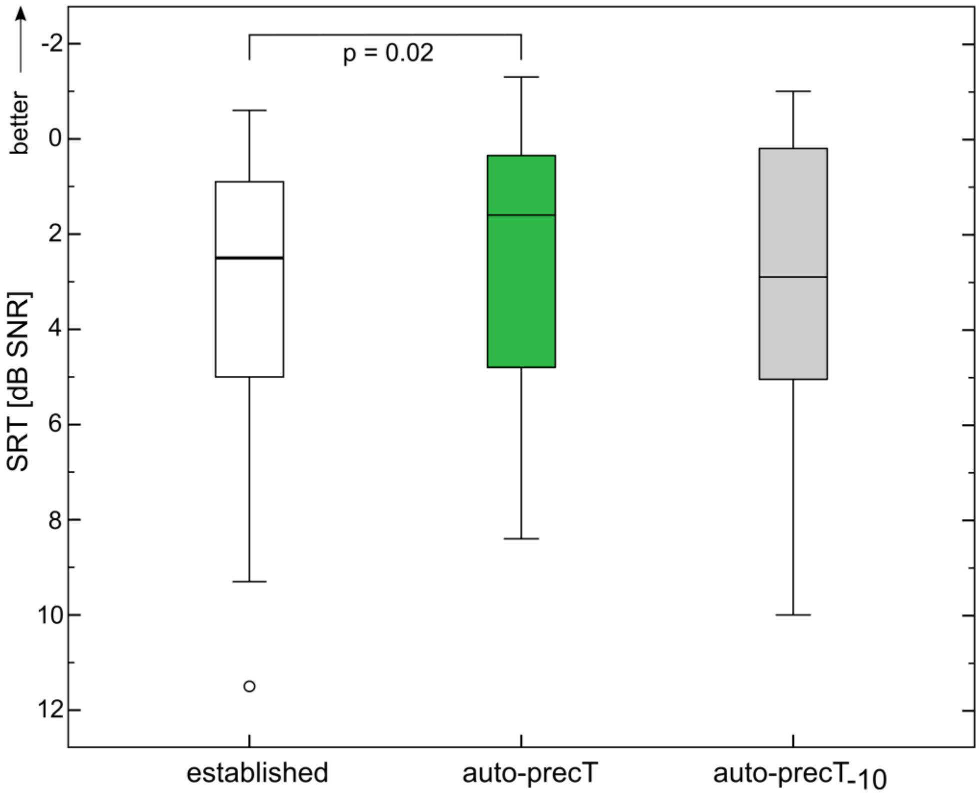


Figure 30. Speech reception thresholds in free field conditions (50 dB SPL speech level, noise adaptive) with three different settings for the electrical thresholds. Established: T-levels used by the subject prior to the testing; auto-precT: T-levels set to the thresholds determined with the proposed method; auto-precT$_{-10}$: T-levels set 10 cu lower than the thresholds determined with the auto-precT application. The box plot contains median, 1st and 3rd quartiles, minimum and maximum values. The circles indicate mild outliers (>1.5*IQR from the first or third quartile).

6. Discussion

6.1. Impact of T-level settings on speech perception

The evaluation of the impact of the T-level shifts on speech perception showed interesting results (Figure 31). All subjects with increased T-levels, on average over all electrodes ($n = 6$), had an improvement in speech perception (upper right quadrant in the figure). Corresponding to that, most subjects with decreased T-levels had deterioration in speech perception (lower left quadrant in the figure). Consequently, the results might suggest that compression of the EDR leads to an improvement in speech perception at soft speech presentation levels (Pearson correlation coefficient ($r = -0.8$, $p < 0.01$). This observation is similar to the results of the previous study from Rader et al. (2018) where a mean elevation of T-levels by 9 cu led to an improvement in speech perception at 50 dB SPL presentation level. However, it needs to be considered that also subjects with only a small mean shift of T-levels had better scores in speech perception. This is probably due to the precise determination of threshold values for each individual electrode. T-levels were increased and decreased at different electrodes of the same subject, so there was only a small average shift. Therefore, a small average shift does not mean that only small changes of T-levels for the individual electrodes were performed. In summary, it is the precise determination of threshold levels that is beneficial for speech perception at soft speech presentation levels.

