Technische Universität München
Fakultät für Elektrotechnik und Informationstechnik

Effects of static interaural time offsets on binaural hearing

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Vollständiger Abdruck der von der Fakultät für Elektrotechnik und Informationstechnik der Technischen Universität München zur Erlangung eines

Doktors der Ingenieurwissenschaften (Dr.-Ing.)

genehmigten Dissertation.

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Prüfer*innen der Dissertation:

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Die Dissertation wurde am 13.04.2022 bei der Technischen Universität München eingereicht und durch die Fakultät für Elektrotechnik und Informationstechnik am 03.06.2022 angenommen.

Abstract

When people with hearing loss are provided with different devices in each ear, these devices usually have different processing latencies. This leads to static temporal offsets between both ears in the order of several milliseconds. Binaural mechanisms, such as sound localization or speech understanding in noise rely on precise timing between both ears. Therefore, such an offset in timing is hypothesized to impair binaural listening. In this thesis, the effects of offsets in stimulation timing on sound localization accuracy and speech understanding in noise were measured in normal-hearing and hearing-impaired listeners. In bimodal listeners utilizing a cochlear implant on one ear and a hearing aid contralaterally, the offset in timing between both ears was reduced and the effects on sound localization and speech understanding were measured. In normal-hearing listeners an artificial offset between both ears was created by delaying the signals on one ear by several milliseconds. The detrimental effect of such an offset on speech understanding in noise was then measured and validated using a phenomenological model. The results show that an offset in stimulation timing significantly impedes mechanisms of binaural hearing in both listener groups. When the offset in bimodal listeners was compensated, sound localization improved instantaneously.

Kurzfassung

Wenn Menschen mit Hörverlust auf beiden Ohren mit unterschiedlichen Geräten versorgt werden, weisen diese Geräte oft unterschiedliche Prozessierungslatenzen auf. Dies führt zu statischen zeitlichen Versätzen zwischen beiden Ohren in der Größenordnung mehrerer Millisekunden. Für binaurale Mechanismen wie Schalllokalisation oder Sprachverstehen im Störgeräusch ist präzises Timing zwischen beiden Ohren notwendig. Daher wird angenommen, dass ein solcher Zeitversatz das binaurale Hören beeinträchtigt. In dieser Arbeit wurden die Effekte eines solchen zeitlichen Versatzes auf die Schalllokalisation und auf Sprachverstehen im Störgeräusch in normalhörenden und hörgeschädigten Personen untersucht. Bei bimodal versorgten Probanden, welche auf einem Ohr mit einem Cochlea-Implantat und auf dem gegenüberliegenden Ohr mit einem Hörgerät versorgt sind, wurde der Zeitversatz zwischen beiden Ohren reduziert und die Effekte auf Schalllokalisation und Sprachverstehen untersucht. In normalhörenden Probanden wurde ein künstlicher Zeitversatz zwischen beiden Ohren durch Verzögerung der Signale auf einem Ohr um mehrere Millisekunden erzeugt. Der nachteilige Effekt eines solchen Versatzes auf das Sprachverstehen wurde dann experimentell untersucht und mithilfe eines phänomenologischen Modells validiert. Die Ergebnisse zeigen, dass ein Zeitversatz zwischen beiden Ohren Mechanismen des binauralen Hörens in beiden Probandengruppen signifikant beeinträchtigt. Wenn dieser Zeitversatz bei bimodalen Probanden kompensiert wurde, verbesserte sich die Schalllokalisation instantan signifikant.

Acknowledgments

I would like this opportunity to thank those who helped me during the creation of this thesis. This were foremost my two supervisors Prof. Werner Hemmert and Prof. Stefan Zirn. Without their continuous support, knowledge and experience, this work would not have been possible. I further want to thank my mentor, Dr. Thomas Wesarg, whose experience I could always rely on, professionally and personally. Thanks also to all my current and previous colleagues at the Peter Osypka Institute for Medical Engineering and the group for Bio-Inspired Information Processing in Munich. It was a pleasure to work in a group with all of you.
I want to thank MED-EL for their financial and material support and their expertise throughout this thesis.
I want to express my deepest gratitude to my subjects that volunteered to participate in the experiments described in this thesis. Without their help and patience, none of this would have been possible.
At last, I want to thank my family and my friends for always believing in me and for their support especially in the final weeks of writing this thesis.

Freiburg, 09. April 2022

Contents

Glossary

ABR ... *auditory brainstem responses*
AD ... *analog-to-digital*
ADC ... *analog-to-digital converter*
BTE ... *behind-the-ear*
CI ... *cochlear implant*
DA ... *digital-analog*
DAC ... *digital-to-analog converter*
DL ... *delay line*
eABR ... *electrically-evoked auditory brainstem responses*
EC ... *equalization-cancellation*
ERB ... *equivalent rectangular bandwidth*
FFT ... *fast Fourier transform*
HA ... *hearing aid*
HATS ... *head and torso simulator*
HL ... *hearing level*
HRIR ... *head related impulse response*
HRTF ... *head related transfer function*
ILD ... *interaural level difference*
ITD ... *interaural time difference*
JND ... *just noticable difference*
OlSa ... *Oldenburg sentence test*
RIC ... *reciever in the canal*
RMS ... *root-mean-square*
RMSE ... *root-mean-square error*
SII ... *speech intelligibility index*
SNR ... *signal-to-noise ratio*
SPL ... *sound pressure level*
SRM ... *spatial release from masking*
SRMR ... *speech-to-reverberation modulation energy ratio*
SRT ... *Speech reception threshold*
SSD ... *single-sided deafness*

1 Introduction

One important advantage of hearing with two ears (binaural hearing) is the possibility to localize sounds in space. This ability of the auditory system is extremely helpful and important in everyday life. From localizing where a car comes from when crossing a road or localizing sounds in situations where visual localization is impaired e.g., in dim lit situations, sound localization helps to characterize one's surroundings. Further, in conversations with multiple speakers humans benefit from the ability to localize sources of sound precisely, facilitating the ability to pick out a speaker of interest among other interfering speakers (Cherry, 1953). While normal-hearing subjects can localize sounds precisely, hearing-impaired listeners often have trouble with this task (Noble, Byrne, & Lepage, 1994; Noble, Byrne, & Ter-Horst, 1997). These problems often persist, even when hearing impairment is treated technically with the help of hearing aids (HA) or cochlear implants (CI). Generally, these problems can have their roots in the poor transmission of binaural cues by the devices used to treat hearing impairment. However, there are further factors possibly impeding sound localization in these subjects, even when binaural cues are transmitted via the HA or the CI. Poor localization performance is especially prominent in subjects where hearing loss is asymmetrically treated, i.e., with different devices on both ears as in bimodal listeners using a CI in one ear and a HA in the other (Dorman, Loiselle, Cook, Yost, & Gifford, 2016). This thesis aims to investigate one prominent effect that occurs in such asymmetric treatment. This effect is an offset in stimulation timing in both ears due to different signal processing latencies in both devices. Since binaural processing is believed to rely on precise temporal coding, offsets in stimulation that are non-physiological in nature are believed to impact this processing. The aim of this thesis is to investigate the effects of interaural temporal offsets in stimulation timing on binaural hearing. This was investigated in two different ways for two subject groups. In bimodal subjects, provided with CIs and HAs, the effect of the temporal offset imposed by the different processing latencies of the devices on sound localization and spatial release from masking (SRM) were investigated experimentally. In the next step, this device delay mismatch was reduced and sound localization and SRM were measured again. For comparison, in normal-hearing subjects, a temporal offset was artificially introduced and detrimental effects on SRM were investigated. SRM as a feat of binaural hearing can be simulated using phenomenological models of binaural processing. However, an influence of static interaural offsets in timing have not investigated with such models. In this thesis such a binaural model, based on an equalization-cancellation mechanism as proposed by Durlach (1963), was utilized. With this it was possible to verify, whether such a model can accurately describe the experimental findings. Further, the experimental data could be extended after verification of the model.

1.1 Structure and Overview

This thesis is structured in three main chapters (chapter 3, 4 and 5). Each chapter, while building on the previous chapters, can be read independently as each contains a specific introduction and discussion, as they are derived from publications (Angermeier, Hemmert, & Zirn, 2021, 2022).

Chapter 2 briefly introduces the most important terminology and metrics used in this thesis. It further introduces the fundamental principles of sound localization relevant to this thesis, as well as the problems arising in sound localization in hearing-impaired subjects to the unfamiliar reader with more detailed introductions being given in the corresponding chapters.

Chapter 3 presents research results for sound source localization accuracy in bimodal listeners utilizing a CI in one ear and a HA contralaterally. The listeners' sound source localization accuracy was compared when the temporal offset in stimulation timing between CI and HA is reduced acutely and after a brief familiarization period. It is shown that bimodal listeners perform significantly better in sound localization tasks, when the temporal mismatch between both devices was reduced. Further, no effects of short-term familiarization on sound source localization accuracy were found leading to the hypothesis that this effect is acute in nature.

Chapter 4 investigates the influence of static temporal offsets in interaural stimulation timing on speech understanding in noise, especially spatial release from masking (SRM) in normal-hearing subjects. To deepen the understanding how such a static offset influences the processing of binaural cues used for spatial unmasking, all experiments were also conducted with the binaural cues isolated. Further, an equalization-cancellation (EC) model for binaural unmasking was utilized to investigate if the behavior of widely used binaural models matched the experimental results in this novel condition. After verification of the model results with experimental data, the model was used in a more realistic scenario with reverberation to extend the experimental findings.

In chapter 5, effects of familiarization to synchronized stimulation timing in bimodal subjects on sound localization accuracy and speech understanding was investigated. Similar methods as in chapter 3 were used with the difference that the delay of the CI stimulation is done via the CI speech processor, since MED-EL included a delay of the CI stimulation in their fitting software in 2020, inspired by the findings of Zirn et al. (2019). This allowed to apply longer familiarization periods. Key findings were that the positive effects of device synchronization were acute in nature and did not further improve after 3-4 weeks of familiarization.
Further findings suggest that bimodal subjects can familiarize to desynchronization of stimulation timing to some extent. In speech understanding, no significant effect of the devices delay settings was found, which is well in line with the results reported in chapter 4.

2 Fundamentals

This chapter aims to briefly explain the fundamentals of sound localization and the binaural cues used by humans to localize sounds. Further, it seeks to give a brief introduction to the technologies used to treat hearing loss, how users of these technologies can utilize binaural cues and finally to the problems arising when these technologies differ on both ears in terms of processing latencies.

2.1 Terminology and metrics

2.1.1 Coordinate system

A position from which a sound originates in a listener centric spherical coordinate system can be described by its azimuth, elevation, and distance from the listener. In this thesis such a coordinate system is used to describe sound positions with 0° azimuth corresponding to the position in front of the listener. In the horizontal plane the azimuthal angle is defined in a clockwise coordinate system in this thesis. This means positive azimuthal angles correspond to source positions oriented to the right side of the listeners head while negative azimuthal angles correspond to source positions oriented to the left side of the listeners' head. While sound localization is usually performed in three-dimensional space, the investigations of this thesis focused on sound localization in the horizontal plane i.e., the plane set up by the interaural axis with an elevation of 0°.

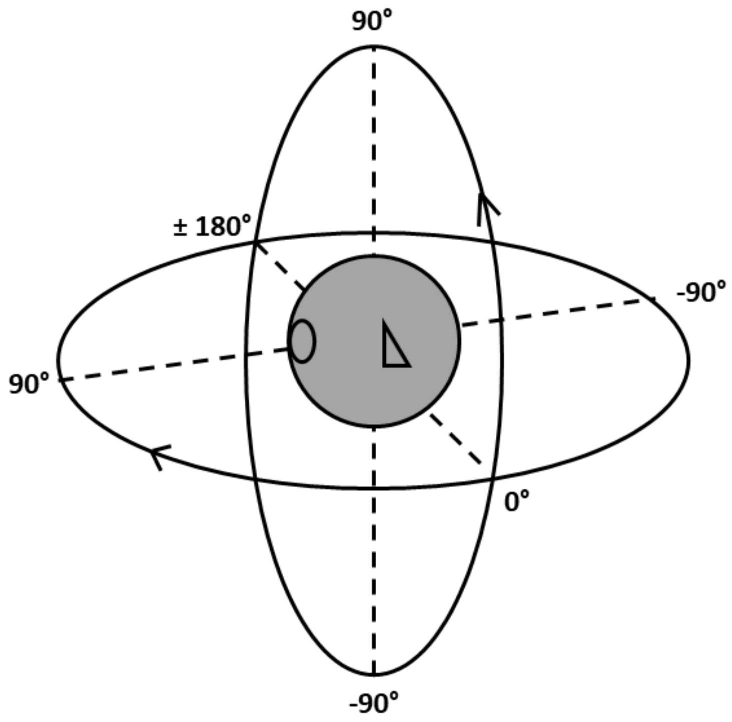

Figure 2.1: Head-centered spherical coordinate system with clockwise azimuthal angles used in this thesis.

2.1.2 Accuracy metric

Throughout this thesis the term "accuracy" is used to describe subject responses. This refers to the definition of the term given by the International Organization for Standardization (ISO) in ISO 5725-1 (International Organization for Standardization, 1994). In this definition, accuracy is comprised of two components: "precision" and "trueness". Trueness refers to the distance between the mean of the subjects' answers and the actual correct answer. Precision refers to the distance between the subjects' answers. So, if a subject's responses have a high spread but are all around the correct answer, trueness is high, but precision is low. If the distribution of responses is rather narrow but there is an offset between the mean response and the true answer, precision is high, but trueness is low. This concept, and how trueness and precision interact in terms of accuracy is shown in Figure 2.2.

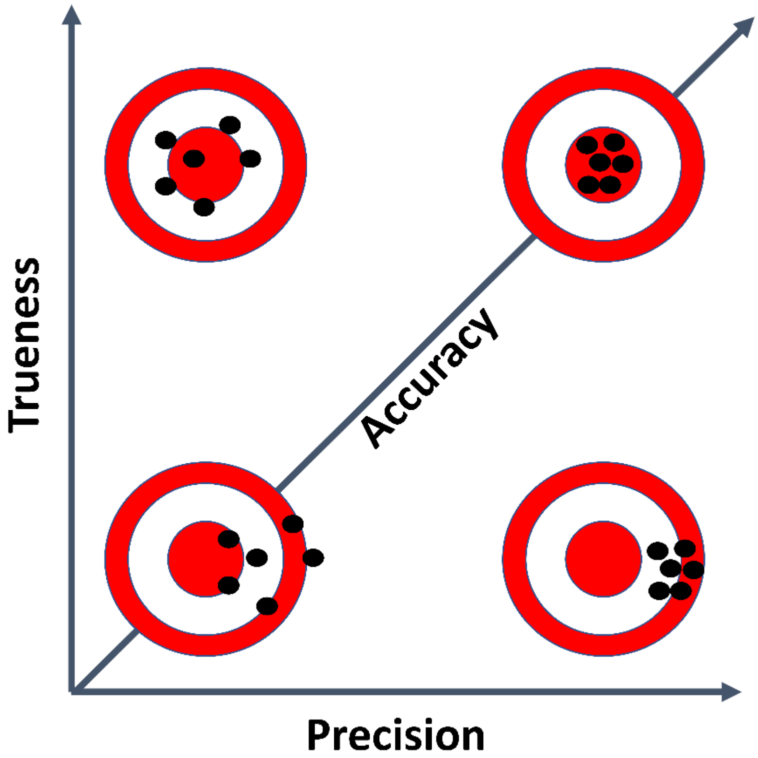

Figure 2.2: Accuracy concept comprised of trueness and precision as described in ISO 5275-1.

2.1.3 Terminology

In this thesis, two different terms are used to describe the static interaural offset in stimulation timing investigated. This is due to the different subject groups investigated, bimodal CI/HA users and normal-hearing listeners and due to different reasons for the offset.

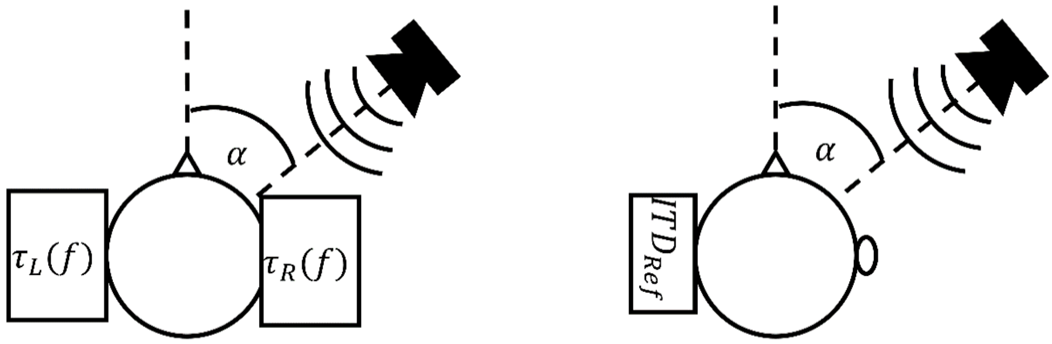

Figure 2.3: Difference between a device delay mismatch (left) and a reference ITD (right).

In bimodal subjects, the mismatch is described as the "device delay mismatch" which can be calculated as the latency of the slower modality, in this thesis this is usually the HA, minus the latency of the faster modality (the CI). This mismatch can be frequency dependent. The specific case of a device delay mismatch in bimodal CI/HA users is discussed in 2.5.1. The influence on interaural time differences (ITDs) on the left ear relative to the right ear (ITD_L)with a device delay mismatch can be calculated as follows:

$$ITD_L(\alpha) = ITD_{L,Nat}(\alpha) + \left(\tau_L(f) - \tau_R(f)\right) \tag{2.1}$$

With α being the angle from which the signal originates and $\tau_L(f)$ and $\tau_R(f)$ being the frequency specific delays of the left and right device. $ITD_{L,Nat}(\alpha)$ describes the naturally occurring ITD on the left ear at the angle α (compare Figure 2.3 left).

In normal-hearing subjects, the systematic delay of signals on one ear is called reference ITD in this thesis (compare Figure 2.3 right). This reference ITD is the ITD for sounds originating directly in front of the subjects, and would normally be roughly 0 µs, for a perfectly symmetrical head shape. If a constant, frequency independent delay is added to one ear of the subject, this reference ITD is changed. This terminology was chosen with respect to the literature, where the term reference ITD is also used to describe constant ITD offsets in normal-hearing listeners (Bernstein, Stakhovskaya, Schuchman, Jensen, & Goupell, 2018; Koehnke, Culotta, Hawley, & Colburn, 1995; Mossop & Culling, 1998). The influence of a reference ITD on the magnitude of ITD on the left ear relative to the right ear can be calculated as follows:

$$ITD_L(\alpha) = ITD_{L,Nat} + (\alpha)ITD_{ref} \tag{2.2}$$

With ITD_{ref} being the frequency independent reference ITD on one ear.