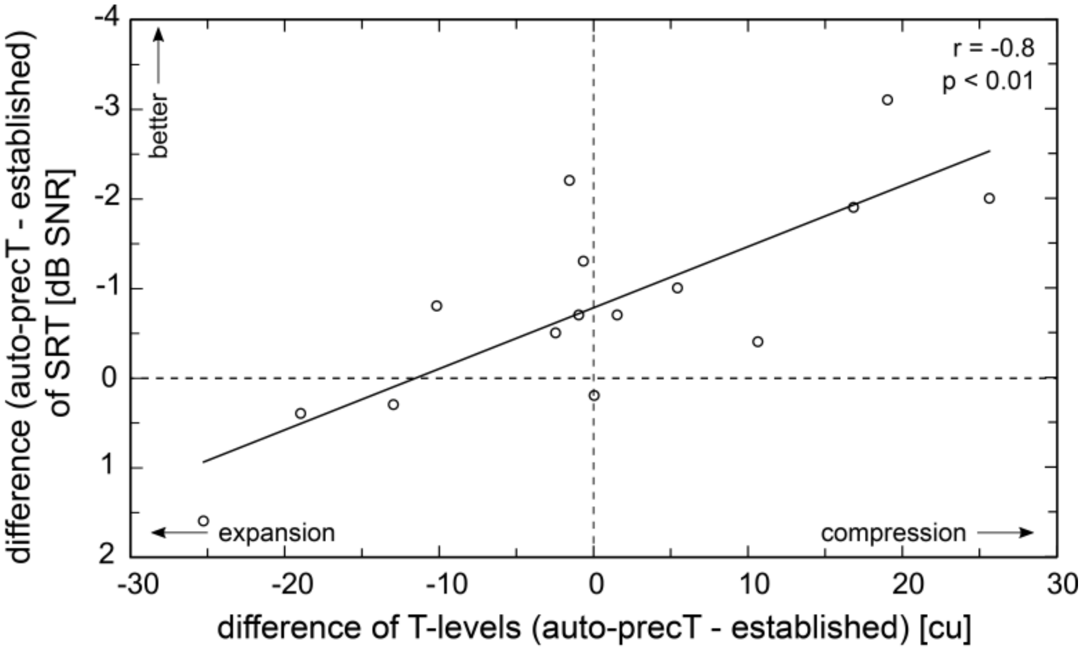
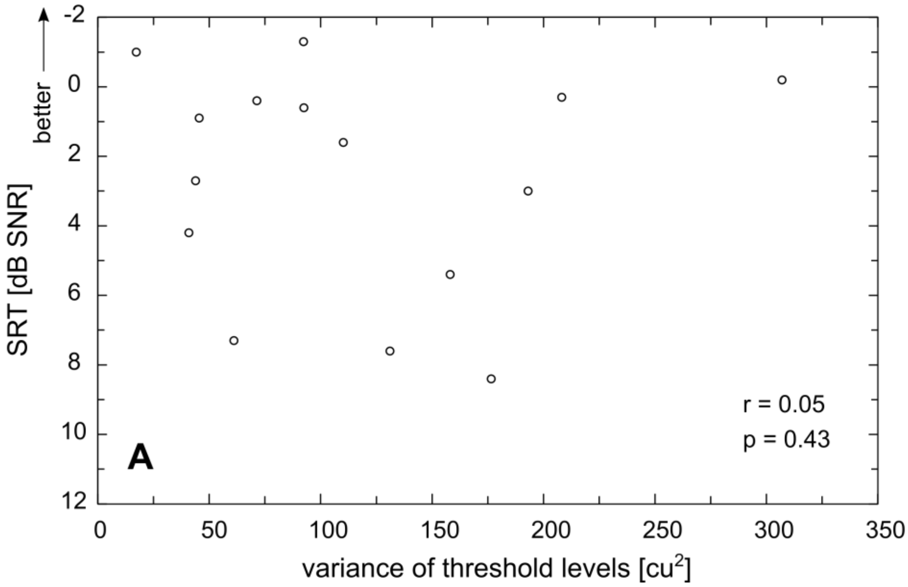


Figure 31. Correlation of the shift of T-levels and changes in speech perception. Compression of the EDR seems to lead to an improvement in speech perception at soft speech presentation levels (upper right quadrant).

In some studies, an improvement of speech perception at soft presentation levels by globally raising T-levels was described. reported that in their study, which they conducted with Nucleus 22 CI users, better consonant nucleus (vowel) consonant (CNC) word and sentence scores at 50 and 60 dB presentation level were observed with minimum stimulation levels raised by 2.04 dB above the clinically determined threshold values. It needs to be mentioned that the parameters (especially stimulation rate and strategy) were quite different from those applied in our study. In another publication, Holden et al. (2011) reported that setting T-levels higher than the recommendation (10% of C-Levels) of the manufacturer (Advanced Bionics CI, Auria and Harmony sound processors) was beneficial for overall speech perception. However, in line with the findings of our study, Holden et al. concluded that an exact determination of threshold levels is necessary.

Dawson et al. (2007) compared speech perception at three different input dynamic ranges of 31 dB, 46 dB and 56 dB. In contrast to our study in which a standard commercial sound processor was used, they used a SPEAR3 research processor for the testing. Another difference to our study sample was that all except one subject used the SPEAK coding strategy, whereas in our study all subjects used the ACE strategy. Speech perception was found to be better with the expanded input dynamic ranges of 46 dB and 56 dB. They reported that an additional reduction of T-levels did not lead to an improvement in speech perception. This is in line with the results for the speech perception scores measured using the auto-precT_{-10} condition in our study.

Pfingst et al. (2004) investigated two hypotheses regarding behaviorally measured threshold values. They suggested that the threshold levels reflect the distance in between an electrode and the auditory nerve fibers it is stimulating and that this distance is influenced by neuropathology along the cochlear spiral. Consequently, they created the following hypotheses: (1) that the across-site variation of thresholds is correlated with the performance in speech perception and (2) that lower average thresholds should correlate with better speech perception scores. Their results showed (1) a negative correlation of across-site variance of thresholds and speech perception, but (2) no correlation between mean threshold levels and speech perception. Our study data were analyzed regarding the hypotheses of Pfingst. Results of speech perception tests conducted with the T-levels set to the threshold values determined with the auto-precT application were analyzed. In contrast to the findings of Pfingst, only a weak and not significant correlation between the variance of thresholds and SRT's was observed (Figure 32-A). With regard to the mean threshold levels, a not significant correlation was found (Figure 32-B). Higher threshold levels were associated with a little better speech reception (Pearson correlation coefficient $r = -0.33$, $p = 0.11$). Comparability of the studies may be limited, because another sound processor (SPrint) was used and another speech coding strategy (SPEAK) was applied by many subjects.