2.2 Cues for sound localization in the horizontal plane

For sounds originating from source positions in the horizontal plane other than directly in front or behind of the listener (i.e., 0° & 180° azimuth) distinct differences in level and timing of the signals arriving at both ears arise, which the auditory system utilizes for sound localization (Rayleigh, 1907). These interaural differences will be introduced in the following. For a more comprehensive overview of sound localization with respect to elevation and distance of sound sources see Blauert (1997).

2.2.1 Interaural time differences

Interaural time differences (ITDs) are the differences in arrival time of a sound at both ears, arising from the difference in distance of both ears towards the sound source (see Figure 2.4). These ITDs range between 0 and around 700 µs in adult humans. The smallest ITDs that normal-hearing humans can perceive are around 18 µs for listeners that are untrained to the task of perceiving minimal ITDs and around 7 µs for trained listeners (Thavam & Dietz, 2019). Up to around 1.3 kHz, ITDs in the temporal fine structure of the signal can be utilized. Above 1.3 kHz, ITD sensitivity to the temporal fine structure decreases rapidly (Brughera, Dunai, & Hartmann, 2013). At these higher frequencies, the ITD of the fine structure is discarded, but still the signal's low frequency envelope can be utilized for sound localization (Blauert, 1997).

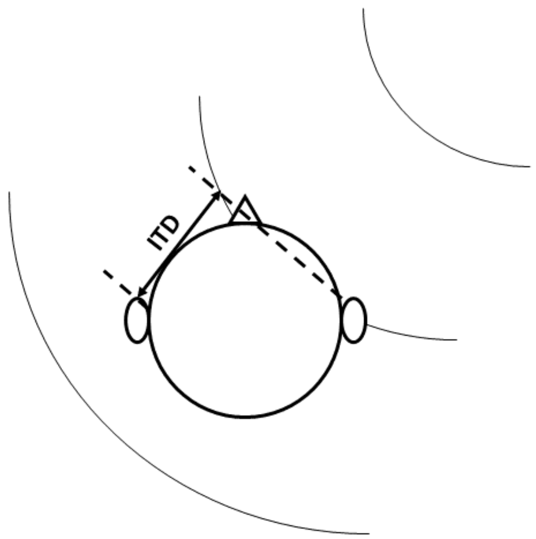

Figure 2.4: Schematic depiction of interaural time differences (ITDs) for low frequency sounds resulting from differences in distance of both ears to the sound source.

2.2.2 Interaural level differences

Interaural level differences (ILDs) are the frequency dependent difference in sound pressure level reaching the listeners eardrums between both ears when sounds originate from horizontal off-center positions. This phenomenon arises from high-frequency sounds being attenuated by the head of the listener. For wavelengths of incoming sounds being smaller than the listeners head, an acoustic shadow is cast by the head leading to different sound pressure levels at both eardrums (see Figure 2.5). ILDs can be as big as 20 dB depending on the frequency and position of the sound (Shaw, 1974). Humans are capable of resolving minimal ILDs of 1,5 dB for broadband signals (Babkoff & Sutton, 1969; Hall, 1964; Von Békésy, 1930). Below ca. 1000 Hz, ILDs do not arise naturally, since the wavelengths of the sound waves are bigger than the standard head diameter in adult humans.

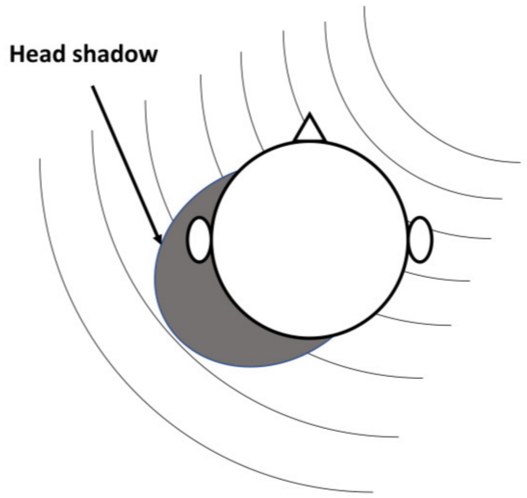

Figure 2.5: Head shadow for wavelengths below the heads diameter creating interaural level differences (ILDs) in the ear opposite to the sound source.

2.3 Cochlear Implants

In humans suffering from severe sensorineural hearing loss or deafness with a functional cochlear nerve, CIs are a neuroprosthetic device used to restore hearing to some extent. CIs are used to stimulate the auditory nerve fibers directly via current pulses, replacing the function of the damaged hair cells of the cochlea, when hearing aids do not provide sufficient excitation.

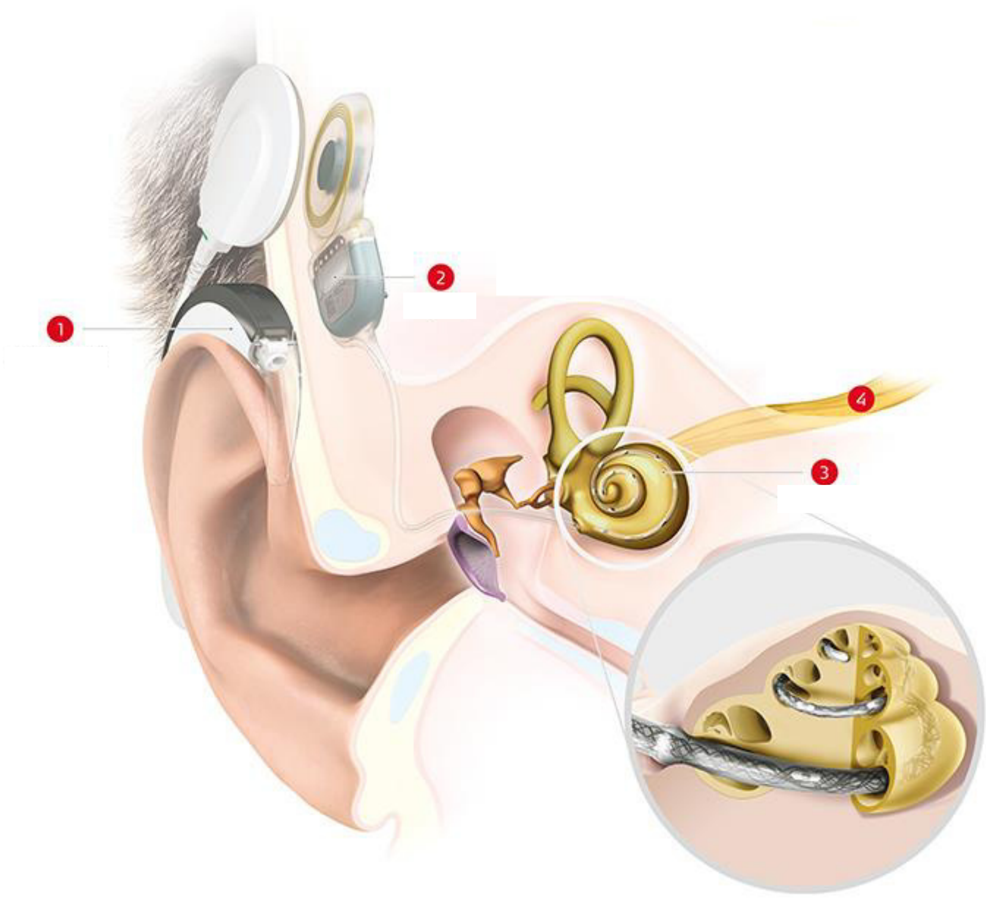

Figure 2.6: Cochlear Implant (CI) consisting of the speech processor worn behind the ear (1) and the implant (2) with the intracochlear electrode (3) and the auditory nerve depicted (4). Picture with permission from MED-EL.

2.3.1 Function of cochlear implants

A CI consists of two main components: the speech processor and the implant itself. The speech processor is usually worn behind the ear much like a conventional HA. It houses microphones to pick up incoming acoustic signals and to enable directionality if more than one microphone is used. Within the speech processor, the signals are converted via an analog-to-digital converter (ADC). Signals are then split into different frequency channels via fast Fourier transformation (FFT) or bandpass filtering. Following, the signals are processed into current pulses according to the manufacturers coding strategy to later stimulate the auditory nerve fibers. The speech processor also contains the power source of the CI in the form of batteries or rechargeable accumulators. The speech processor and the implant are connected via magnets in the transmission coil of the speech processor and the implant. The signal and power transfer between the speech processor and the implant is done via induction. The implant itself is placed subcutaneously in the skull of the patient. From this implant, an electrode array is inserted via the round window into the scala tympani and thus into the cochlea. This electrode array has 12 to 22 electrode contacts, varying between manufacturers. Via these electrode contacts the auditory nerve can be stimulated directly with biphasic or triphasic current pulses. Due to the tonotopic nature of the cochlea and of the auditory nerve, different acoustic frequencies can be mapped to different electrode contacts, allowing for frequency specific stimulation. Electrode contacts at the base of the cochlea are used to stimulate portions of the auditory nerve encoding higher frequencies and apical electrode contacts stimulating lower frequency regions. This allows users of a cochlear implant to gain a surprisingly high degree of speech understanding despite the limited frequency resolution, which is limited by the spread of current within the cochlear. This current spread reduces the number of utilizable frequency channels due to channel interaction (Friesen, Shannon, Baskent, & Wang, 2001; Shannon, 1983)

2.3.2 Binaural cues in cochlear implants

For simplicity in this subsection, the binaural cues available for bilaterally implanted cochlear implant users are described to limit the reporting to the cues that are available with CIs in general.

CI processing preserves ILDs (Grantham, Ashmead, Ricketts, Haynes, & Labadie, 2008; Grantham, Ashmead, Ricketts, Labadie, & Haynes, 2007; Schoen, Mueller, Helms, & Nopp, 2005; van Hoesel & Tyler, 2003), still, ILDs can be compressed by the CI coding strategy (Dorman et al., 2014). The use of ITDs is limited as a localization cue as the temporal fine structure ITD is not sufficiently transmitted by the CI, since CIs extract the envelope of the incoming signal to modulate pulse trains with high rates of stimulation, usually around 1000 pulses per second (pps). In experimental stimulation setups, sensitivity to fine structure ITD in CI users has been shown when using lower stimulation rates (Thakkar, Kan, Jones, & Litovsky, 2018). With regular fixed pulse rates however, only ITD cues in the envelope of the signals are sufficiently transferred by the CI. The preservation of binaural cues and in the case of fine structure ITD, the lack thereof, is illustrated in Figure 2.2.7.

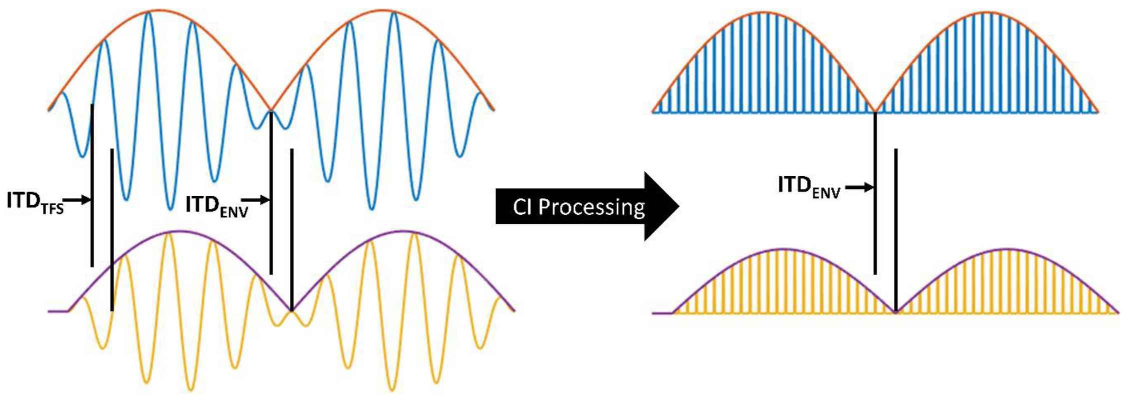

Figure 2.2.7: Influence of CI signal processing on ITD cues in the temporal fine structure (ITD_{TFS}) and the envelope of the signal (ITD_{ENV}) adapted from Laback et al. (2015).

Even with coding strategies which try to facilitate low frequency ITD cues, the benefits are limited compared to stimulation strategies with a fixed stimulation rate (Ausili et al., 2020; Fischer et al., 2021; Zirn, Arndt, Aschendorff, Laszig, & Wesarg, 2016). For a more comprehensive overview on ITD perception in bilateral CI provision see Laback et al. (2015). Thus, while ITDs can be used in theory by CI users, ILDs are the dominant and more reliable cue for sound source localization in these subjects with current CI coding strategies.

2.4 Hearing Aids

For patients with sensorineural hearing loss and some form of residual hearing, hearing aids (HA) are the standard treatment. There are different types of HAs but for the sake of brevity this section will only cover behind-the-ear (BTE) HAs.

2.4.1 Function of hearing aids

Digital HAs record incoming acoustic signals via one or more microphones. Signals are converted into digital signals via analog-to-digital (AD) conversion after analog preamplification and low pass filtering. These digital signals are then filtered into different frequency bands using either bandpass filters or an FFT. A variety of digital signal processing algorithms can then be performed such as noise reduction, amplitude compression, feedback cancellation and more sophisticated processing like automatic scene recognition. The most important processing step although is the frequency specific amplification of the signals to counter the individual hearing loss of the user. Signals are then converted back into sound waves via an receiver, that can either be housed within the HAs case connected to an earpiece via a sound tube or be placed in the ear canal (receiver in the canal - RIC) (Popelka, Moore, Fay, & Popper, 2016).

2.4.2 Binaural cues in hearing aids

Both ITDs and ILDs can be used in principle by hearing aid users with some limiting factors. For ITDs, the type of HA must be considered. If a subject uses a so-called open fitting, meaning an earpiece which simultaneously allows direct sound to enter the ear canal, both amplified, but through the processing delayed wavefronts and direct wavefronts superimpose in the ear canal. This can lead to distortion of timing information and thus distort ITDs (Denk, Ewert, & Kollmeier, 2019). When subjects experience hearing loss at higher acoustic frequencies, that cannot be countered by amplification, ILDs in these frequency regions are not accessible to these subjects. Further, ILD cues can be distorted by non-synchronized dynamic range compression between HAs in the case of bilateral HA use (Schwartz & Shinn-Cunningham, 2013; Wiggins & Seeber, 2011). So, although both localization cues are present in users of HAs, they are limited to some extent. For an comprehensive overview on the transmission of binaural cues in HAs see Denk et al. (2019).

2.5 Problems in asymmetric treatment of hearing loss

Hearing loss is often not symmetrical between both ears. Examples of this are unilateral hearing loss, with its most extreme form being single-sided deafness (SSD), where patients have normal hearing in one ear and severe hearing loss or deafness in the contralateral ear. Then there is asymmetrical hearing loss, which is defined as a difference of hearing level of at least 15 dB between both ears at three contiguous frequencies. Results of these asymmetries in hearing loss are asymmetries in treatment between both ears. This can range from unilateral HA or CI provision to bimodal provision of a HA in one ear and a CI in the contralateral ear. This special case of asymmetric provision was the focus of this thesis and will be introduced in the following.

2.5.1 Asymmetries arising in bimodal listening

Bimodal provision leads to several differences between both ears. These differences or mismatches arise in the domain of latency, stimulation level and stimulated frequencies. Since in this thesis only the difference in processing latencies was investigated, the other differences will not be discussed in this section. For an comprehensive overview see Pieper et al. (2021). Zirn et al. (2015) measured auditory brainstem responses (ABR) and electrically-evoked auditory brainstem responses (eABR) in bimodal CI/HA users and found a difference in ABR/eABR wave V latency of around 7 ms, meaning that the neural representation of sounds arriving at both ears has a static temporal offset in the order of several milliseconds. This offset arises due to vastly different processing paths in the ear treated with an CI and in the ear treated with the HA. Figure 2.8 from Zirn et al. (2015) illustrates these different processing paths for bimodal listeners.

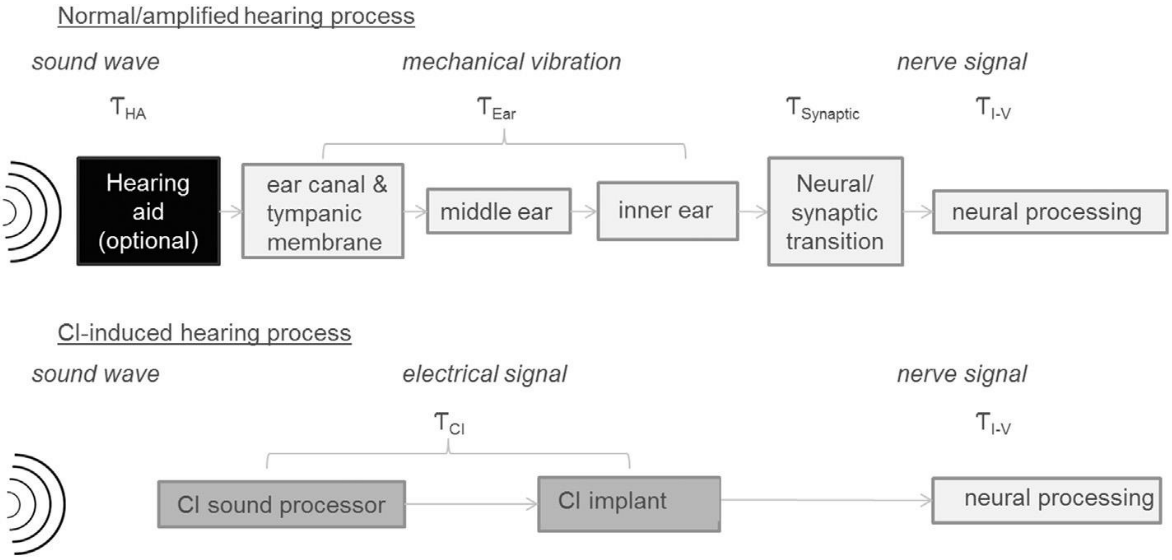

Figure 2.8: Different processing paths in bimodal listening with a hearing aid (HA) and cochlear implant (CI). Figure from Zirn et al. (2015).

On the ear provided with the HA, incoming sound is first processed by the HA with its latency component T_{HA}, afterwards the amplified sound travels through the ear canal to the tympanic membrane and via the middle ear to the inner ear, where a frequency dependent travelling wave delay is added (T_{Ear}) afterwards, the mechano-electrical transition is performed, converting mechanical vibration of the basilar membrane into neural excitation of auditory nerve fibers ($T_{Synaptic}$). Following this, the neural signals are processed in the brainstem.

On the ear provided with the CI, the latency of signals arriving at the level of the brainstem is only influenced by the latency of the CI speech processor and the implant (T_{CI}) and the delay between electrical activation of auditory nerve fibers and its processing on the level of the brainstem. Keeping in mind that the physiological range of ITD goes up to around 700 µs and the processing of ILD is hypothesized to be done within a temporal window of roughly 3 ms (Brown & Tollin, 2016), larger temporal offsets between both ears are expected to impact the usage of these binaural cues. In bimodal listeners, binaural cues are further limited in their availability in HAs (see2.4.2) and CIs (see 2.3.2) isolated and are dependent on sufficient spectral overlap in both ears.
Dorman and colleagues conducted comparative localization tests with normal-hearing-, hearing-impaired- and CI listeners, and found that bimodal listeners performed almost as bad as listeners only utilizing one ear for sound localization (Dorman et al., 2016). Subjects with symmetric devices (i.e., bilateral HAs or bilateral CIs) performed significantly better, although still worse than normal-hearing subjects. It must be noted that the bimodal subjects in this study had moderately severe to severe hearing loss at the frequencies 0.5, 1, 2, and 4 kHz in the ear provided with the HA, thus possibly not being representative of bimodal listeners with less pronounced hearing loss. In bilateral CI users, mismatches between both CIs can arise in the domain of level (Fitzgerald, Kan, & Goupell, 2015), and frequency (Kan & Litovsky, 2015) but not in latency in terms of a constant

offset compared to bimodal CI/HA users. This led to the hypothesis that the delay offset between both modalities could be a detrimental factor in sound localization accuracy in bimodal subjects.
Zirn et al. (2019) could report a highly significant improvement in sound source localization accuracy when the device delay mismatch was reduced. It is however unclear, how to optimally delay CI stimulation to reduce the detrimental effects of a device delay mismatch and how other binaural processes such as speech understanding in noise are affected by a device delay mismatch.

Fundamentals

3

Effects of device delay mismatch reduction on bimodal sound source localization accuracy

The contents of this chapter have been previously published as a peer-reviewed article.

Angermeier, J., Hemmert, W., & Zirn, S. (2021). Sound Localization Bias and Error in Bimodal Listeners Improve Instantaneously When the Device Delay Mismatch Is Reduced. *Trends in Hearing*, *25*, 23312165211016164. https://doi.org/10.1177/23312165211016165

3.1 Introduction

Bimodal stimulation for cochlear-implant (CI) users has become a common approach. In such cases, one ear is provided with a CI and the contralateral ear receives a conventional digital hearing aid (HA). Many studies have been published showing a benefit for most bimodal listeners in binaural performance, when both devices were used instead of just one (Ching, Incerti, & Hill, 2004; Ching, Incerti, Hill, & van Wanrooy, 2006; Hoppe, Hocke, & Digeser, 2018; Sheffield, Schuchman, & Bernstein, 2017), along with an improvement in quality of life (Farinetti et al., 2015). Despite the reported benefits, Dorman et al. (2016) showed that in terms of sound source localization bimodal CI/HA users performed more poorly than bilateral CI users and bilateral HA users.

The binaural cues for sound localization in the horizontal plane are interaural level differences (ILDs) and interaural time differences (ITDs) in the temporal envelope and fine structure. ILDs are most prominent at frequencies greater than 1500 Hz. Sound information at such high frequencies is well transmitted with the CI but is often barely audible with the HA because of limited residual hearing (Hoppe et al., 2018). Thus, due to little spectral overlap in both ears, ILD perception may be hampered in many bimodal listeners (Dorman et al. 2015; Seeber, Baumann, and Fastl 2004). But even with limited high-frequency residual hearing at the HA side, the perception of envelope ITDs may still be possible. Dirks et al. (2020) showed in single-sided deaf subjects (SSD) that binaural beats are perceivable with electric/acoustic stimulation over a wide frequency range. Further Francart, Brokx, and Wouters (2009) could measure ITD just noticeable differences (JNDs) for bimodal subjects in direct stimulation experiments with adjusted interaural stimulation timings at acoustic frequencies above 1 kHz, suggesting sensitivity to envelope ITDs. On the other hand, ITDs conveyed in the temporal fine structure of the ear signals are likely to be not perceivable by bimodal listeners because interaural phase information is typically not provided by current CI coding strategies (Zirn et al., 2016).

Another, yet rarely discussed, problem faced by bimodal listeners is that the two different hearing devices may have very different processing delays. Zirn et al. (2015) showed that there can be a temporal delay in the range of 3 to 10 milliseconds between an ear provided with a MED-EL Maestro CI system and the contralateral ear provided with a HA. In the ear provided with the CI, the frequency-dependent latencies arise from the signal processing by the speech processor and have been shown to be relatively close to latencies occurring in the normal-hearing ear in MED-EL Maestro CI systems. In an ear provided with a HA, the absolute latency is a combination of the (mostly) frequency-independent HA processing latency and the physiologically occurring latencies arising from the transmission of sound through outer, middle, and inner ear, where a frequency-dependent latency component is added due to the basilar-membrane travelling wave delay. This temporal asymmetry between the modalities is further referred to as device delay mismatch and may, if present, hamper the perception of envelope ITD and ILD. In the latest study by Zirn et al. (2019), a significant improvement in sound source localization ability was reported for bimodal CI/HA users, when the device delay mismatch was minimized by delaying the CI stimulation. An unresolved question so far is whether the measured hearing aid delay ($t_{Delay} = \tau_{HA}$) is the optimal value to compensate the device delay mismatch. Figure 3.1 shows three different delay values for the CI stimulation (τ_{HA}-1 ms, τ_{HA} and τ_{HA}+1 ms), which were applied in the current study. The latency curves reveal the resulting temporal overlap. The motivation to use these three values for t_{Delay} was based on the findings of Zirn et al. (2015). According to this previous work, τ_{HA}-1 ms leads to a good temporal adjustment of the modalities at lower frequencies (0.5 – 1 kHz), τ_{HA} in the middle frequency range (1-2 kHz), and τ_{HA}+1 ms at higher frequencies (2-4 kHz). The resulting frequency-dependent latencies are relatively close but do not perfectly match the latencies of an ear provided with a HA.

Effects of device delay mismatch reduction on bimodal sound source localization accuracy

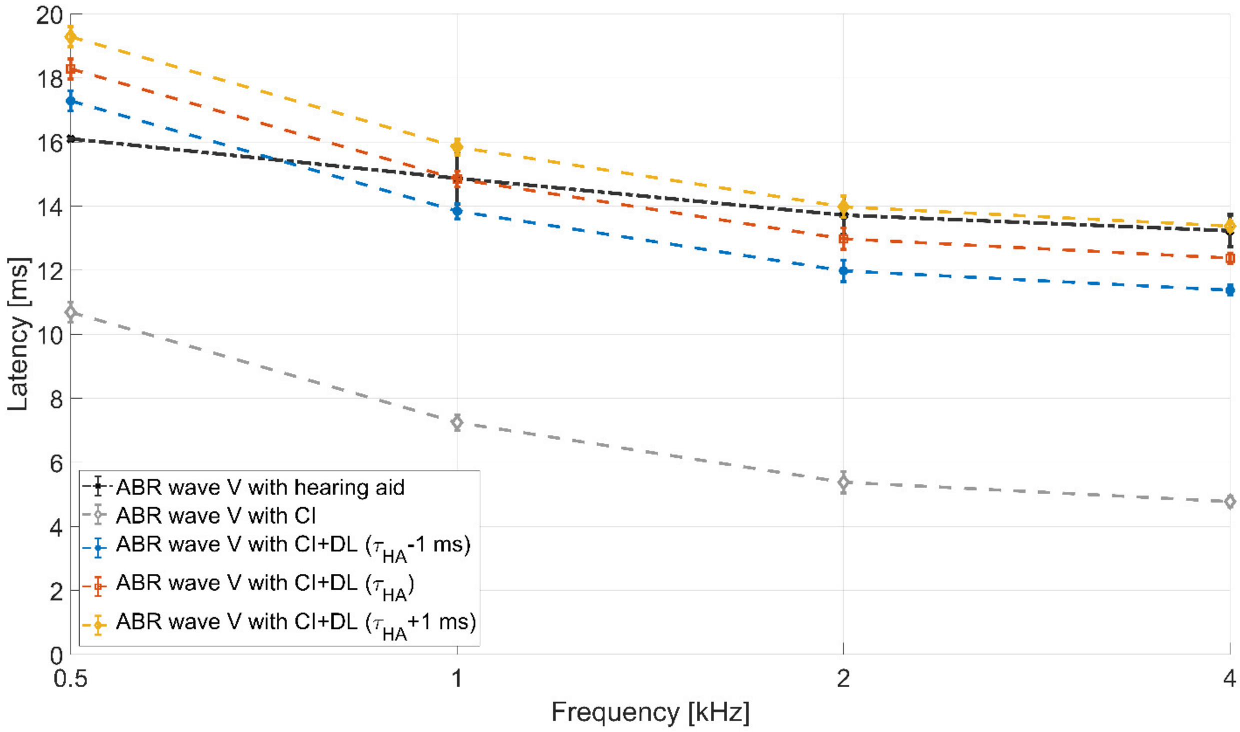

Figure 3.1: Effects of the three applied delays for device delay mismatch reduction on ABR wave V latency with a CI compared to ABR wave V latency with a Phonak Una M HA (τ_{HA} = 7 ms) showing the temporal adjustment in the different frequency ranges for the three applied values for τ_{HA} . Adapted from Zirn et al. (2015).

Evidence that the 1-ms stepsize around τ_{HA} makes a difference comes from Seebacher et al. (2019), who could show an improvement in sound localization if the CI stimulation was delayed by 1 ms in SSD CI users leading to a lower device delay mismatch in the corresponding high-frequency band (2-4 kHz).
Compared to Zirn et al. (2019) the A-B study design was extended to an A-B-B-A design in the present work. In the former A-B study design, localization tests were conducted without minimizing the device delay mismatch (condition A) and after a familiarization period of 1 hour with the minimized device delay mismatch (condition B). In the current study design, localization tests were conducted without minimizing the device delay mismatch (first test in condition A), acutely after minimizing the device delay mismatch (first test in condition B), after one hour of familiarization to the changed delays (second test in condition B), and acutely after resetting the device delay mismatch to its initial value (second test in condition A). This allows us to investigate the effects of the familiarization as well as ruling out procedural training over the course of the study. Furthermore, not only the root-mean-square (RMS) error is analyzed but also the signed bias, which delivers information about the direction of the localization error. Further additional details improved the study design, and all changes are described in the methods section.

3.2 Material and methods

3.2.1 Test subjects

Nine adult bimodal listeners (age: 61.1 ± 6.9 years, min. 47 years, max. 71 years; 3 females, 6 males) participated in the study. Details are listed in Table 3.1 and Table 3.2. None of the subjects had participated in earlier studies.

Table 3.1: Data of bimodal subjects.

Subject	Age [y]	Aetiology	CI type (processor/ implant)	Im-planted side	CI & bimodal experience [years]	HA experience [years]	CI Coding strategy
Bim201	56	progressive	RONDO2/ CONCERTO Flex28	left	1	5	FS4
Bim202	71	Acute hearing loss	OPUS2/ CONCERTO Flex 28	right	8.5	29	FS4
Bim203	61	Blast trauma	RONDO2 / SYNCHRONY Flex28	right	2	6	FS4
Bim204	59	Sudden hearing loss	SONNET / SYNCHRONY Flex28	left	2	9	FS4-p
Bim205	64	Acute hearing loss	SONNET / SYNCHRONY Flex28	right	5.5	4	FS4
Bim206	66	unknown	SONNET EAS / SYNCHRONY Flex28	right	2	3	FS4
Bim207	58	progressive	RONDO / SYNCHRONY Flex28	left	6	11	FS4
Bim208	68	Meniére	SONNET / SYNCHRONY Flex28	right	2	7	FS4-p

Bim209	47	unknown	SONNET / SYNCHRONY Flex28	left	5.5	18	FS4

On the ear provided with the HA, the subjects had mild to severe sensorineural, conductive, or mixed hearing losses (see Figure 3.2 for air conduction pure tone thresholds). Criteria for inclusion of bimodal subjects in the study were: i) everyday use of their HA and CI; ii) a percent correct score of more than 50% obtained in the Freiburg monosyllabic (word) test (Hahlbrock, 1953) at 65 dB sound pressure level (SPL) in a free field measurement (both on the CI and HA side monaural as well as binaural). The test subjects had no residual acoustic hearing in the ear provided with the CI at the frequencies 0.5, 1, 2 and 4 kHz. All subjects had complete insertions except for Bim206, where the 12th electrode was outside the cochlea and the 11th and 12th electrode were deactivated due to high impedances. All testing was conducted in accordance with the Code of Ethics of the World Medical Association (Declaration of Helsinki) for experiments involving humans. Approval by the Technical University of Munich ethics committee was obtained (340/19). All subjects provided written informed consent.

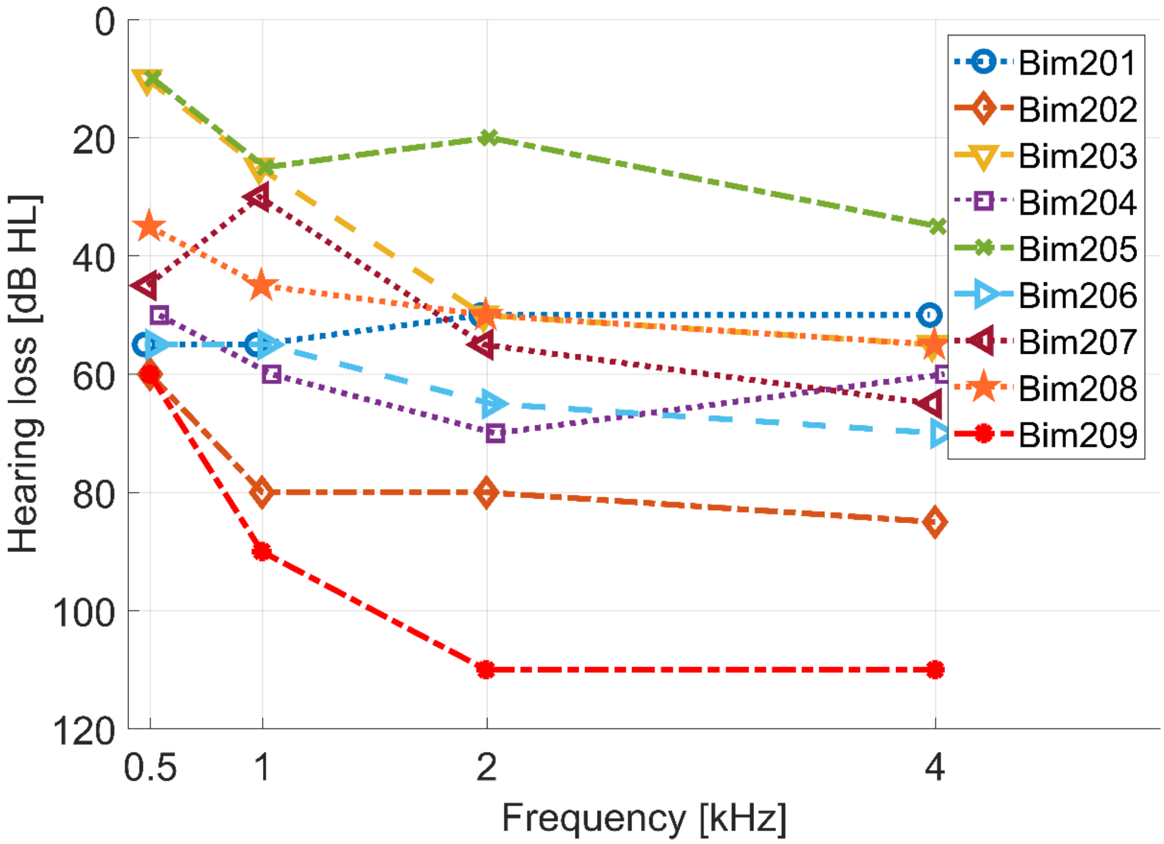

Figure 3.2: Residual hearing of all subjects included in this study on the ear provided with the HA.

3.2.2 The delay line

As the subjects used their own hearing aids in this study, the hearing aid delay (τ_{HA}) was measured for every individual HA the test subjects came with, using either the hearing aid analyzer unit ACAM 5 from Acousticon GmbH, Reinheim, Germany, as described in Zirn et al. (2015), or a self-constructed measuring setup, which delivers similar values. In this setup a white noise burst (100 Hz – 20 kHz) at a level of 65 dB (A) was presented to a reference microphone (Behringer ECM 8000) and the HA microphone. The sound tube of the HA was connected to a measurement microphone, identical to the reference microphone, via a 2 ccm coupler. To calculate frequency-dependent delays, a digital zero-phase bandpass filterbank with center frequencies of 500 Hz to 6 kHz and bandwidths of 500 Hz was implemented in MATLAB. After bandpass filtering, the frequency specific delay was calculated for each band via cross correlation of the hearing aid signal and the reference signal. In case of high-frequency hearing loss, only the delays at frequencies where the hearing loss was ≤ 80 dB HL were considered for averaging. The corresponding τ_{HA} values can be found in Table 3.2.
To delay sound signals at the CI with sufficient temporal resolution in bimodal listeners a programmable delay line (DL) based on the Arduino Due microcontroller (µC) board with a built-in Atmel SAM3X8E ARM Cortex-M3 CPU was used. The DL corresponds exactly to the one used and described in detail in Zirn et al. (2019). With a sampling frequency of 48 kHz, it provides the possibility to delay signals by integer multiples of 20.8 µs while ensuring a very low minimum delay of only 50 µs. The delay applied between analog-to-digital (AD) and digital-to-analog (DA) conversion is based on a ring buffer and is therefore frequency-independent. Both analog-to-digital converter (ADC) and digital-to-analog converter (DAC) have a 12-bit resolution.
The only difference to the DL used in the study from 2019 is the type of power supply. Whereas a 9V rechargeable battery was used earlier, a 2500 mAh lithium-ion battery combined with a step-up voltage regulator was used to provide power to the delay line. With this type of rechargeable battery, weight of the DL could be reduced, and the runtime extended.
In the present study, the time by which the CI stimulation was delayed (t_{Delay}) was set to τ_{HA} (see Table 3.2) which was considered as an estimate for the device delay mismatch (i.e. τ_{HA} ≈ device delay mismatch). This was decided based on the results presented by Zirn et al. (2015) in which the latencies of the CI-stimulation in MED-EL CIs (where the coding strategy uses a filterbank) were comparatively close to the physiological delays introduced mainly through the travelling wave delay on the basilar membrane. In a subset of the test subjects (Bim203, Bim204, Bim205, Bim206, Bim207, Bim208, Bim209), t_{Delay} was also set to τ_{HA}-1 ms and τ_{HA} + 1ms to evaluate if τ_{HA} is really an appropriate value by which to delay the CI stimulation.