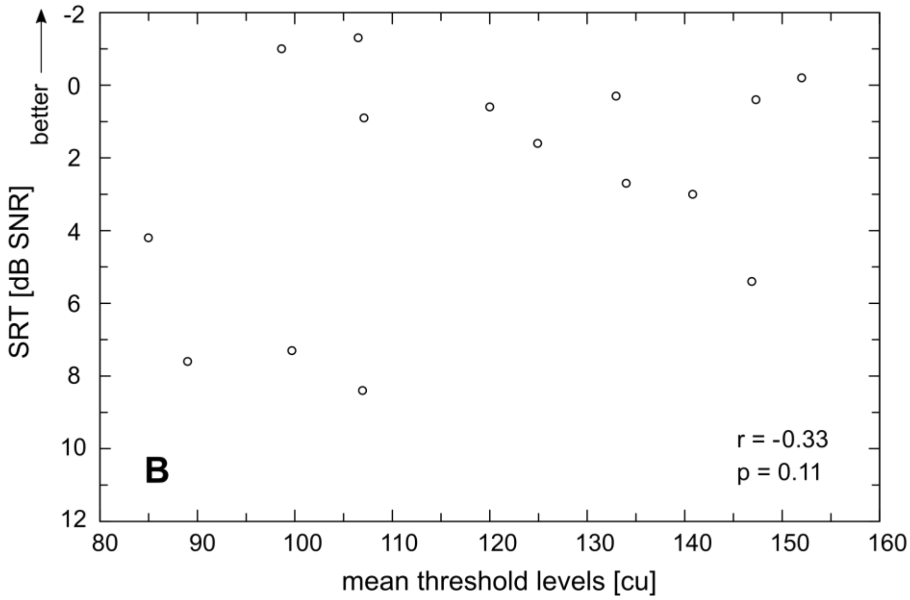


Figure 32. Impact on speech perception in noise (SRT) of (A) variance across all electrodes and (B) mean values of threshold levels (of all 22 electrodes). Data for speech perception tests conducted with T-levels set to the threshold levels determined with the auto-precT application.

In the same study in which Botros et al. (2013) introduced a new Nucleus cochlear implant fitting suite, they simultaneously presented and evaluated a new fitting methodology. They compared speech perception applying different maps. T- and C-levels were either set to the values determined with a remote fitting software (1), the nucleus fitting software (2) or to the values behaviorally measured for all electrodes (3). No significant difference in speech perception was observed. These findings seem to be contrary to our observation, that a precise determination of threshold levels is beneficial for speech perception. However, Botros et al. (2013) also stated that individual threshold measurements can improve sound quality for patients with difficulties in perceiving soft sounds. Moreover, the study showed that patients are capable of adjusting the fitting of their CI by themselves. One of the main motivations for the development of streamlined fitting methods like those used by Botros et al. (2013) is to free audiologists from the time-consuming task of measuring T- and C-levels for every electrode. In this regard, the proposed auto-precT application is a powerful tool. Patients can determine hearing thresholds for the individual electrodes themselves without an attending audiologist. Thus, the resources of an audiology department are saved while the workflow as well as the quality of the fitting is preserved or even improved. Even though the auto-precT application might be more time intensive for patients than the methods Botros et al. (2013) used in the remote fitting, the results of our study suggest that the additional time is most likely well spent.

Busby and Arora (2016) investigated the impact of varying T-levels from the actual threshold levels. They tested speech perception with five different conditions: T-levels decreased by 30 and 60 percent of the dynamic range and T-levels raised by 30, 60 and 90 percent of the dynamic range, C-levels were not changed. They reported that speech perception did not significantly change with raising or lowering T-levels by 30 percent, but stated that there was 'generally a negative impact for more compression or expansion'. Busby and Arora (2016) concluded that determining threshold values precisely might not be so important. This is contrary to our findings, as in our study many subjects had an improved speech perception with the precisely determined threshold values using the auto-precT application. Actually, in our view the results from the study of Busby and Arora (2016) do not allow to make a statement concerning the impact of precisely determining threshold value. The T-levels they set were not threshold values measured for each electrode, but interpolated values. They determined threshold values only for six electrodes, using the Hughson-Westlake procedure (Carhart & Jerger 1959). In this procedure, after an initial descent from a clearly detectable level below the hearing threshold, the stimulus level is increased by the set ascending step size until a sound is perceived. Next, the T-level is lowered by the set descending step size until no hearing sensation is perceived anymore and then increased again until the sound is detected.

This cycle is done until the level of the sound detection was the same for at least half of the iterations. Busby and Arora (2016) performed the procedure with an ascending step size of 2 cu and a descending step size of 4 cu for six electrodes and interpolated the values for the remaining ones. Our findings suggest that the precise determination of threshold values for every single electrode is beneficial for speech perception, especially at low speech levels.