Effects of device delay mismatch reduction on bimodal sound source localization accuracy

Table 3.2: Hearing aids of the bimodal subjects and average processing delays (τ_{HA}) of these devices

Test subject	**HA type**	**Averaged τ_{HA} [ms]**
Bim201	ReSound LiNX2 LS9	5.9 (min: 4.3; max: 8.3)
Bim202	Widex Daily 100 Fashion	3.9 (min: 2.6; max: 5.6)
Bim203	Bernafon IN1 N	5.8 (min: 5.5; max: 6)
Bim204	Widex Evoke 220 Fa P	2.8 (min: 1.7; max: 5.3)
Bim205	Widex Daily 50 D-FA	2.8 (min: 2.1; max: 3.0)
Bim206	Widex Beyond 330 B3-F2	2.8 (min: 1.4; max: 5.4)
Bim207	Oticon Selectic Napoli Pro	6 (min: 5.8; max: 7.1)
Bim208	Oticon NovaSense Geneve	5.2 (min: 4.9; max: 5.5)
Bim209	Phonak Naida Q90 UP	7.2 (min: 6.7; max: 7.7)

The DL is inserted into the signal path of the CI as follows: An OPUS2 CI audio processor, worn behind the ear, was used to capture the acoustic signal. This unprocessed signal was fed into the DL where it was delayed. After the delay was applied, the signal was fed into another OPUS2 CI audio processor programmed with the same settings as the subjects' everyday program. Further, in this second OPUS2, the microphones were internally switched off. For further details see Zirn et al. (2019), Figure 3.

3.2.3 Test environment

All tests were conducted in the same audiometric booth as for the earlier study published (Zirn et al., 2019). Seven loudspeakers (type Genelec 8030C) were located at an angular spacing of 30° between -90° (loudspeaker #1) to 90° (loudspeaker #7) in a semicircle in the frontal horizontal plane at the subject's head level with 1 m between the subject's head and each loudspeaker. The loudspeakers carried number plates from 1 to 7. The study participants used an app running on a tablet computer depicting the loudspeaker arc with the numbers of the speakers to type in their responses. These responses were sent to a personal computer outside the audiometric booth via Bluetooth where the data was then processed in MATLAB.

3.2.4 Stimuli

In this study, we presented multiple noise bursts as stimuli in the sound source localization experiments. Stimuli were generated in MATLAB (The Mathworks Inc., Natick, MA, USA) and consisted of five Gaussian white noise bursts (125 Hz to 20 kHz). Bursts had a duration of 70 ms with 3 ms Gaussian-shaped slopes and were separated by 30-ms pauses. A similar type of stimulus was used by Seeber et al. (2004) in a study involving bimodal listeners. To avoid the use of monaural cues in the localization task, spectral roving and level roving was applied to the stimuli for each trial. Level roving was achieved by randomly presenting stimuli at 60, 65 or 70 dB (A). For spectral roving, the stimulus was filtered either by the ipsilateral or contralateral HRTF taken from an open HRTF Database (Kayser et al., 2009) for a stimulus azimuth of 90° according to Van de Heyning et al. (2016). This led to a total of 42 (7 loudspeakers x 3 levels x 2 spectra) different combinations.

3.2.5 Experimental procedure

The complete measurement procedure is illustrated in Figure 3.3.

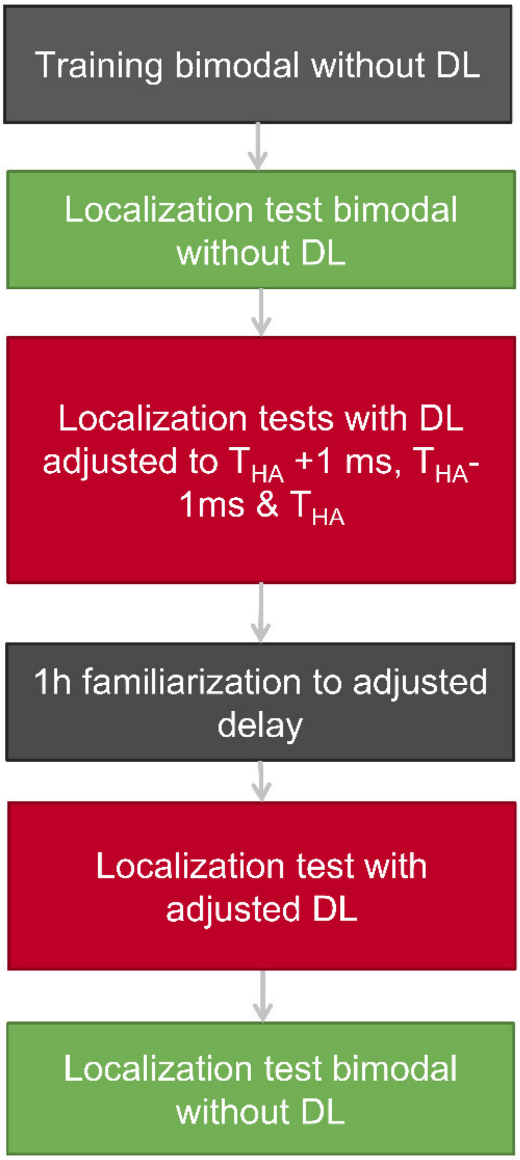

Figure 3.3: Schematic of the A-B-B-A test design (A = green, B = red).

Effects of device delay mismatch reduction on bimodal sound source localization accuracy

Prior to the localization tests, training consisting of 42 trials was provided for every participant, where the participants received feedback via the tablet computer. In case of a wrong response, the correct source position was highlighted in the app. The objective of this training was to familiarize the subjects to the procedure as well as the stimuli used. Subjects were not allowed to search the presenting speaker by moving their heads during the stimulus presentations but could search it before giving their final answer. After training the participants performed at least 4 localization tests in an A-B-B-A paradigm. A total of 84 stimuli were presented in each localization test, meaning that each combination of speaker, level, and spectrum was presented twice. The subjects received no feedback during the tests. In the first test (A), the DL was programmed with a delay of 0 ms, thus representing the everyday device delay mismatch (+50 µs added by the microcontroller) of the participant. After this first localization test, the DL was set to $t_{Delay} = \tau_{HA}$, and another localization test (B) was conducted acutely. This test was conducted to determine whether the effects reported in Zirn et al. (2019) were acute or if a familiarization period to the changed device delay mismatch is required. In 7 of 9 participants t_{Delay} was also set to τ_{HA}-1 ms and τ_{HA}+1ms in randomized order and tested acutely before familiarization. After these acute tests, the DL was programmed with t_{Delay} that yielded the best localization results in the acute tests, that is, the combination of lowest RMS error and lowest absolute signed bias, and the participants had a 1 hr, familiarization period to adapt to the reduced delay mismatch. During this familiarization period of 1 hr, the participants went for a walk on the campus. They were instructed to pay attention to environmental sounds and to locate sound sources (e.g., birds) if possible. After this familiarization period another localization test (B) was conducted to check for effects of familiarization and audiovisual training. Finally, the DL was programmed to t_{Delay} = 0 ms to test training effects over the course of the study. It should be noted that the participant Bim08 only conducted the first A and B test due to a hardware malfunction during the experiment (the battery charging unit was damaged and had to be replaced afterward). The entire measurement procedure, including the familiarization period and breaks when needed, took 3-4 hours, consisting of a minimum of 378 trials for the subjects Bim201 and Bim202 and a maximum of 546 trials for all other subjects except Bim208.

3.2.6 Evaluation and statistical analysis

The RMS errors and signed bias of localization accuracy were calculated as proposed by Rakerd and Hartmann (1986). The RMS error describes the discrepancy between the azimuth of a source and the azimuth of a subject's response to that source corresponding to equation (3.1). Therefore, the RMS error corresponds to the precision of the subject's judgements according to ISO 5725 (International Organization for Standardization, 1994).

$$RMS\ error\ =\ A\sqrt{\frac{1}{M}\sum_{i=1}^{M}(r_i - k_i)^2} \tag{3.1}$$

$$signed\ bias = \frac{A}{M}\sum_{i=1}^{M}(r_i - k_i) \tag{3.2}$$

A corresponds to the angle between two adjacent speakers (30° in the test setup used), *M* is the number of responses, r_i is the response (1 to 7) on the *i*th trial, and k_i is the number of the source on the *i*th trial. The reported RMS error corresponds to the final calculation of the RMS error after all 84 trials. The signed bias reflects the constant error or an error in trueness according to ISO5725 (International Organization for Standardization, 1994) in the listeners response. The signed bias can either be positive, indicating a bias of the listener to the right or negative, indicating a bias towards the left.

Statistical analysis of the localization RMS errors and signed bias of the test subjects included pairwise comparisons using Wilcoxon signed-rank tests with an alpha level of 0.05. Before statistical analysis, the signed bias for subjects having their CI on their left side were inverted. Therefore, a positive signed bias corresponds to a bias towards the CI.

3.3 Results

3.3.1 Best delay for device delay mismatch reduction

Figure 3.4 and Figure 3.5 show the localization results for the seven participants that conducted localization tests with t_{Delay} set to τ_{HA} , τ_{HA}-1ms and τ_{HA}+1ms. Figure 3.5 shows the RMS error for those 7 participants, which instantaneously improved when the device delay mismatch was reduced. There was no clear tendency for which value for t_{Delay} yielded the best results.

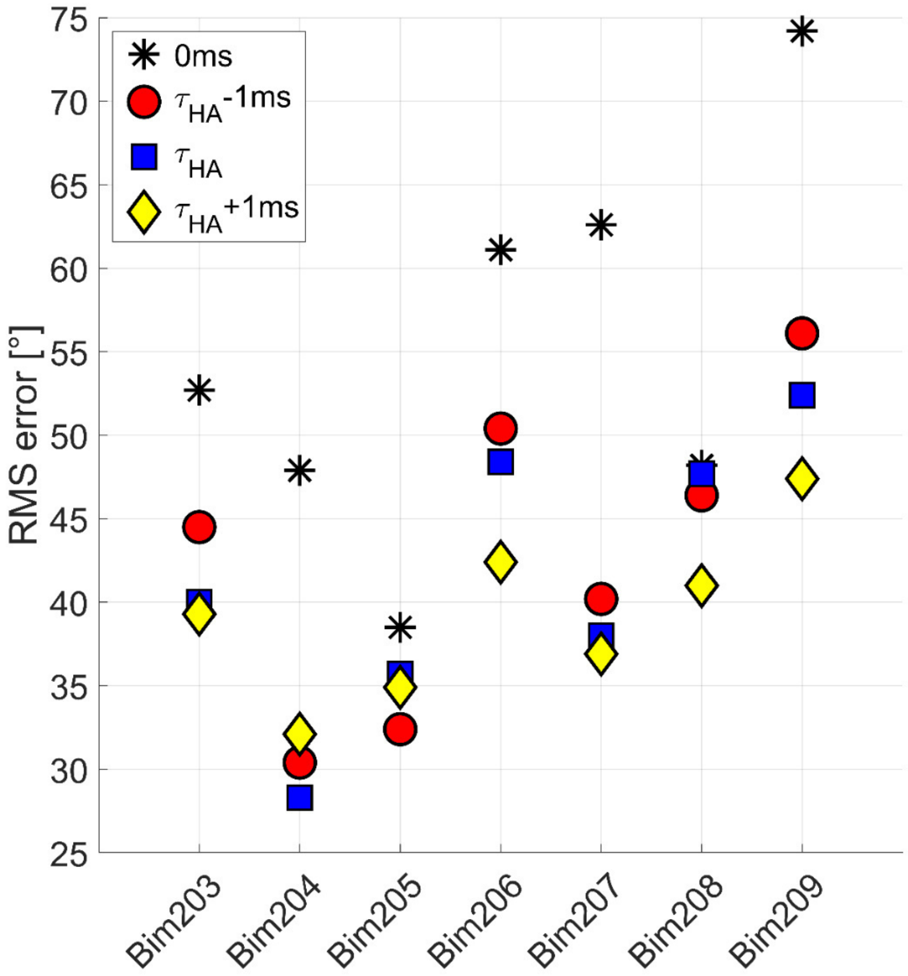

Figure 3.4: RMS errors for three different delay values in acute sound source localization testing for subjects Bim203 to Bim209 in the first A and B conditions.

In Figure 3.5 the results for participants having their CI on the left side were inverted so that positive values always indicate a bias towards the CI. The dashed line represents zero bias which corresponds to perfect trueness in localization judgments. The data clearly showed a bias towards the CI that can be shifted towards 0° when the delay mismatch is compensated, for the values we tested; however, it reversed its sign only in one subject.

Effects of device delay mismatch reduction on bimodal sound source localization accuracy

When considering the signed bias, most patients had the best outcome with t_{Delay} set to τ_{HA}+1ms. As the RMS error was similar in the acute tests for each delay applied to CI stimulation compared to the initial condition without a CI delay, the value of t_{Delay} that yielded the lowest signed bias (i.e., the value closest to zero) was chosen and programmed into the DL for the following tests. An exception was Bim204 where τ_{HA}+1ms led to a direction reversal, i.e., a negative signed bias. Furthermore, in his case, the RMS error was worse with τ_{HA}+1ms compared to τ_{HA}. Therefore, τ_{HA} was programmed into the DL for further testing in case of Bim204.

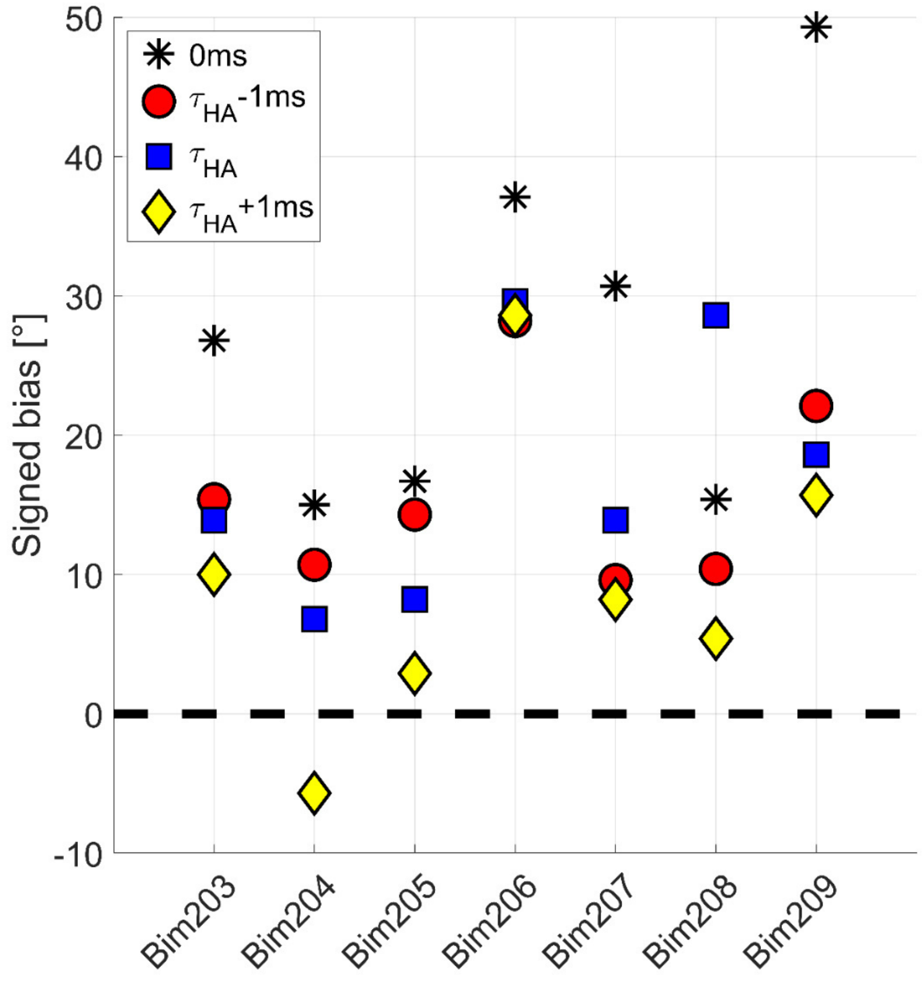

Figure 3.5: Signed bias for three different delay values in acute sound source localization testing for subjects Bim203 to Bim209 in the first A and B conditions.

3.3.2 Sound source localization accuracy

Figure 3.6 and Figure 3.7 show the localization results for all 9 participants in the A-B-B-A test design. The thick black line represents group means and standard deviations. In Figure 3.7 the data for participants wearing their CI on the left side have been inverted so that positive values always indicate a bias toward the CI. The average RMS error and standard deviation in the initial condition was 52.6 ± 11.4°. The average RMS error was 37.9 ± 5.7° when the device delay mismatch was reduced. After 1 hr of familiarization, the average RMS error remained almost unchanged at 40.1 ±8.3°. In the last condition (A), when the CI delay was removed, i.e., the device delay mismatch set to the initial value, the mean RMS error was at 47.6 ± 9.3°.

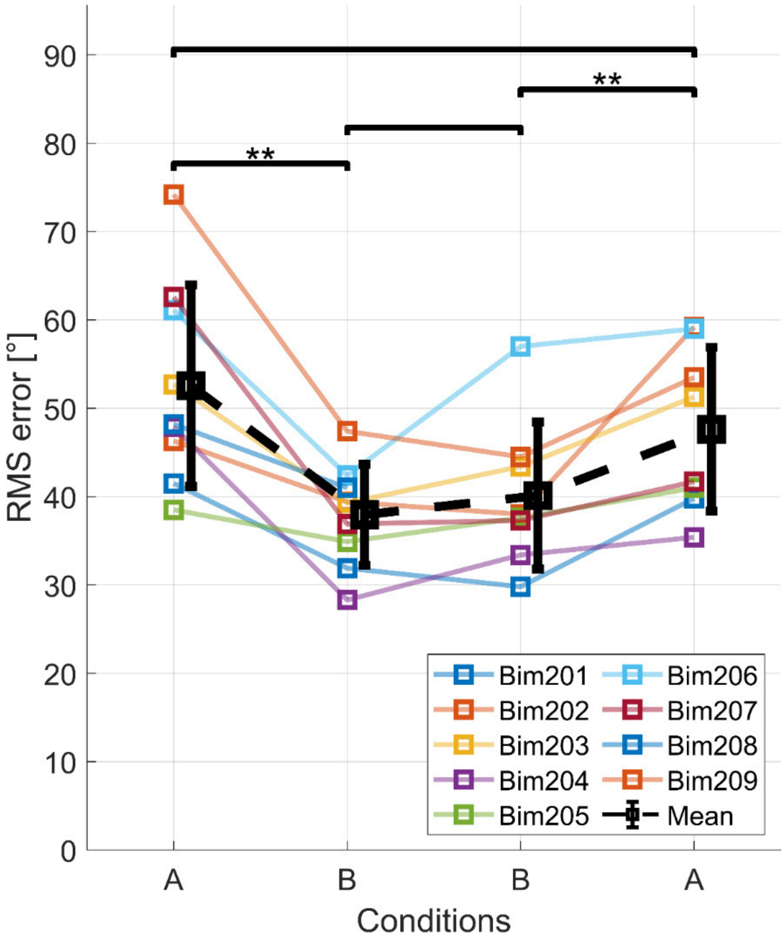

Figure 3.6: RMS errors in the sound source localization test for nine participants in the A-B-B-A test design (significance levels: ** represents p ≤ 0.01).