At the Annual Meeting of the German Association of Audiologists (DGA) Mewes and Hey (2017) presented a study that also dealt with the impact of T-level settings on speech perception. They stated that their clinical experience contradicts the findings from Botros et al. (2013) as well as those Busby and Arora (2016). Mewes and Hey (2017) conducted speech perception tests with four different conditions: T-levels set 40 cu below C-level (T = C – 40 cu), T-levels set to the hearing thresholds determined with the common clinical procedure using the Nucleus fitting software (T = HT), T-levels lowered by 25 percent of the dynamic range (T = HT – 25% DR) and T-levels lowered by 50 percent of the dynamic range (T = HT – 50% DR). They tested speech perception in quiet with the 'Freiburger monosyllable test' (FMS) at 70 dB and with the 'Freiburger multi-syllable test'. In the latter, the threshold of 50% understood syllables was determined. Furthermore, speech perception in noise at 65 dB presentation level was assessed with the 'Oldenburger sentence test' (OLSA), which was also used in our study. They reported that the impact of the T-level settings on speech perception in quiet and on speech perception in noise was contrary. Lowering T-levels improved speech perception in noise at 65 dB presentation level, but worsened speech perception in quiet at low levels below 50 dB. The impact of expanding the dynamic range on speech perception in noise seems to be contrary to our findings at first glance. However, it needs to be considered that the speech perception tests were done with different conditions than in our study (speech perception in noise was tested at 50 dB presentation level), so it is difficult to compare the results. Nonetheless, our observations support the conclusion from Mewes and Hey (2017) that T-levels need to be individually optimized to reach the best possible speech perception in noise.

6.2. Impact of the electrical dynamic range on speech perception

Values for the electrical dynamic range (EDR) and its impact on speech perception in our study data were compared to the studies described above (Botros et al., 2013; Busby and Arora, 2016; Mewes and Hey, 2017) (Figure 33). The median EDR with T-levels set to the hearing thresholds determined using the auto-precT application was 39.8 cu. Busby and Arora reported that the median EDR was 39 cu with T-levels set to the hearing thresholds and Botros suggested a default EDR setting of 40 cu. Mewes and Hey reported that in their study the median EDR was 48 cu, when set to the hearing threshold. It is important to note that the values for the EDR of the individual subjects in our study were widespread, the standard deviation was 17 cu. That means that in order to at least include 68 percent of the patients, in terms of their EDR's, the EDR needs to be altered from the average value within a range of +/- 17 cu.

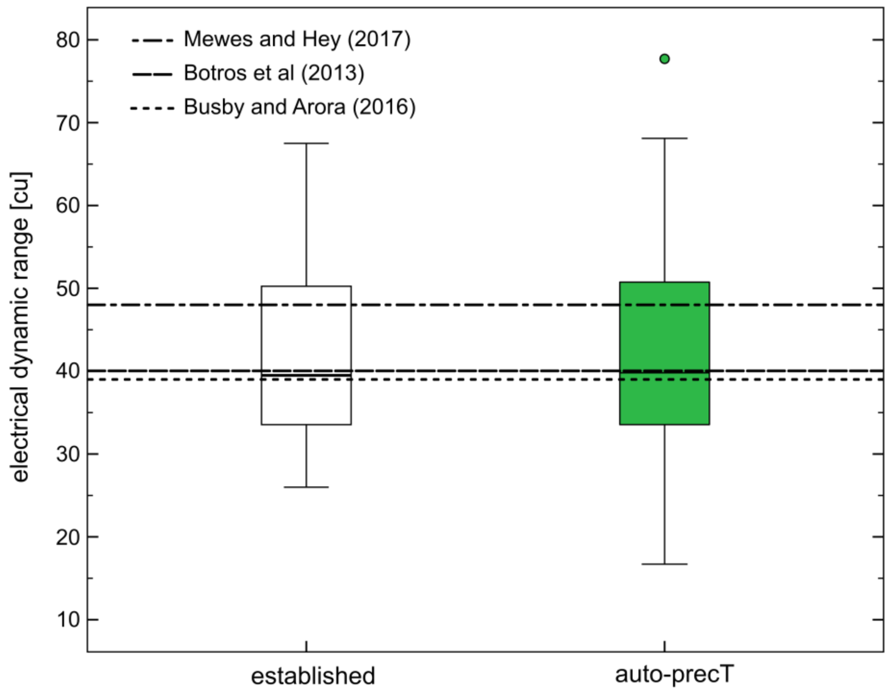


Figure 33. Electrical dynamic range (EDR) with T-levels set to the hearing thresholds determined using the auto-precT application. The box plot contains median, 1st and 3rd quartiles, minimum and maximum values. The circles indicate mild outliers (>1.5*IQR from the first or third quartile). Furthermore, the plot contains the median EDR's with T-levels set to the measured hearing threshold in the studies from Mewes & Hey and Busby & Arora and the globally default EDR used in the study from Botros et al.