The mean signed bias was 25.2 ±11.9° in the initial condition. After reduction of the device delay mismatch, the mean signed bias was 10.5 ± 8.2°. After one hour of familiarization to the reduced device delay mismatch the mean signed bias was 14.1 ± 11.9°. When the initial

device delay mismatch was restored, the average signed bias increased again to 26.8 ± 9.7°. Comparisons based on Wilcoxon signed-rank tests revealed that a reduction of the device delay mismatch led to a statistically significant instantaneous improvement of average -14.6 ± 8.5° in RMS error ($p < 0.01$) and -14.7 ± 9.2° signed bias ($p < 0.01$). After 1 hr of familiarization, no further improvement could be shown in RMS error with a mean difference and standard deviation of 2.6 ± 5.7° ($p = 0.3$) or signed bias with a mean difference and standard deviation of 2.9 ± 7° ($p = 0.3$). When the device delay mismatch was set to the initial condition, RMS error and signed bias showed a statistically significant deterioration ($p < 0.01$). The mean difference for RMS error was at 7.5 ± 6.4° and the for the signed bias the mean difference was at 12.7 ± 8.1°.
Because there was no significant difference between the two A conditions in RMS error ($p = 0.3$, mean difference and standard deviation: 5.5 ± 11.7°) and signed bias ($p = 0.7$ mean difference and standard deviation: 0.4 ± 13.9) effects of procedural learning over the course of the experiments can be ruled out. All reported data is available on request.

Effects of device delay mismatch reduction on bimodal sound source localization accuracy

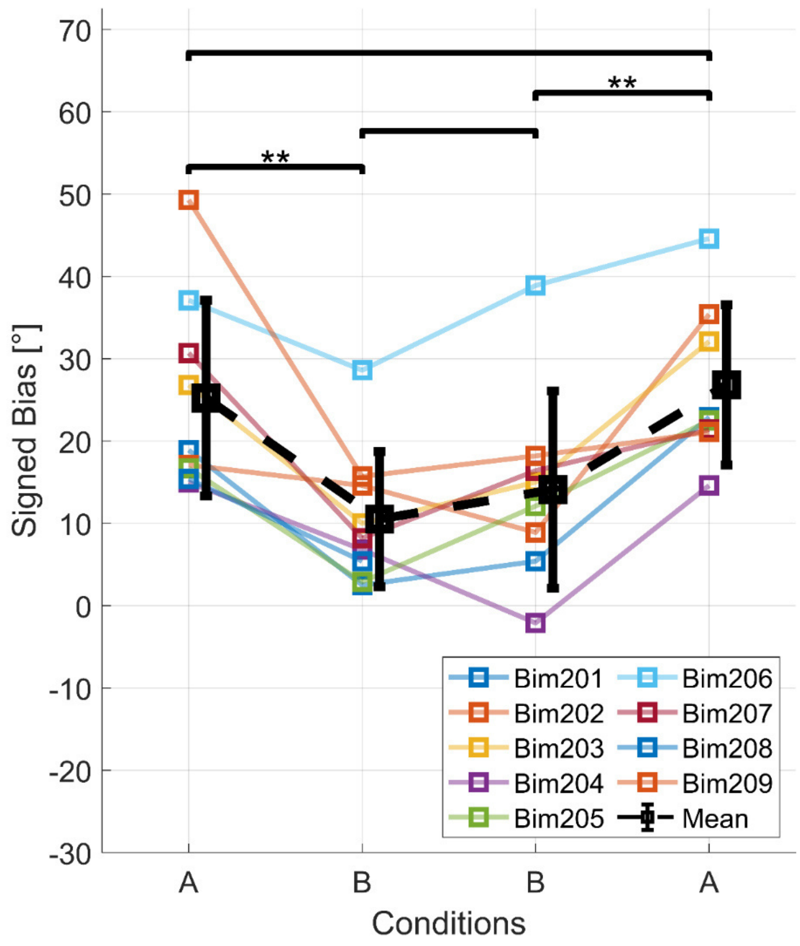

Figure 3.7: Signed bias in the sound source localization test for nine participants in the A-B-B-A test design (significance levels: ** represents $p \leq 0.01$).

In Figure 3.8, angle-dependent results show that the RMS error and signed bias before device delay mismatch reduction is highest on the HA side. The negative signed bias at 90° is most likely an edge effect, because the subjects could not input any speakers at more positive angles than 90°. The reduction of device delay mismatch led to an improvement of RMS error and signed bias at almost every speaker position, being most prominent on the speaker at -90° i.e., the speaker directly on the HA side. For the speaker at 90° a slight deterioration of RMS error and signed bias could be observed.

Effects of device delay mismatch reduction on bimodal sound source localization accuracy

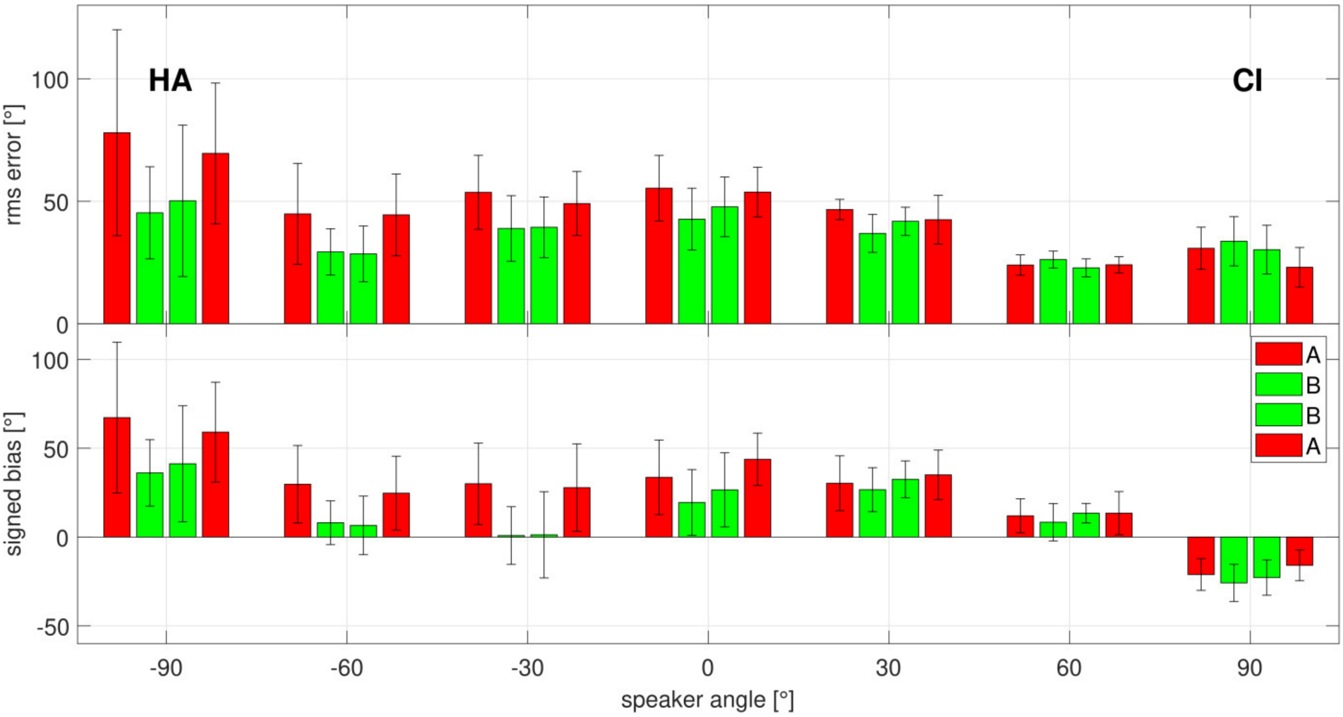

Figure 3.8: Speaker dependent means and standard deviations for RMS error and signed bias for nine participants in the A-B-B-A test design. The data was inverted so the position of the CI is on the right ear in all participants.

3.4 Discussion

In this study, the effect of the temporal adjustment of the CI processing delay and the HA processing delay was investigated in bimodal listeners. The outcomes show that i) differences of the processing delays of hearing devices in bimodal listeners severely impair sound source localization in the horizontal plane and ii) this impairment can be mitigated at least partially by adding a simple DL to reduce the device delay mismatch. We found that the improvement in sound source localization is immediate, which is in line with a previous study (Zirn et al., 2019). No further improvement in localization accuracy was found after a familiarization period of one hour, but it is unclear if this one-hour training was sufficient. Another outcome of the applied A-B-B-A test design is that effects of procedural learning can be ruled out, as the results deteriorated instantly in the second A condition to performance levels similar to the initial A condition.
Furthermore, this study shows that the reduction of the device delay mismatch improved not only the precision of the subjects' judgements, expressed by the RMS error, but also the trueness of the localization judgements expressed by the signed bias. This bias showed an orientation towards the faster modality (i.e., the CI) in all nine participants when the CI stimulation was not delayed. This indicates that a readjustment of sound localization is not achieved by neural plasticity even after several months or years of bimodal hearing (in our

study the mean bimodal experience was 3.8 years). Interestingly, sound localization accuracy improved instantly in all nine participants with all values of t_{Delay} namely τ_{HA} *-1 ms,* τ_{HA} and τ_{HA} *+1 ms.* This further shows that even if the optimal setting of $t_{Delay}$is not yet determined, adjustment did not prove to be disadvantageous for any of the participants. It should be noted that the device delay mismatch is frequency dependent (Zirn et al., 2015). Auditory brainstem response (ABR) wave V delays match best at a frequency of 1 kHz, when the CI is delayed by the overall time delay of the hearing aid τ_{HA} . For 2 and 4 kHz, matching is better with $t_{Delay} = \tau_{HA}$ *+1 ms* and for 500 Hz with $t_{Delay} = \tau_{HA}$ *-1 ms*. The fact that all participating bimodal listeners benefitted from reducing the device delay mismatch is in line with the hypothesis that envelope ITD sensitivity improves with temporal alignment. Envelope ITD can be perceived across a wide frequency range by SSD CI users (Dirks et al., 2020) and at frequencies above 1 kHz by bimodal CI/HA users with sufficient temporal alignment of both modalities (Francart et al., 2009). Furthermore, improved temporal alignment in higher frequency regions by delaying the CI stimulation with τ_{HA} *+1 ms* may especially be helpful for ILD perception for those bimodal listeners who have considerable residual hearing at higher frequencies. This is in accordance with findings by Seebacher et al. (2019), who found that sound source localization precision in MED-EL CI users with SSD is highest when the CI stimulation is delayed by 1 ms on top of the processing delay reported by Zirn et al. (2015), resulting in an improved temporal matching of electric and acoustic hearing at higher frequencies in SSD CI users.
For bimodal listeners provided with other CI systems than those of MED-EL different CI delays have to be considered. Wess, Brungart, and Bernstein (2017) reported a delay of the CI ear relative to a normal-hearing ear of 10.5-12.5 ms for Cochlear Ltd. and 9-11 ms for Advanced Bionics, which is considerably more than for CI systems of MED-EL. In such cases the HA instead of the CI stimulation must be delayed to reduce the device delay mismatch. However, this approach has limitations, as HA processing latencies above 10 ms have been shown to cause subjective disturbances in patients (Agnew & Thornton, 2000; Bramsløw, 2010; Groth & Søndergaard, 2004).
Another, yet unknown, factor is how crucial a potential interaural tonotopic mismatch between modalities is for binaural processing. For example, in cases of incomplete insertion or for short electrode arrays, predominantly basal fibers are excited by the CI whereas on the ear provided with the HA, often only the apical region of the cochlea is sufficiently stimulated. In envelope ITD detection tasks it was found that a sufficient interaural match of excited characteristic frequencies is important (Bernstein et al., 2018; Dirks et al., 2020; Hu & Dietz, 2015). Furthermore, in bimodal listeners with pronounced residual hearing at higher frequencies (> 1500 Hz) ILD may facilitate sound localization. In contrast to envelope ITD, ILD are presumably relatively robust against an interaural tonotopic mismatch as Francart and Wouters showed (2007). In their experiments ILD was usable for lateralization of sounds even for interaural frequency shifts up to 1 Octave. Kan et al. (2019) showed that shifts in interaural place-of stimulation have high effects on the lateralization based on ITDs but lateralization based on ILDs was more robust in bilateral CI users. If the tonotopic alignment is found beneficial for binaural processing, this could

be achieved by adjustment of the frequency allocation table in CI systems either by using post-operative imaging (Landsberger, Svrakic, Roland, & Svirsky, 2015) or through psychoacoustic frequency matching techniques such as ITD discrimination (Bernstein et al., 2018; Hu & Dietz, 2015), interaural pitch comparison (Hu & Dietz, 2015), or sensitivity to binaural temporal envelope beats (Dirks et al., 2020). The optimization in the frequency domain just discussed, in combination with the optimization in the time domain (by the reduction of device mismatch), is a promising way to further improve bimodal hearing in the future.

3.5 Conclusions

Our study shows that sound source localization improves in bimodal listeners when the temporal mismatch in the processing delays of HA and CI is reduced. A simple implementation with a frequency-independent DL on the CI side is already very effective. Because none of the nine test subjects showed a deterioration in localization accuracy in this study, we conclude that a temporal alignment between CI and HA by delaying the CI stimulation is a viable step to further improve bimodal provision.

4

Spatial release from masking in the presence of a reference ITD

The contents of this chapter have been accepted for publication on the 28.03.2022 as a peer-reviewed article.

Angermeier, J., Hemmert, W., & Zirn, S. (2022). Measuring and Modeling Cue Dependent Spatial Release From Masking in the Presence of Typical Delays in the Treatment of Hearing Loss. *Trends in Hearing*, *26*. https://doi.org/10.1177/23312165221094202

4.1 Introduction

Both sound localization and spatial release from masking (SRM) rely on precise timing of the signals reaching both ears and on symmetrical frequency-place mapping. However, listeners with hearing loss who use a hearing device only in one ear or who are provided with different devices in both ears can suffer from vast asymmetric hearing perceptions. In this study, we focus mainly on temporal asymmetries. For example, hearing instruments like hearing aids (HAs) and cochlear implants (CIs) can have large processing delays in the order of several milliseconds, which depend on their exact implementation. Timing differs due to processing delays but also due to physiologic latencies of acoustic hearing introduced by the ear canal, middle ear and mainly the cochlear delays in the inner ear (Ruggero & Temchin, 2007), which are not present when the auditory nerve is directly excited by electric current in the case of CI stimulation. The impact of this temporal alteration on subjective disturbance or sound quality has been studied extensively for provision with HAs (Agnew & Thornton, 2000; Bramsløw, 2010; Groth & Søndergaard, 2004; Stone & Moore, 2003; Stone, Moore, Meisenbacher, & Derleth, 2008). However, most of the studies were looking at maximal preferable delays and did not take the possibility of asymmetric timing between both ears into account. In treatment of asymmetric hearing loss, the ears often must be provided with different technical devices. Examples for this are unilateral hearing loss which requires provision with a HA on only

one ear, treatment of single-sided deafness (SSD) with a CI, or bimodal provision with a CI in one ear and a contralateral HA. Besides interaural differences in signal processing and sound information (e.g., fine structure and spectral information), a considerable interaural difference typically is a static temporal mismatch of the ear signals. For bimodal listeners provided with a MED-EL CI and a contralateral HA, an average interaural temporal difference of 7 ms in the neural representation of incoming signals at the level of the brainstem was reported (Zirn, Arndt, Aschendorff, et al., 2015). Furthermore, interaural differences in latency in the millisecond range have recently been found in some hearables (Denk, Schepker, Doclo, & Kollmeier, 2020).
This interaural temporal mismatch severely impairs the ability of bimodal listeners to localize sounds in the frontal horizontal hemisphere. Equalizing the device delay mismatch by delaying the CI stimulation according to the measured HA processing delay resulted in highly significant improvements in the root-mean-square error and in the bias of localization judgements (chapter 3; Zirn, Angermeier, Arndt, Aschendorff, & Wesarg, 2019). The static interaural temporal mismatch will further be referred to as reference interaural time difference (ITD), i.e., the ITD at 0° azimuth, which is close to 0 µs when no temporal interaural asymmetry is present. What has not yet been systematically studied is the influence of large reference ITDs on speech understanding in noise, especially SRM. For SRM, processing of ITDs and interaural level differences (ILDs) contribute to unmasking when speech and noise are spatially separated (Lavandier & Best, 2020).
In listeners provided unilaterally with a HA or two different HAs with different processing latencies, ITDs are conveyed by the HAs and can be used for SRM alongside with ILDs. It has to be noted that these ITDs can be distorted in case of an open fitting, due to direct sound through the open earpiece overlapping with the delayed amplified signal (Denk et al., 2019). On the other hand, in HA users where the residual hearing in the lower frequencies is sufficient and no amplification is needed by the HA, processing latencies will not affect ITD processing. Bimodal listeners, however have very limited access to ITDs (Francart & McDermott, 2013; Veugen, Chalupper, Snik, Opstal, & Mens, 2016). Therefore, bimodal users rely mostly on ILDs for SRM. Dieudonné & Francart (2020) have reported that SRM in bimodal listeners is mostly driven by monaural head shadow effects, i.e., one ear having access to a better signal-to-noise ratio when noise and speech are spatially separated. This effect should theoretically not be affected by an additional reference ITD.
Differences in the accessibility of cues in the presence of a reference ITD makes it interesting to examine the cues facilitating SRM separately. This study aimed to investigate if and how different reference ITDs affect SRM. To get a more complete picture of how the different binaural mechanisms underlying this process are affected by these asymmetries, tests were conducted with cues presented separately: A) ITDs and ILDs, B) ITDs only or C) ILDs only. SRM without a reference ITD but with respect to the available cues has been measured in prior studies. Bronkhorst & Plomp (1988) showed a substantial SRM for speech and laterally displaced noise of 7.8 dB when only ILDs were available, a 5 dB SRM difference when ITDs were available and 10.1 dB when both cues could be utilized together. Further studies investigated the role of different cues exploited

for SRM with symmetrically placed maskers to reduce better-ear listening with symmetrically placed speech maskers (Ellinger, Jakien, & Gallun, 2017; Glyde, Buchholz, Dillon, Cameron, & Hickson, 2013; Kidd, Mason, Best, & Marrone, 2010). These studies found that ITDs and ILDs alone were sufficient to elicit SRM. Another objective of this study was to determine how well the effect of a reference ITD on SRM can be predicted by an existing phenomenological computational spatial unmasking model. Computational binaural models have already been successfully applied in the past to accurately predict SRM in hearing-impaired listeners (Beutelmann, Brand, & Kollmeier, 2010; Vicente, Buchholz, & Lavandier, 2021; Williges, Dietz, Hohmann, & Jürgens, 2015; Zedan, Williges, & Jürgens, 2018). If a model can replicate the experimental results in normal-hearing listeners, it might also be a valuable tool to predict outcomes in unilateral HA users or even in CI users with single-sided deafness (SSD) and bimodal listeners.