More relevant than the median values of the dynamic range themselves is its impact on speech perception. Therefore, the correlation between the three different EDR's, which correspond to the MAP conditions applied in our study, and speech perception was evaluated for each subject individually (Figure 34). Upwards pointing arrows indicate an improvement while downwards pointing ones indicate a deterioration. The arrows with full lines show the difference in speech perception between the established condition (P1) and the auto-precT condition (P2), those with dashed lines show the difference between the auto-precT condition (P2) and the auto-precT$_{-10}$ condition (P3). Figure A is visualizing the impact of the absolute values of the EDR. Figure B was created for the analysis of the data in terms of compression and expansion. It shows the changes of the EDR relative to the established one.

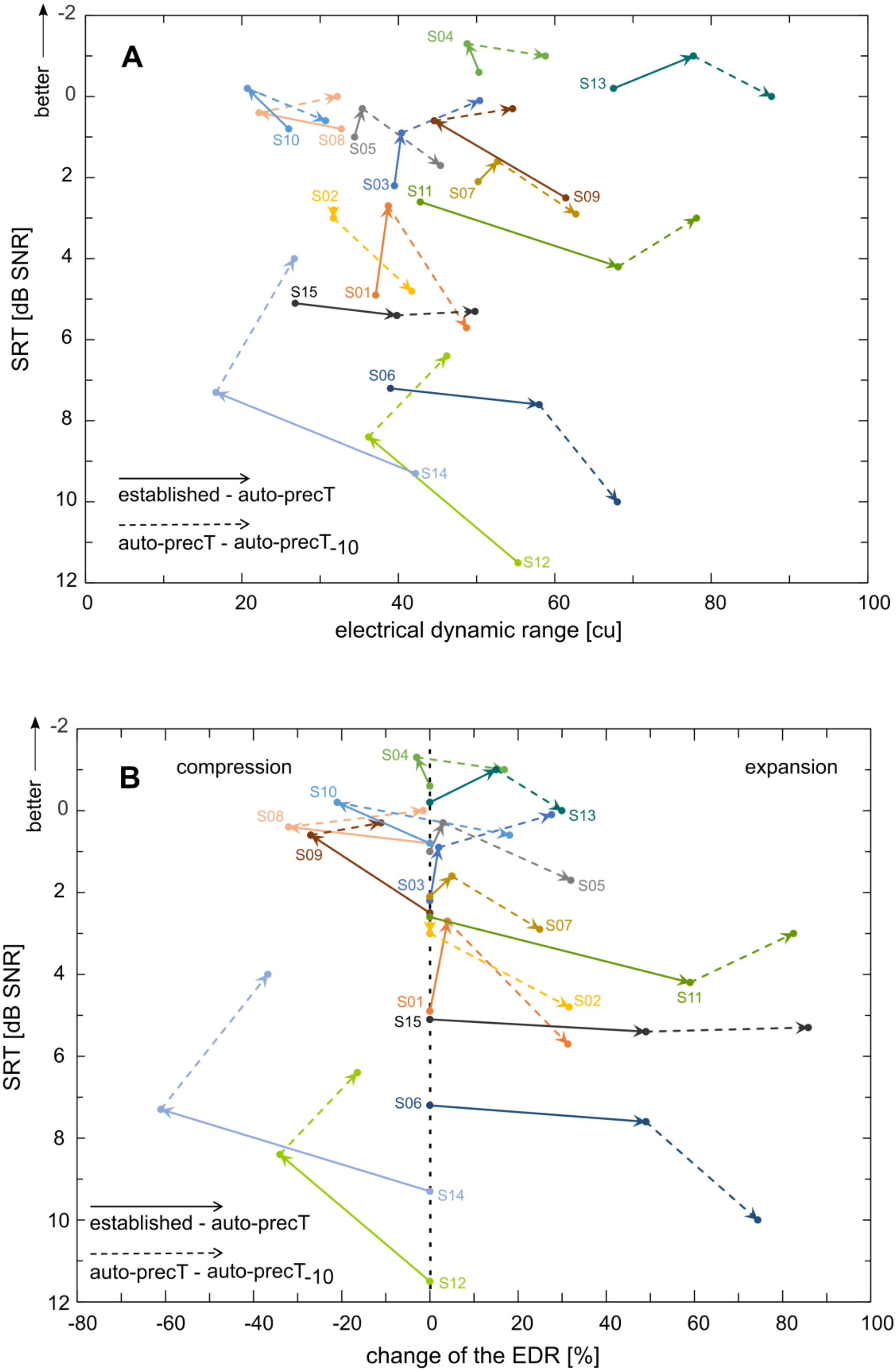


Figure 34. Intra-subject changes of speech reception thresholds (SRT) with the three different map conditions. The different colors represent the subjects. Full arrows lead from the SRT's measured with the established T-levels to those with the T-levels set to the

threshold values determined with the auto-precT application, dashed arrows lead from the SRT's measured with the auto-precT condition to those measured with T-levels set 10 cu below the threshold values determined with the auto-precT application. Figure A shows the absolute values of the EDR's (average EDR's across all electrodes), Figure B the changes of the EDR relative to the established EDR. The change of the EDR's from the established settings to the auto-precT was based on the precise determination of threshold values for every single electrode using the auto-precT application. The EDR's of the auto-precT-10 condition resulted from global lowering of the T-levels of the auto-precT condition.

As indicated before, there is a broad spectrum of the values for the EDR across subjects. In line with findings of the recent study from Kim et al. (2018) there seems to be no strong correlation between the size of the electrical dynamic range and speech perception. This supports the assumption that there is no such thing as a 'standard dynamic range' that is best for all patients and reflects the individuality of the patients' hearing anatomy and physiology. The optimal individual dynamic range depends on the functionality of the auditory nerves. Consequently, average values for the dynamic range might be helpful as a default setting for the initial fitting, but in the CI-hearing rehabilitation process individual measurements of threshold levels should be performed.

Subgroup analysis: The impact of the three different map conditions on the speech perception, shown in Figure 34, was interpreted for the individual subjects. The results suggested, that the subjects could be divided into the following subgroups, based on their results of the auto-precT application: First, subjects whose established EDR was on average compressed by 20 percent or more when T-levels were set to the hearing thresholds determined using the auto-precT application (n = 5); second, subjects who had their EDR changed by less than 20 percent on average, compressed as well as expanded (n = 7); and third, subjects whose established EDR was averagely expanded by more than 40 percent (n = 3). An overview of the subgroups and their characteristics is shown in Table 4. The subjects who had the most significant improvement in speech perception (subgroup average impact of auto-precT: improvement by -1.7 dB SNR) were those with a strong compression of the EDR, who had their T-levels notably raised relatively to their established EDR respectively. There are several conceivable reasons for the fact that the threshold levels they determined were a lot higher than their established T-levels. Possibly, their T-levels were not adjusted well and had been set too low in the clinical fitting so far. Another reason for the estimation of higher, and probably more precise, threshold levels could be the decreased likelihood of phantom hearing when using the auto-precT application instead of the common clinical fitting approach. Contrary to the common clinical fitting procedure, where only one stimuli is presented before the subject responses, the auto-precT application presents two sequential stimuli and then the subject gives a response of how many stimuli he or she perceived (two, one or none). For some subjects this way of stimuli presentation may decrease the probability of phantom hearing and thus lead to determining higher thresholds values than with the common fitting procedure. Presumably, phantom hearing is more likely to occur if the stimuli are continuously lowered or raised, which is what is done in the common fitting approach. In counteract this, the order of the two sequential stimuli is randomized in the auto-precT application, so sometimes the higher stimulus is presented first and sometimes the lower one. When analyzing the impact of global lowering of the T-levels in this subgroup, it is interesting to see that opposed to the other subgroups, speech perception was improved even more (subgroup average impact of lowering T-levels: improvement by -1 dB SNR). That has been especially the case for the subjects S12 and S14. This observation leads to the assumption that the subjects might have overestimated their threshold values with the auto-precT application. A possible reason for this could be that they have difficulties in perceiving soft sounds in general. The largest subgroup had only a mild change of the EDR (-3 to +15 % compared to the established one), but nevertheless clearly benefited from the auto-precT application and showed an average improvement of speech perception of -0.9 dB SNR (compared to the score with the established settings). This confirms the hypothesis that the precise determination of hearing thresholds is a worthwhile procedure, even if it is ‘only’ fine-tuning, particularly for optimizing speech perception at low levels. For all but one subject in this subgroup the impact of globally lowering T-levels by 10 cu was clearly negative (average deterioration by 1.1 dB SNR). The smallest subgroup constituted the subjects who had a strong expansion (averagely +52%) of the EDR as a result of the auto-precT application. Those three subjects were the ones who did not benefit from the auto-precT application. Their speech perception was deteriorated by an average of 0.8 dB SNR.

Table 4. Subgroup analysis of the study results based on changes in the EDR.

	strong compression	**fine-tuning**	**strong expansion**
rel. change of average EDR *(auto-precT :: established)*	< -20 % averagely -35 %	within +/- 20 % averagely +4 %	> +40 % averagely +52 %
subjects	S08, S09, S10, S12, S14	S01, S02, S03, S04, S05, S07, S13	S06, S11, S15
average change of SRT's using auto-precT *(auto-precT :: established)*	improvement by 1.7 dB SNR	improvement by 0.9 dB SNR	deterioration by 0.8 dB SNR
impact of global expanding 'auto-precT EDR' by 10 cu *(auto-precT :: auto-precT$_{-10}$)*	improvement by 1 dB SNR	deterioration by 1.1 dB SNR	deterioration by 0.4 dB SNR

Subgroup division was based on the changes of the EDR's relative to the established ones. The changes resulted from setting the T-levels to the threshold values determined with the auto-precT application.

6.3. Other methods for determining threshold values

A number of adaptive and adjustment procedures for determining threshold values were compared by van Wieringen and Wouters (2001). The auto-precT application proposed in this paper is in some way a combination of the procedures van Wieringen and Vouters were evaluating. In one adaptive procedure they assessed, two to five stimuli - of the same duration, frequency and loudness - were presented and the subjects had to 'count the pulses' they perceived. In the other adaptive procedure, four buttons were shown to the subjects on a screen, which represented four time intervals. One button after another was highlighted and at some point the stimulus was presented. The subjects had to choose the button that corresponded to the time interval during which they heard a sound. In contrast to the auto-precT application, those are tasks with a correct/false answer whereas in the auto-precT application there is no correct of false answer. In that way the auto-precT application resembles an adjustment procedure, where the subjects state how loud they perceived the stimulus. This kind of adjustment procedure was also evaluated by van

Wieringen and Vouters. Patients had to rank the stimulus loudness from '---' to '+++'. The results of their study showed that the 'count the pulses' procedure led to a more precise threshold determination, than the 'choose the interval' or the adjustment procedure. However, both adaptive procedures took significantly more time than the adjustment procedure. The proposed auto-precT application seems to be a good balance between precision as well as time.