4.2 Methods

4.2.1 Test environment and Stimuli

All tests were conducted in an audiometric booth. Speech reception thresholds (SRTs) were measured using the Oldenburg Sentence Test (OlSa), a German matrix sentence test (Wagener, Brand, & Kollmeier, 1999). To measure SRM, subjects performed speech tests with noise and speech coming from the front (S_0N_0) and speech coming from the front and noise coming from 90° azimuth (S_0N_{90}). Olnoise was used as a masker which is composed by overlaying the jittered speech material of the OlSa sentences 30 times resulting in a broadband noise with the same long-term average speech spectrum as the speech material. Speech material and noise were presented via Sennheiser HD 280 Pro headphones with a fixed noise level of 65 dB SPL and the speech level adaptively varied according to the subjects' answers. Subjects used a tablet computer displaying the word material of the OlSa to input their answers. The answers were sent to a computer outside the audiometric booth via Bluetooth and processed on a computer using MATLAB (The MathWorks Inc., Natick, MA, USA). In-ear head related impulse responses (HRIRs) were used to allow for spatial separation of speech and noise (Kayser et al., 2009). A reference ITD was introduced by delaying the signals on the right ear channel by 0, 1.75, 3.5, 5.25 and 7 ms. To further investigate the effect of the reference ITD on different cues used for spatial unmasking, the HRIRs were manipulated according to Kulkarni et al. (1999) and Ellinger et al. (2017) to only offer either A) ITDs and ILDs, or B) ITDs only, or C) ILDs only. In the A) condition the HRIRs for 0° and 90° azimuth were not manipulated. In the B) condition for 90° azimuth the right ear channel of the HRIR at 0° azimuth was delayed by the delay found in the HRIR at 90° azimuth, thus not offering any ILDs and purely ITDs. In the C) condition the right ear channel of the HRIR at 90° azimuth was shifted by the delay such that ILDs were present, while removing ITDs. A graphical representation can be found in Figure 4.1.

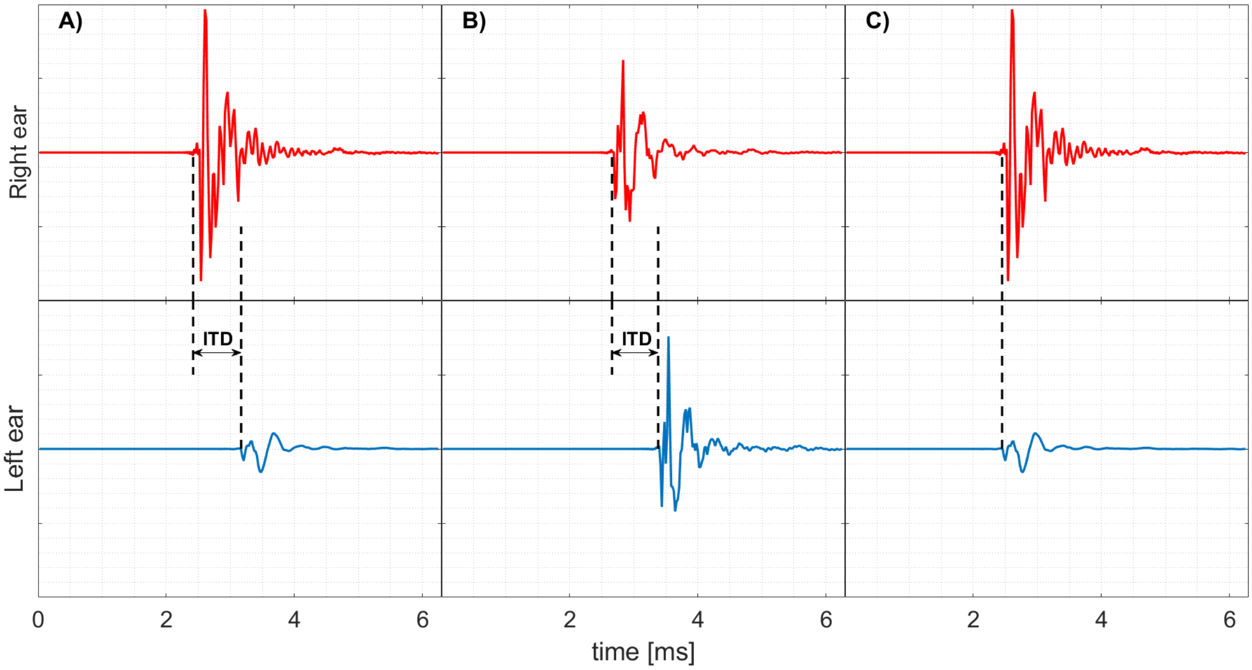

Figure 4.1: Head-related impulse responses at 90° azimuth for the different cue conditions (A: ITD and ILD; B: only ITD; C: only ILD)

4.2.2 Experimental Procedure

Prior to the speech tests pure tone audiometry was performed at the frequencies of 0.5, 1, 2 and 4 kHz to ensure normal hearing in the participants according to WHO standards (Olusanya, Davis, & Hoffman, 2019). For each reference ITD, each condition (A, B, C) and each spatial scenario, two OlSa lists of 20 sentences were measured. This resulted in a total of 60 measured OlSa test lists per participant. Each combination of interaural cue and reference ITD was tested in blocks of 4 lists containing two lists in each condition, S_0N_0 and S_0N_{90}, with both lists being averaged for the calculation of the SRM. The presentation was randomized within and between these blocks. To avoid fatigue effects on the results the testing was done in up to 3 sessions lasting approximately 3 hours with breaks after each block or if the subjects desired a break. Two training lists of 20 OlSa sentences were applied at the beginning of each session.

4.2.3 Subjects

Ten normal-hearing subjects (mean age: 25.6 ± 6; min: 21; max: 42; 2 female; 8 male) participated in this study. Their hearing threshold at 0.5, 1, 2, and 4 kHz did not exceed 20 dB HL with mean thresholds of 8.4 ± 1 dB HL for the right ear and 7.5 ± 0.7 dB HL for the left ear. All subjects provided written informed consent prior to their participation. All testing was conducted in accordance with the Code of Ethics of the World Medical

Association (Declaration of Helsinki) for experiments involving humans and approved by the Technical University of Munich ethics committee (340/19).

4.2.4 Simulations

4.2.5 Model Structure

For each condition, SRTs were simulated using a blind equalization-cancellation (EC) model (Hauth, Berning, Kollmeier, & Brand, 2020) implemented in the auditory modeling toolbox (Majdak, Hollomey, & Baumgartner, 2021). The model uses the mixed speech and noise signals of the right and left ear channels as inputs and splits them into 30 ERB spaced frequency bands between 150 Hz and 8500 Hz using a gammatone filterbank (Hohmann, 2002). The frequency bands up to 1500 Hz are then fed into an EC mechanism (Durlach, 1963) to model binaural unmasking. The cancellation is performed either by subtracting (level-minimization) or adding (level-maximization) the equalized left and right ear channel to account for negative as well as positive SNRs. After this step a blind decision stage based on a modulation analysis by Santos, Senoussaoui, & Falk (2014) is used to select whether the level-minimization or the level-maximization yields the better SNR. The selected path is then used for further processing. The frequency bands above 1500 Hz are only used for a better-ear processing with a SRMR selector to determine which ear channel has the better SNR. Both paths (EC and better ear) are then combined via a gammatone synthesis filterbank to be further processed by the back end of the model. As a back end the speech intelligibility index (SII; ANSI S3.5-1997) was used. For more details concerning the model structure, see Hauth et al. (2020).

4.2.6 Modeling Parameters and SRT Calculations

Each measured combination of noise azimuth, utilized cue and reference ITD was modeled. To further extend the modeling predictions all modeling was additionally done at 10 ms reference ITD. For each condition the mixed speech and noise were used as an input to the model at 21 SNRs between 0 and -20 dB SNR at 1 dB steps. For each SNR, ten sentences of the OlSa were presented to make sure every word of the OlSa speech material appeared once. Further, 10 Monte-Carlo simulations were run per sentence to account for the random jitter in the EC process of the model. This process resulted in 2100 simulations per combination of noise azimuth, cue, and reference ITD, with an overall number of 69300 simulations for all combinations. The SRT was calculated as described by Hauth et al. (2020) by using the intersection of the mean SII over all Monte-Carlo simulations and sentences and the mean experimental SRT measured at S_0N_0 without a reference ITD and with both spatial cues available. The resulting SII was further used as a reference SII for all other conditions modeled.
To investigate the influence of more realistic environments, especially the influence of reverberation, the model was also applied with an in-ear HRIR set recorded in a cafeteria within the same HRIR database by Kayser et al. (2009). In this setup the head and torso simulator (HATS) used to record the HRIRs was seated at a table with the target speaker at

0° azimuth opposite to it. The distance between HATS and target speaker was 102 cm. The target speaker faced the HATS. The spatially separated speaker was located at 90° azimuth to the HATS at a distance of 52 cm. In the original HRIR dataset this speaker is at -90° but the room was flipped by switching the right-ear and left-ear channel of the HRIRs. The spatially separated speaker was oriented towards the frontal speaker. For the given cafeteria setting the reverberation time T_{60} was 1250 ms. For this setting reference ITDs were to the same as the ones applied in the anechoic condition. For SRT extraction the experimentally determined reference SII from anechoic measurements was used.

4.2.7 Statistical Analysis

Non-parametric Friedman tests with an alpha level of 0.05 were used to test for differences between different reference ITDs in the measured SRTs and SRM results for all conditions. In the case of significant outcomes post-hoc pairwise testing was applied via Wilcoxon signed-rank tests with a Bonferroni-Holm correction for multiple comparisons. To compare the measured data with the modeling results linear regression was performed. We performed all statistical testing in MATLAB.

4.3 Results

4.3.1 Experimental Results

Figure 4.2 shows the measured SRTs as boxplots for the respective cue available to the listeners. For each reference ITD the green boxplots correspond to the S_0N_0 condition, and the blue boxplots correspond to the S_0N_{90} condition. In the S_0N_0 condition Friedman tests revealed no significant difference for rising reference ITD for the condition A), with both cues available ($\chi2(4) = 5.56$, $p = 0.2$), B) ($\chi2(4) = 7.98$, $p = 0.09$), and C) ($\chi2(4) = 7.56$, $p = 0.1$). In the spatially separated condition significant differences for differing reference ITD could be seen in condition A) ($\chi2(4) = 34.71$, $p < 0.001$) and in condition B) ($\chi2(4) = 37.01$, $p <0.001$). If only ILDs were present (condition C)), a rising reference ITD did not lead to a significant change in the measured SRTs ($\chi2(4) = 7.67$, $p = 0.1$). Test-retest reproducibility was calculated as mean absolute difference between the two measured testlists per condition, reference ITD and spatial configuration revealing 0.7 ± 0.2 dB in condition A) 0.7 ± 0.1 dB in condition B) and 0.9 ± 0.2 dB in condition C). No significant differences were found between the conditions.

Figure 4.3 shows the calculated SRM for each of the cue conditions. In condition A), the mean SRM at 0 ms reference ITD was 8.82 dB with a standard deviation of 1.12 dB. With a rising reference ITD the SRM decreased to 4.63 ± 0.89 dB at 7 ms reference ITD. Friedman tests revealed a significant influence of reference ITD in this condition ($\chi2(4) = 37.61$, $p < 0.01$). For condition B), the SRM at 0 ms was 5.48 ± 1.21 dB and decreased to 1.12 ± 0.41 dB at a reference ITD of 7 ms. This effect also proved highly significant ($\chi2(4) = 35.04$, $p < 0.01$). In condition C), the SRM at 0 ms reference ITD was 4.81 ± 0.97 dB and slightly decreased at 7 ms reference ITD to 3.94 ± 0.81 dB. The SRM in this condition was also significantly affected by the rising temporal asymmetry between both ears ($\chi2(4) = 10.64$, $p = 0.031$). Pairwise comparisons showed significant differences between all measured SRM values in condition A) ($p < 0.05$). In condition B) only the differences in SRM at 0 ms vs 1.75 ms ($p = 0.09$) and at 3.5 ms vs 5.25 ms reference ITD ($p = 0.09$) did not prove significant. In condition C) none of the tested differences proved to be significant. We hypothesize this to be due to better-ear listening dominating the SRM, which is not influenced by a rising reference ITD, given its monaural nature. At a reference ITD of 7 ms, there was almost no more SRM due to ITDs in condition A) and the SRM measured approached the SRM caused by ILDs seen in condition C) which was not influenced by the reference ITD.

4.3.2 Modeling Results

Figure 4.2 shows the modeled SRTs for S_0N_0 as diamonds and for S_0N_{90} plotted as circles. The model's behavior matched the experimental results qualitatively. For S_0N_0 the coefficients of determination (R^2) and root-mean-square errors (RMSE) are reported in Table 4.1. The low values for R^2 in the conditions B) and C) are not surprising given the shallow slope within the datasets.

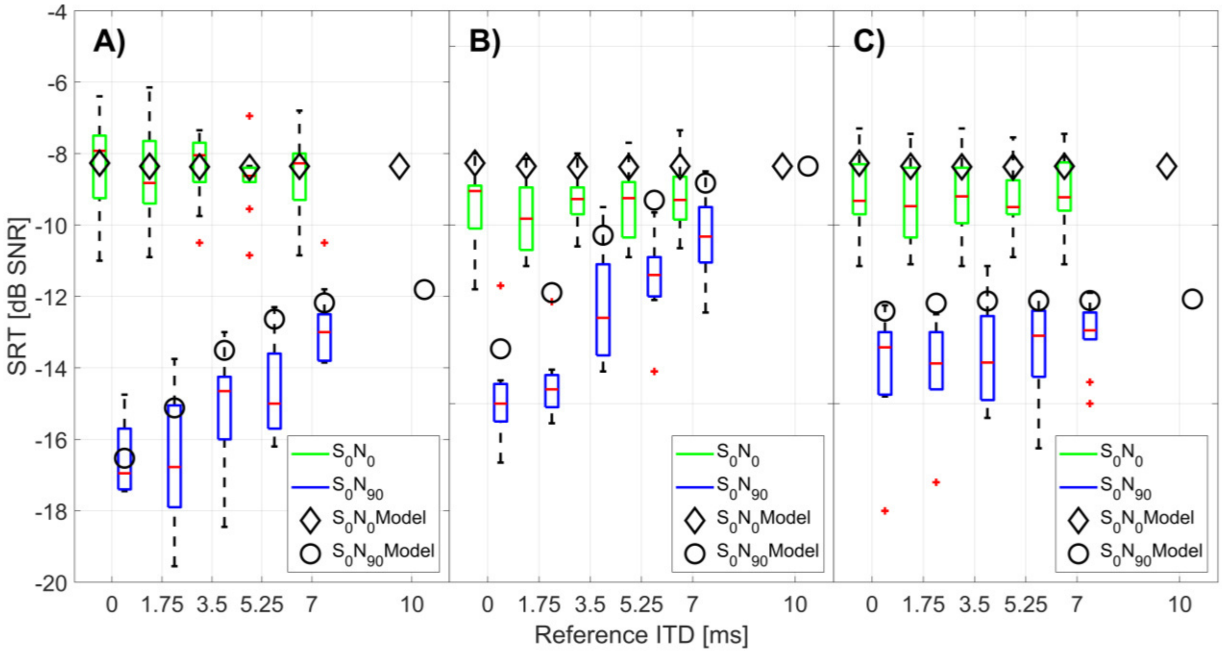

Figure 4.2: Measured speech reception thresholds (SRT) in ten normal-hearing subjects for reference ITD of 0, 1.75, 3.5, 5.25 and 7 ms as boxplots (red line: median; box: 1st-3rd quartile; whiskers: minimum and maximum without outliers; outliers in red) for either collocated speech and noise at 0° azimuth (green boxes) or speech at 0° and noise at 90° azimuth (blue boxes). Modeled SRTs for collocated speech and noise are denoted by diamonds, and for spatially separated speech and noise by circles. Modeled SRTs depicted up to 10 ms reference ITD. A) ITDs & ILDs condition B) ITDs only condition C) ILDs only condition.

For the S_0N_{90} condition the coefficient of determination were generally much higher as Table 4.1 shows.
The additional modeled SRTs at a reference ITD of 10 ms show further deterioration in the S_0N_{90} condition in conditions A) and B) but not in condition C).

Table 4.1: Coefficients of determination (R^2) and root-mean-square errors (RMSE) for the linear regression between modeled and measured results. A) ITDs & ILDs condition B) ITDs only condition C) ILDs only condition.

	A		**B**		**C**	
	R^2	**RMSE [dB]**	**R^2**	**RMSE [dB]**	**R^2**	**RMSE [dB]**
S_0N_0	0.6	0.03	0.02	0.05	0.07	0.05
S_0N_{90}	0.92	0.6	0.93	0.5	0.3	0.1
SRM	0.95	0.5	0.97	0.4	0.38	0.2

Figure 4.3 depicts a comparison between the modeled SRM and the measured SRM. Linear regression was performed to determine the accuracy of the model. The corresponding coefficients of determination and RMSEs for the comparison between modeled and measured SRM can be found in Table 4.1. The additionally modeled SRM for a reference ITD of 10 ms show that a reference ITD of 10 ms eliminates SRM based on ITD as seen in condition B).