Another study in which different ways of obtaining hearing thresholds are compared was published by Skinner et al. (1995). In the first adaptive procedure, stimuli were presented using a keyboard. The first stimulus was above the hearing threshold. Then the stimulus level was decreased by 10 level steps until the subject did not perceive a stimulus anymore. After that the stimulus level was increased by 5 level steps until the stimulus was heard again. This sequence was repeated with 4 level steps down and 2 level steps up. The whole procedure was done four times. The second procedure was done using a knob. After the presentation of pulse trains well above the hearing threshold, the stimulation level was lowered clearly below the hearing threshold. It was then increased turning the knob in steps of 1 to 3 levels until the stimulus was perceived. This procedure was also repeated three more times. A common problem with such an approach is that patients with tinnitus find it hard to distinguish between the tinnitus and the stimulus which is approximating the hearing threshold from below. As to be expected, the hearing thresholds determined with the keyboard procedure were higher than those determined with the knob procedure. An important difference of the auto-precT application is that in auto-precT two stimuli with different levels (with the current step size in between) are presented automatically immediately after one another. Close to the hearing threshold one stimulus is above the threshold and one below. It is suggested that this makes it easier for patients, especially those with tinnitus, to precisely determine their hearing thresholds.

Mewes and Hey (2017) used a common approach for determining electrical hearing thresholds. They approached the threshold from above with a step size of 4 cu using the Nucleus clinical fitting software. When the patients did not perceive the stimulus anymore the stimulus was raised again by 4 cu until they heard it again. The same procedure was repeated with a step size of 2 cu. Again, the main difference to our study is that only one stimulus was presented at a time, whereas in the auto-precT application two stimuli are presented sequentially. Moreover, while the behaviorally measurements were performed by an audiologist in the study from the Mewes and Hey, the auto-precT application was performed by patients themselves.

6.4. Applicability

All subjects, aged from 20 to 71 years, were able to perform the MatLab based auto-precT application on a touchscreen and thereby determined their electrical hearing threshold levels completely by themselves. No clinical fitting software or help from an audiologist was needed to run the program. The subjects' feedback to the application was entirely positive – they stated that even though the task of threshold determination requires a lot of concentration, the application is 'very intuitive', 'comfortable to use' and 'easy to understand'. The 11 subjects who took only smaller breaks (overall < 5 min) needed an average time of 39 minutes to run the program, 107 seconds per electrode respectively. 125 seconds per electrode were needed for the manually executed precT procedure in the previous study from Rader et al. (2018) (Figure 35). In the study of van Wieringen and Wouters (2001) the time needed to determine the T-levels with adaptive procedures ranged from 177 to 363 seconds per electrode. Compared to these findings, the method proposed in this study is much faster.

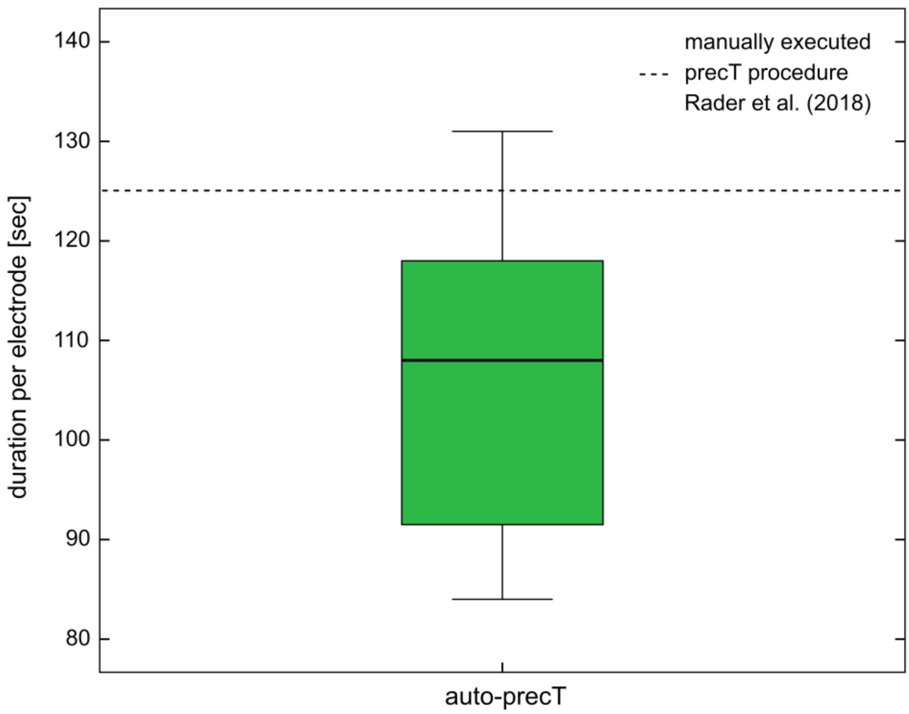


Figure 35. Time needed per electrode for threshold determination with the auto-precT application. The box plot contains median, 1st and 3rd quartiles, minimum and maximum values and the median time needed per electrode with the manually executed precT procedure as described by Rader et al. (2018).