Spatial release from masking in the presence of a reference ITD

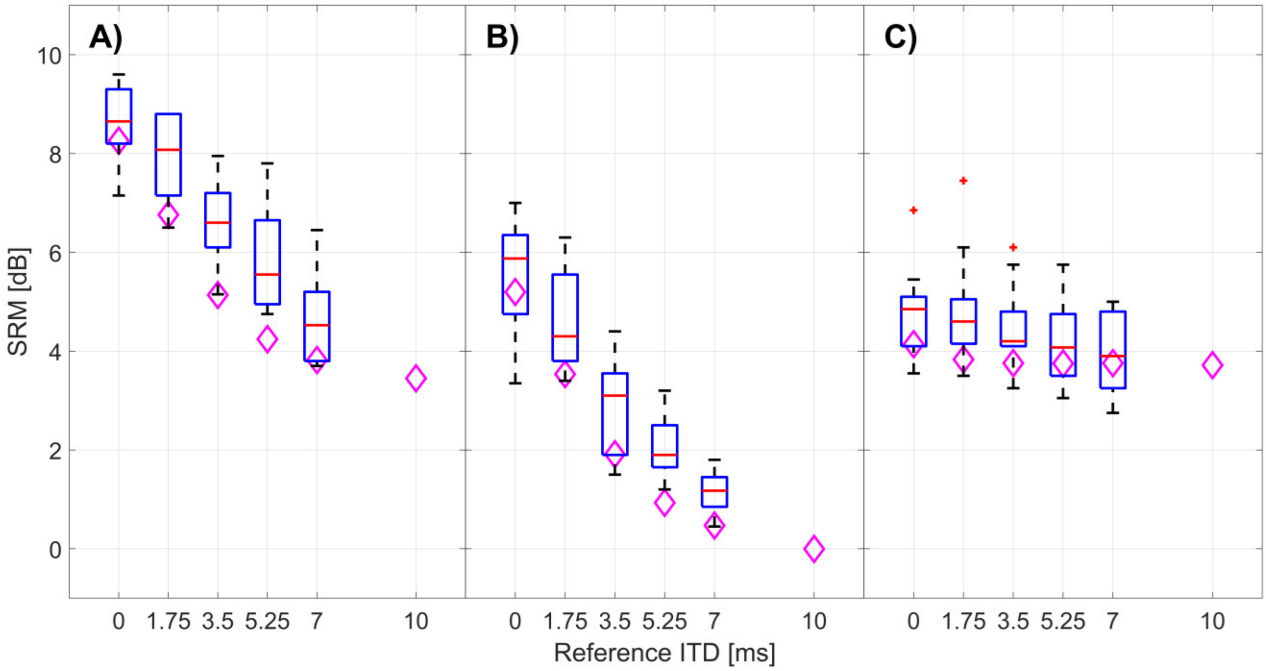

Figure 4.3: Measured spatial release from masking (SRM) in ten normal-hearing subjects for reference ITD of 0, 1.75, 3.5, 5.25 and 7 ms as boxplots (red line: median; box: 1st-3rd quartile; whiskers: minimum and maximum without outliers; outliers in red). Modeled SRM denoted by diamonds. Modeled SRM depicted up to 10 ms reference ITD. A) ITDs & ILDs condition B) ITDs only condition C) ILDs only condition.

In Figure 4.4 the modeled SRTs for S_0N_0 and S_0N_{90} condition are depicted for the anechoic condition in magenta and the reverberant (cafeteria) condition in blue. For S_0N_0 condition the difference between anechoic condition and cafeteria condition was quite low with a mean absolute difference of 0.15 dB for condition A), 0.14 dB for condition B) and 0.14 dB for condition C). In the S_0N_{90} condition, the mean absolute difference between the anechoic condition and the cafeteria condition was 1.1 dB for condition A), 0.9 dB for condition B) and 1.2 dB in condition C).

Spatial release from masking in the presence of a reference ITD

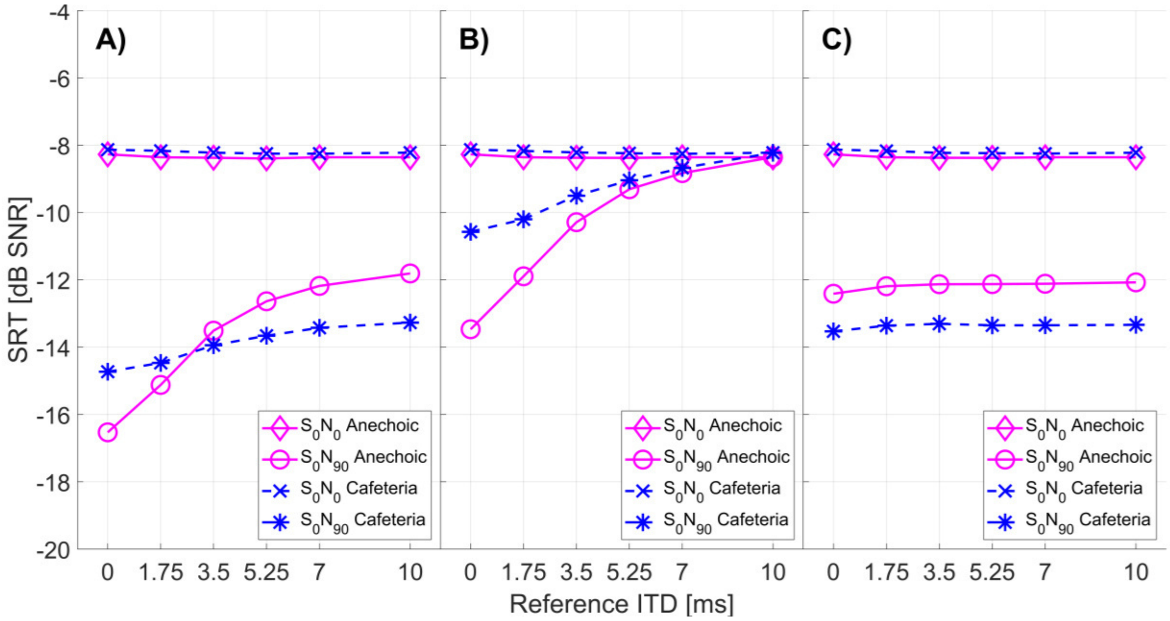

Figure 4.4: Modeled speech reception thresholds (SRT) for reference ITDs of 0, 1.75, 3.5, 5.25, 7 & 10 ms for spatially collocated and spatially separated target and masker in an anechoic setting (magenta) and a reverberant cafeteria environment (blue). A) ITDs & ILDs condition B) ITDs only condition C) ILDs only condition.

Figure 4.5 displays the modeled SRM for the anechoic environment in magenta and the cafeteria environment in blue. For condition A) the initial SRM at 0 ms reference ITD is 8.3 dB in the anechoic environment and 6.6 dB in the cafeteria environment. At a reference ITD of 10 ms the model predicted SRM of 3.5 dB for the anechoic environment and 5.1 dB in the cafeteria environment, showing a lower influence of reference ITD on SRM in the reverberant environment. When only ITDs were present in the signal (condition B)), SRM at 0 ms reference ITD is at 5.2 dB in the anechoic environment and at 2.4 dB in the cafeteria environment. At 10 ms reference ITD both SRM in the anechoic environment and in the cafeteria were at 0 dB in this condition.

With only ILDs present in the signals SRM for the anechoic environment was 4.1 dB at 0 ms reference ITD and slightly deteriorated to 3.7 dB at 10 ms reference ITD. In the cafeteria environment SRM by ILD was at 5.4 dB for a reference ITD of 0 ms and at 5.1 dB at 10 ms reference ITD.

Spatial release from masking in the presence of a reference ITD

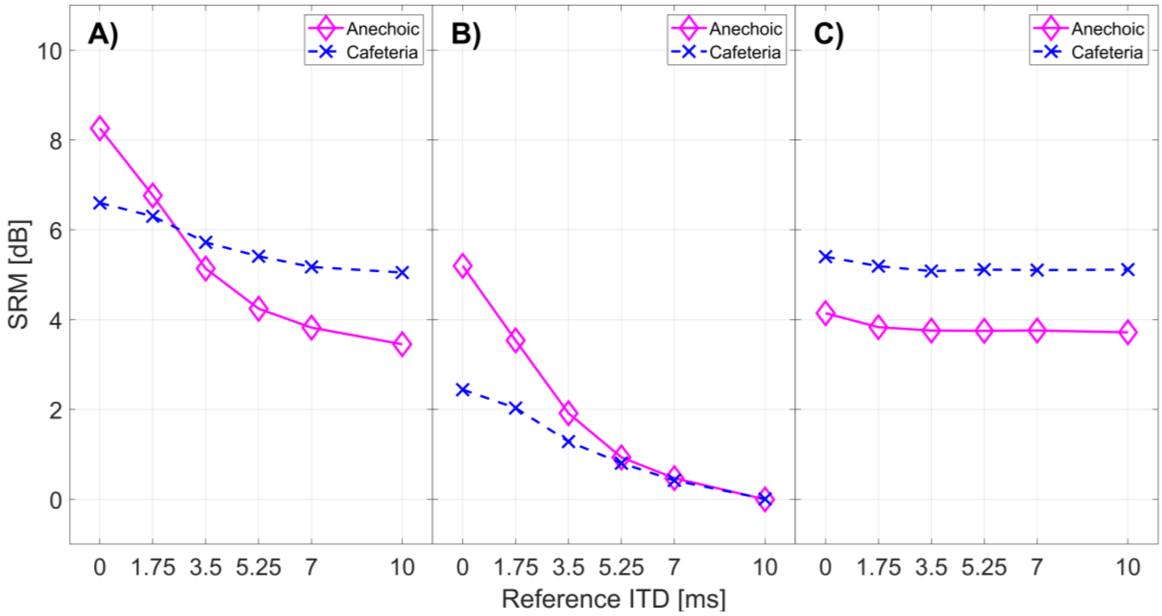

Figure 4.5: Modeled spatial release from masking (SRM) for reference ITDs of 0, 1.75, 3.5, 5.25, 7 & 10 ms in an anechoic setting (magenta) and a reverberant cafeteria environment (blue). A) ITDs & ILDs condition B) ITDs only condition C) ILDs only condition.

4.4 Discussion

In this study, the influence of reference ITDs of several milliseconds on SRM was investigated. Unilateral delays in this range were found in bimodal HA/CI users (Zirn, Arndt, Aschendorff, et al., 2015). To deepen the understanding of the effect of a reference ITD, all measurements were conducted with the different binaural cues that enable SRM separately. Further, we investigated whether the measured effects can be reproduced by an EC-model for binaural unmasking. The results showed that an increasing reference ITD led to a significant decrease in SRM, which affected unmasking by ITDs more drastically than unmasking though ILDs. When both cues were available to the subjects, unmasking due to ITDs and ILDs were roughly additive. Furthermore, the measured results were accurately predicted by an EC-model. When a reverberant environment was used with the model the decrease in SRM by reference ITDs was less drastic than in an anechoic condition.
In contrast to previous studies investigating SRM with isolated cues, the reported results showed that with a reference ITD of 0 ms, SRM achieved when ITDs were used to separate target and masker was bigger than the SRM achieved when only ILDs were present in the signal. These differences probably originated from different masker types. Glyde et al. (2013) applied symmetrically placed speech maskers to measure SRM, whereas Bronkhorst & Plomp (1988) used a single source of masking noise with a speech envelope. Both maskers allow the use of dynamic head shadow in the quiet parts of the masker, allowing better speech understanding when only ILDs are present compared to

ITDs alone. These methodological differences would require further experiments to allow for sufficient comparisons between our data and the data reported in the literature.
The presented results reveal a limiting factor to SRM in listeners with asymmetric treatment of hearing loss, which has not yet been reported. A reference ITD in the range of a few milliseconds has already significant detrimental effects on the unmasking of speech when ITDs and ILDs are conveyed sufficiently by the hearing devices. In the ITDs only condition (B), a reference ITD of 7 ms almost eliminated SRM. In the ILDs only condition (C) a reference ITD did not influence SRM significantly, processing in the auditory system in this condition seems more robust against a reference ITD. However, with binaural measurements it is only possible to differentiate between SRM based on binaural mechanisms and access to one ear having a better signal to noise ratio, which is a monaural mechanism to some extent. To disentangle these mechanisms, monaural measurements could be utilized to measure the effect of head shadow as proposed by Dieudonné & Francart (2019). The present findings also suggest that, while SRM based on ITDs diminishes for a reference ITD of 7 ms, considerable SRM remains due to better-ear listening. In bimodal listeners provided with a CI and a contralateral HA, the device delay mismatch between HA and CI is expected to have no big impact on SRM, since only ILDs and not ITDs are sufficiently conveyed by the CI (Francart & McDermott, 2013; Veugen, Chalupper, Snik, Opstal, et al., 2016). Consequently it has been shown that SRM in bimodal listeners is driven by head shadow (Williges et al., 2019). However, as soon as more sophisticated CI coding strategies code ITDs with higher fidelity in the future, the impact of a reference ITD on SRM needs to be considered as a significant limiting factor, especially since significant deterioration of SRM can already be seen at a reference ITD of 1.75 ms. This leads to the hypothesis that, when both cues are available for listeners, a matching accuracy of processing latencies in their respective devices below this reference ITD of 1.75 ms should be considered favorable.
The applied model was able to predict the experimental data with sufficient precision. This is probably due to the combination of an EC-processing and a monaural SRMR processing, where the monaural part still offers substantial SRM in the presence of ILDs when the EC-processing fails to deliver release from masking. The detrimental effect of a reference ITD on the EC-processing can be easily explained by the processing errors included in this model, that were originally proposed by Vom Hövel (1984). These processing errors for the estimation of ITDs are dependent on the actual ITDs in the signal. Thus, as the reference ITD increases, the processing errors of the ITD equalization within the model increase, making the equalization less accurate. The level equalization, however, is not affected by the reference ITD since mathematically, the ILD processing errors are dependent on the ILDs that are found within the signal. This allows the model to accurately describe the effects that can be measured experimentally. The low value for R^2 in the ILDs only condition can be explained due to the measured data being explained better by its own mean than by the model, which is not surprising as to the SRM was only minimally affected by an increasing reference ITD. However, the low RMSE in this condition still demonstrated a good accuracy of the model. Overall, the model can predict the measured SRM in the presence of a reference ITD of several milliseconds with high precision.

Spatial release from masking in the presence of a reference ITD

In future studies the model by Hauth et al. (2020) could possibly be used to model performance in listeners with asymmetric hearing loss in which a reference ITD greater than 0 ms occurs due to the asymmetries in treatment. Williges et al. (2015) and Zedan et al. (2018) both used EC-based models to accurately predict SRM in simulated bimodal listeners, given some additional preprocessing of the signal to simulate the CI and the HA. These results in combination with the high accuracy the model provides in normal-hearing listeners supports the hypothesis that modeling the effects of a reference ITD in hearing-impaired listeners with EC-based models is possible and that models are a valuable tool for the prediction of treatment outcomes.
When comparing the modeling results in an anechoic environment to a more realistic reverberant scenario the influence of a rising reference ITD is much lower. This is due to the SRM based on ITD being vastly diminished in a reverberant environment. The influence of reverberation on ITD processing has been shown for fine structure ITD (Devore & Delgutte, 2010) and for envelope ITD (Monaghan, Krumbholz, & Seeber, 2013). In the ILD only condition SRM was better in the reverberant environment than in the anechoic environment irrespective of the reference ITD, suggesting that ILDs play a more important role in spatial unmasking in reverberation. However even in the reverberant environment an overall SRM decay of 1.6 dB between 0 and 10 ms reference ITD can be observed when both ILD and ITD are present in the signal. To further verify whether these modeling results can be considered realistic, more experimentation in the same reverberant conditions with normal-hearing listeners should be undertaken in the future.

4.5 Conclusions

In normal-hearing listeners, a reference ITD has a significant detrimental influence on SRM. Our results show that this reference ITD mainly impairs the effect of ITDs in the signal on SRM. When only ILDs were presented, a reference ITD of a few milliseconds showed no significant effect on SRM. These results may be particularly relevant for bimodal listeners with different devices in each ear. The measured results can be accurately predicted by an EC-based model from Hauth et al. (2020) for binaural unmasking of speech. In a reverberant environment, modeling showed a smaller but still detrimental effect of reference ITD on SRM, mainly due to the small proportion of SRM based on ITD in reverberation. These results however should be verified experimentally.

5

Influence of longer familiarization periods on the effects of device delay mismatch reduction in bimodal subjects

This chapter will be submitted as a journal article in a shortened form in the future in an open access journal.

5.1 Introduction

Sound localization is an important mechanism in everyday life. From localizing an approaching car to spatial unmasking in multi-talker environments, it helps humans to navigate in everyday life. In bimodal listeners provided with a hearing aid (HA) in one ear and a cochlear implant (CI) in the contralateral ear, sound localization is significantly impaired compared to normal-hearing listeners (Dorman, Natale, & Loiselle, 2018; Seeber et al., 2004; Veugen, Hendrikse, et al., 2016). One possible explanation of this poor performance are mismatches between both modalities (Pieper et al., 2021). In bimodal listeners, both modalities can be theoretically mismatched in three dimensions: level, pitch, and latency. In the domain of level, this mismatch has its origin in both devices usually being fitted independently and little attention is paid to the balancing of both devices in terms of perceived loudness. Further, the optimal strategy to balance loudness in bimodal subjects is subject of ongoing research (Francart & McDermott, 2012; Veugen, Chalupper, Snik, van Opstal, & Mens, 2016). In the case of frequency mismatch, the limited insertion depth of the CI electrode does usually allow the lowest frequencies in the apex of the cochlea to be stimulated. This leads to a mismatch of low frequencies processed by the CI. The most apical electrodes stimulate slightly higher frequency regions than the acoustic ear, since often for the frequency-to-electrode mapping the default setting is used, irrespective of the insertion depth of the CI electrode. Such a mismatch between both ears has been shown to reduce the binaural benefit such as binaural sensitivity, ITD and ILD sensitivity and speaker separation in CI users with single-sided deafness (SSD), bilateral users of CIs and in normal-hearing listeners (Bernstein et al., 2018; Francart & Wouters, 2007; Hu & Dietz, 2015; Wess et al., 2017). Another form of mismatch in frequency is the reduced spectral overlap between both ears (Veugen, Hendrikse, et al., 2016). With HAs only sufficiently stimulating at low acoustic frequencies when high-frequency hearing loss is present and CIs stimulating at higher acoustic frequencies, the spectral overlap in stimulation between both ears is reduced. The last mismatch and thus one of the possible reasons for reduced accuracy in sound source localization accuracy is latency. In bimodal listeners, a systematic offset in stimulation timing is present. This offset originates from different processing latencies in the two devices, called a device delay mismatch (Zirn, Arndt, Aschendorff, et al., 2015). All three of these dimensions of mismatch are thought to interact with each other. In this work, the effects of a latency mismatch in bimodal listeners were investigated separately from the other potential mismatches. Previous studies reported a significant improvement in sound source localization accuracy when this device delay mismatch was reduced (see chapter 3; Zirn et al., 2019), irrespective of mismatches in level or frequency. However, these studies utilized experimental signal processing to delay CI stimulation in the form of programmable, wearable delay lines (DL). These experimental setups, which were worn around the neck, had the shortcoming that effects of familiarization periods of several weeks to changes in device delay mismatch have been hard to investigate. Long term effects become especially interesting when results presented by Trapeau & Schönwiesner (2015) are considered. In their study, familiarization to a systematic delay of 625 µs on one ear was investigated in normal-hearing subjects over several days. Subjects showed decreased localization accuracy that was subject to familiarization over the course of one week with the most significant effects of familiarization taking place after one day. However, familiarization was not able to completely compensate for the detrimental effect of this unilateral delay.