The auto-precT application is a good combination of both, precision and time-efficiency which are important criteria for the clinical application. Another great advantage of auto-precT is that it is self-explanatory and can be run by the patient himself. No help from an audiologist is needed while the patient is completing the program, so personnel time is saved as well. That opens up great opportunities. Consequently, the software used in this study should be refined, so that it can be run on handheld devices and smart phones.

7. Conclusion

A psychoacoustic application was developed in order to allow patients to precisely and independently determine their electrical hearing thresholds, without an attending audiologist. The applicability of the application was confirmed in a clinical study. Subjects benefited from adjusting the T-levels to the threshold levels determined with the auto-precT application, resulting in a median improvement in speech perception in noise of -0.9 dB SNR. The auto-precT application is a useful tool for the precise determination of hearing thresholds. Thus, not only speech perception at low levels is improved with the auto-precT application, but also clinical resources are spared and the workflow is optimized. Consequently, we recommend the integration of auto-precT in the clinical fitting as well as in a remote fitting software. Furthermore, future possibilities of auto-precT include the implementation as an app on tablets or smart phones.

Zusammenfassung auf Deutsch

Bei der Optimierung des Sprachverstehens von Cochlea Implantat Patienten spielt die Anpassung des Signalprozessors eine elementare Rolle. Die Anpassung, das sogenannte Fitting, erfordert viel Zeit und Konzentration - sowohl von Patienten als auch von Audiologen. Ebenso aufwendig wie relevant ist dabei die genaue Bestimmung der elektrischen Hörschwellen. In dieser Arbeit wird eine neu entwickelte Software-Anwendung vorgestellt, mit der CI-Patienten ihre elektrischen Hörschwellen präzise und eigenständig, ohne die Hilfe eines Audiologen, bestimmen können. Die Anwendbarkeit des neuen Verfahrens und der Nutzen für das Sprachverstehen wurden in einer klinischen Studie untersucht.

Für die präzise und eigenständige Hörschwellenbestimmung wurde eine neue Anwendung (*auto-precT*) entwickelt. Dazu wurde ein adaptives psychoakustisches Verfahren, das auf einer „3-alternative-forced-choice (3-AFC)“-Methode basiert, in MatLab implementiert und eine grafische Benutzeroberfläche erstellt. Die Tonsignale wurden mit einem CIC4 Implantat-Decoder (DIET) kalibriert.

Zur Evaluation des Verfahrens wurde eine prospektive Studie mit 15 erfahrenen CI-Patienten durchgeführt. Die elektrischen Hörschwellen wurden zunächst mit der *auto-precT* Anwendung bestimmt. Anschließend wurden drei verschiedene Programme eingestellt: (P1) Die T-level, die bei den Studienteilnehmern zuvor eingestellt waren; (P2) Die von Probanden mit der *auto-precT* Anwendung bestimmten elektrischen Hörschwellen als T-level; (P3) Die aus (P2) um 10 cu erniedrigten T-level. Mit allen drei Programmen wurde das Sprachverstehen im Störgeräusch bei einem Sprachpegel von 50 dB SPL gemessen.

Alle Studienteilnehmer konnten die *auto-precT* Anwendung über einen Touchscreen bedienen und eigenständig, ohne anwesenden Audiologen, ihre elektrischen Hörschwellen bestimmen. Die T-level (P2) wurden durchschnittlich um einen Absolutwert von 10,5 cu im Vergleich zur Voreinstellung (P1) verändert. Es zeigte sich eine deutliche Verbesserung des Sprachverstehens im Störgeräusch bei leisen Pegeln - die mittlere Sprachverständnisschwelle besserte sich signifikant von 2,5 dB SNR (P1) auf 1,6 dB SNR (P2) ($p = 0,02$). Mit den global erniedrigten T-leveln (P3) war die Sprachverständnisschwelle am höchsten (Median: 2,9 dB SNR, nicht signifikant im Vergleich zu P1 und P2).

Die klinische Studie zeigte, dass die *auto-precT* Anwendung machbar ist und einen Nutzen für das Sprachverstehen bei leiser Sprache im Störgeräusch hat. Die Studienteilnehmer konnten eigenständig ihre elektrischen Hörschwellen bestimmen und profitierten von der Anpassung der T-level auf diese Schwellenwerte. Zur Durchführung der *auto-precT* Anwendung war keine Anwesenheit eines Audiologen nötig. Somit lässt sich mit *auto-precT* sowohl das Sprachverstehen als auch der alltägliche klinische Arbeitsablauf verbessern. Daher wird die Integration der Anwendung in die klinische Fitting- und die Remote-Fitting-Software empfohlen. Darüber hinaus ist die Implementierung der *auto-precT* Anwendung als App für Smartphones und Tablets ein vielversprechender Ansatz für die weitere Optimierung des Anpassungsprozesses von Cochlea-Implantaten.

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Figure references

Figure 1: Overview of the anatomy of the ear – Adapted from:
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Figure 2: Ossicles – Adapted from: .
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Figure 3: Sound wave transmisson in the cochlea – Adapted from:
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Figure 4: Tonotopy:
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Figure 6: The internal and external components of a cochlear implant
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