Further, previous studies only measured the effects of a device delay mismatch without paying attention to directionality features or other pre-processing algorithms. This was tested specifically in this study to check for interactions between these parameters.

With the latest generation of their speech processors, the CI manufacturer MED-EL offers the possibility to delay CI stimulation in their fitting software. Further, they supply clinicians with a list of HA delays from the HA manufacturers. Since HA latencies can vary slightly depending on the pre-processing enabled as shown by Alexander (2016), it is unclear whether these HA delays provided by the manufacturers match the delays that are measured acutely in the HAs and if differences between these delays have effects on improvement in sound source localization. Thus, these effects were investigated in the presented study.

Another topic that has not been thoroughly addressed in previous studies is speech understanding and especially the improvement in speech reception thresholds (SRTs) when the masker and the target are spatially separated, the so-called spatial release from masking (SRM). Other studies only investigated the influence of a device delay mismatch on SRTs when speech and noise were presented from the same loudspeaker in front of the subjects and could not report any effect of device delay mismatch in SSD subjects (Seebacher et al., 2019). This is not too surprising, however, since speech understanding is not affected by binaural cues when both masker and target are collocated. It is currently believed, that SRM in bimodal subjects is manly moderated by one ear having a better signal-to-noise ratio (SNR) and thus being driven by monaural factors that should in theory not be affected by a device delay mismatch (Dieudonné & Francart, 2020; Williges et al., 2019). However, an influence of a device delay mismatch on these findings has not been investigated systematically so far, leaving the question whether this missing access to binaural mechanisms in SRM is due to impaired binaural processing in subjects experiencing such a mismatch.

5.2 Methods

5.2.1 Subjects

Eleven bimodal listeners participated in this study (8 male / 3 female), with a mean age of 58.7 years (min: 33; max: 73). Inclusion criteria for this study were everyday use of both CI and HA and a bimodal experience of at least 6 months. Details about the subjects can be found in Table 5.1.
Subjects Bim201, Bim202, Bim203, Bim204, Bim205, Bim206 and Bim209 already participated in an earlier study (see chapter 3). All subjects had complete insertions of their CI electrode except for subject Bim206, who had an incomplete insertion of the electrode array and thus did not use electrodes 11 and 12. Bim212 had electrode 12 deactivated due to poor sound quality. Subject Bim209 had electrode 5 deactivated due to excessive noise from this electrode. Finally, subject Bim213 had electrode 4 deactivated because of subjective disturbance of sound quality. All testing was conducted in accordance with the Code of Ethics of the World Medical Association (Declaration of Helsinki) for experiments involving humans. Approval by the Technical University of Munich ethics committee was obtained (340/19). All subjects gave written informed consent for each individual study date and were financially compensated for their travel expenses and time.

Table 5.1: Data of all bimodal subjects (CI = cochlear implant; HA = hearing aid)

Subject	**Age**	**Aetiology**	**Implanted side**	**CI (implant/electrode)**	**CI experience [years]**	**HA experience [years]**	**CI coding strategy**
Bim201	56	Progressive	Left	CONCERTO/Flex 28	1.5	6.5	FS4-p
Bim202	72	Sudden hearing loss	Right	CONCERTO/FlexSoft	9	30	FS4
Bim203	62	Blast trauma	Right	SYNCHRONY/Flex 28	2.5	6.5	FS4
Bim204	60	Sudden hearing loss	Left	SYNCHRONY/Flex 28	2.5	9.5	FS4-p
Bim205	65	Sudden hearing loss	Right	SYNCHRONY/Flex 28	6	4.5	FS4
Bim206	67	Unknown	Right	SYNCHRONY/Flex 28	3	3.5	FS4
Bim209	48	Unknown	Left	SYNCHRONY/Flex 28	6	19	FS4
Bim210	43	Progressive	Right	SYNCHRONY/Flex 28	6	8	FS4

Bim211	33	Progressive	Left	SYNCHRONY/Flex 28	2.5	15	FS4
Bim212	73	Progressive	Right	SYNCHRONY/Flex 24	2.5	51	FS4
Bim213	67	Acute hearing loss	Left	SONATA/Standard	13	41	FS4

On the ear provided with the HA, the subjects' hearing loss ranged from mild to severe sensorineural hearing loss. Hearing thresholds can be found in Figure 5.1. On the ear provided with the CI none of the subjects had residual hearing at the acoustic frequencies of 0.5, 1, 2, and 4 kHz.

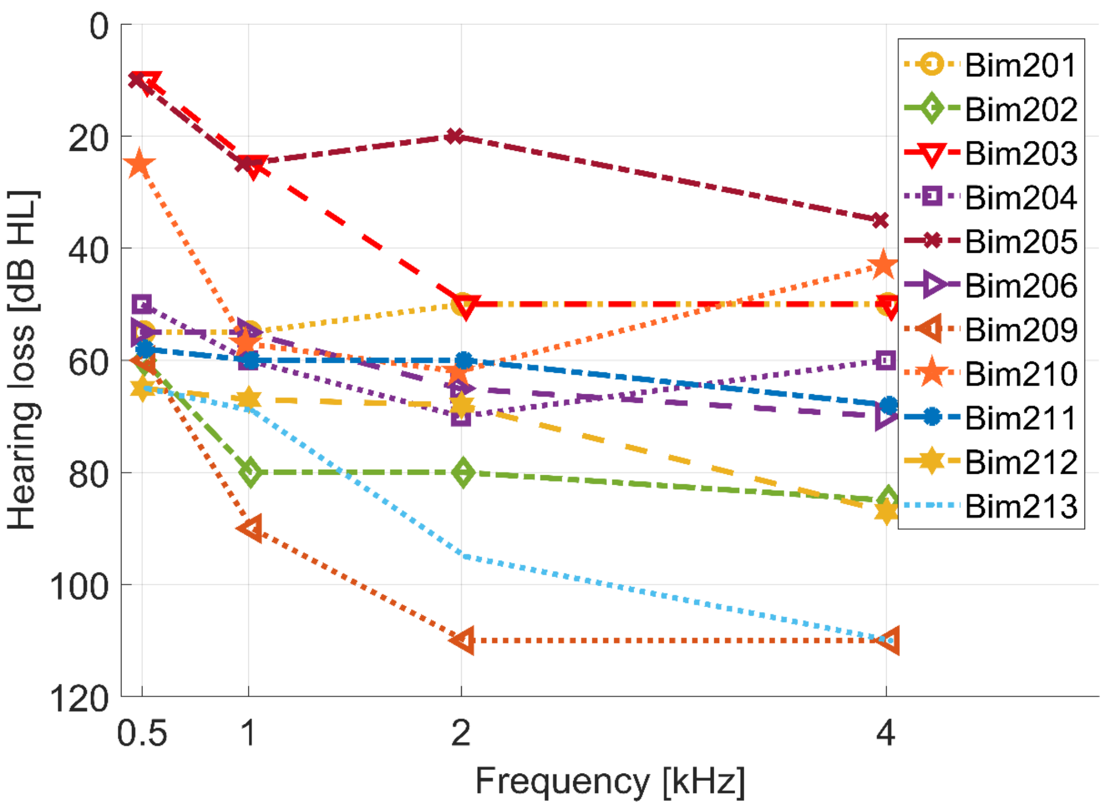

Figure 5.1: Hearing thresholds of the eleven subjects on the ear provided with a HA.

5.2.2 Experimental procedure

The study consisted of a total of four study dates with a testing duration of 2-3 hours each. All study dates took place in the NeuroAkustik laboratory of the University of Applied Sciences Offenburg. Between each study date a familiarization period of 3-4 weeks was administered to investigate the effect of longer familiarization periods as in previous studies (chapter 3; Zirn et al., 2019). We used an A-B-B-A test design to measure the effects of a mismatch compensation as in a previous study. In condition A no reduction of the device delay mismatch was performed to measure baseline performance of the subjects. In the first B condition we performed measurements acutely after reducing the device delay mismatch. The second B condition was performed on the following study date, thus enabling a familiarization to the adjusted delay of three to four weeks. Finally, the second A condition was measured acutely after switching off the device delay mismatch reduction. With this study design acute improvements or deteriorations can be measured by comparing the first A and B condition and the second B and A condition. Improvements due to familiarization can be investigated by comparison of the first and second B conditions. With the comparison of the first A condition versus the second A condition, learning effects over the course of the study and effects of familiarization to a device delay mismatch can be investigated. Two approaches to the compensation of the device delay mismatch and the utilized preprocessing algorithms on the HA and the CI were compared, with each being tested on two consecutive study dates. These two approaches will be called the “clinical approach” and the “basic research approach”. The aim of the clinical approach was to find out if the reduction of the device delay mismatch can be performed within the MAESTRO 9 software based on the manufacturers' HA delay values. This is especially interesting for CI audiologists who do not have access to measurement devices to determine HA delay themselves. In the basic research approach, we wanted to investigate interactions between directionality and device delay mismatch compensation. In this approach we delayed the CI stimulation by the measured HA delay and not by the manufacturers’ HA delay values. The fitting for the subjects CI and HA is described in detail the following passages. An overview of all study dates, conditions and utilized fitting can be found in Table 5.2. In the following all conditions within the A-B-B-A tests in the clinical approach are denoted by the subscript notation “C” e.g., A_C. And with for the basic research condition the subscript notation “B” is used (e.g., A_B).

Table 5.2: Overview of the experimental procedure for all four study dates (CI = cochlear implant; HA = hearing aid).

Approach	Study date	Condition	CI/HA directionality	Programmed CI Delay	Tests performed
Clinical approach	1st	A_C	natural/ default settings	None (= 1.5 ms)	Sound localization Speech in Noise
	1st	B_C	natural/ default settings	Manufacturer delay for subjects HA/ HA delay measured (only used for sound localization)	Sound localization Speech in Noise
	3-4 weeks familiarization				
	2nd	B_C	natural/ default settings	Manufacturer delay for subjects HA	Sound localization Speech in Noise
	2nd	A_C	natural/ default settings	None (= 1.5 ms)	Sound localization Speech in Noise
Basic research approach	3rd	A_B	Omnidirectional/ omnidirectional	None (= 1.5 ms)	Sound localization Speech in Noise
	3rd	B_B	Omnidirectional/ omnidirectional	HA delay measured / Manufacturer delay for subjects HA (only used for sound localization)	Sound localization Speech in Noise
	3-4 weeks familiarization				
	4th	B_B	Omnidirectional/ omnidirectional	HA delay measured	Sound localization Speech in Noise
	4th	A_B	Omnidirectional/ omnidirectional	None (= 1.5 ms)	Sound localization Speech in Noise

5.2.3 CI fitting

On the first study date, all subjects received a MED-EL SONNET 2 speech processor as an experimental device for the entire duration of the study. This processor in combination with the fitting software MAESTRO 9 enables an adjustable delay of the CI stimulation of up to 20 ms with a 0.1 ms resolution within its fitting software. This was true for all subjects except subject Bim202 who already owned a SONNET 2 but did not use the adjustable delay before the study. In MAESTRO 9 delay of the CI stimulation can be activated by adding a HA on the contralateral side to the CI and typing in a specific HA delay by which the CI stimulation should be delayed. After initial fitting we asked the subjects to adjust the loudness of the study processor to their own processor via their remote control. In each fitting session, subjects were asked to adjust the perceived loudness

so that it matched with the loudness of the prior fitting. Within the clinical approach of the study, the subjects used their everyday map with their desired preprocessing enabled (e.g., wind reduction etc.). In cases where the subjects had adaptive beamforming enabled, the directionality was changed to "natural" within the fitting software, mimicking the directionality of a human pinna. For the initial test the adjustable delay of the CI speech processor was set to the minimum value of 1.5 ms. For the testing with the reduced device delay mismatch manufacturer delays were used for the subjects' HA which can be found online provided by MED-EL. No further alterations to the CI fitting were performed. After the second B condition, the CI delay was acutely reset to 1.5 ms.
For the basic research approach all preprocessing was set to mild and the directionality of the CI was set to omnidirectional within the MAESTRO 9 fitting software to be able to investigate the effects of directionality settings on sound source localization with a reduced device delay mismatch. This programming of directionality and pre-processing was performed at the end of the second study date, to allow for familiarization of three weeks before performing further testing. The delay used for the correction of the device delay mismatch was the HA delay which was measured for each HA using a measurement setup as described in chapter 3.

5.2.4 HA fitting

All subjects used their own HA throughout the study. On the first study date, HA delays were measured for each subject. Within the clinical approach no changes were made to the subjects HA directionality settings. For the basic research approach, subjects were asked to have their HAs fitted to an omnidirectional directionality setting by their hearing aid acoustician immediately after the second study date. They therefore had at least 21 days of familiarization time to the changed HA directionality before the third study date. No further changes in preprocessing were made on the HA. An overview of the subjects HAs with their respective delays given by the manufacturer and the delays measured in the lab can be found in Table 5.3. The HA delay given by the manufacturers were on average 5.2 ± 2.1 ms with a maximum delay of 8.1 ms for Bim213 and a minimum delay of 2 ms for Bim203 and Bim205. The HA delays measured in the laboratory via a self-designed measurement setup averaged at 5.2 ± 2.4 ms with a maximum delay of 7.8 ms for Bim202 and a minimum delay of 2.8 ms for subjects Bim203, Bim205 and Bim206, all using HAs from Widex. The mean absolute difference between manufacturer given delay and measured HA delay was 0.68 ms with a maximum difference of 1 ms for Bim213 and a minimum difference of 0 ms for Bim201.

Table 5.3**:** Hearing aid types and processing delays given by manufacturers and measured during the study.

Subject	**HA type**	**HA processing delay provided by the Manufacturer [ms]**	**HA processing delay measured [ms]**	**Absolute difference [ms]**
Bim201	ReSound LiNX2 LS9	5	5	0
Bim202	Oticon Xceed 2UP	8	7.8	0.2
Bim203	Widex ENJOY50 FM	2	2.8	0.8
Bim204	Oticon Ruby 2 PP	8	7.5	0.5
Bim205	Widex Daily50 D-FA	2	2.8	0.8
Bim206	Widex Beyond 330 B3-F2	2.9	2.8	0.1
Bim209	Phonak Naida Q90 UP	6.1	7.2	1.1
Bim210	Audio Service SUN	6.2	4	2.2
Bim211	Phonak Audeo V90-13 RIC	6.1	6.3	0.2
Bim212	Widex Unique Fs 330	2.9	3.5	0.6
Bim213	Phonak Naida B90 UP	8.1	7.1	1

5.2.5 Test environment

All testing was conducted in an audiometric booth (IAC Acoustics, Niederkrüchten, Germany). Subjects were seated in the center of a loudspeaker circle consisting of twelve loudspeakers (type Genelec 8030C) with an angular spacing of 30° and a diameter of 2 meters. The height of the loudspeakers was 1.15 m, approximately aligning the center of the loudspeaker membranes with the ears of the seated subjects. Loudspeakers were labeled clockwise with numbers between 1 and 12 with speaker number 1 representing -90° azimuth. All stimuli were presented via a RME Fireface 802 soundcard (Audio AG, Haimhausen, Germany) which was connected to a measurement PC outside the audiometric booth. Answers were captured using a tablet computer and sent via Bluetooth to a PC outside the audiometric booth for processing in MATLAB.

5.2.6 Sound localization

Sound localization accuracy was measured in the frontal horizontal hemisphere using the seven loudspeakers between -90° and 90° azimuth with an angular spacing of 30° (loudspeaker 1 to loudspeaker 7). Stimuli consisted of five noise bursts (125 Hz to 20 kHz). Each burst had a duration of 70 ms with 3 ms gaussian-shaped slopes and pauses of 30 ms between each burst as described by Seeber et al. (2004) and as used in a previous study (see chapter 3). Spectral roving with two different spectral characteristics and level roving (60 dB, 65 dB and 70 dB) was applied to the stimuli in each trial to minimize the usage of monaural cues according to Van de Heyning et al. (2016). Stimulus generation was performed in MATLAB. Each localization test consisted of a total of 84 stimuli, presenting each combination of spectrum, level, and speaker position twice (2 x 2 x 3 x 7) in random order. Subjects had a visual representation of the seven loudspeakers with their respective speaker numbers in the frontal horizontal hemisphere on the tablet computer and responded to stimuli by clicking on the speaker that they assumed the stimulus originating from. During testing no feedback was given to the participants whether their responses were correct or wrong. Prior to the localization tests participants completed a training run in which feedback of the correct source position was given via the tablet after subjects submitted their answers. Subjects were instructed to face the loudspeaker with the number 4 which was at 0° azimuth. For more details about the localization testing see chapter 3. Subjects performed one localization test for each delay condition. Additionally in conditions B_C and B_B acute localization tests with the manufacturer given HA delay and the measured HA delay were conducted to investigate differences between both delay values on sound source localization accuracy acutely irrespective of the interactions with the study approach.

5.2.7 Speech tests

To measure the effect of a reduced device delay mismatch on speech perception the Oldenburg sentence test (OlSa) was used. The OlSa is a german matrix sentence test with each sentence being made up of five words in the following manner: Name – verb – number – adjective - object. Sentences of the OlSa speech material are semi-nonsensical to avoid effects of guessing words based on the sentence's context. Speech reception thresholds (SRTs) were measured which is the signal-to-noise ratio (SNR) between speech signal and masking noise at which 50% of the speech material is understood correctly. Olnoise was used as a masker, generated from the speech material overlapped 30 times with random temporal shifts creating a broadband noise masker with the same long-term average speech spectrum as the speech material. Each test list consisted of 20 test sentences. Subjects gave their answers via a tablet computer displaying the word material of the OlSa. Subjects were allowed to guess or to leave specific words blank if they did not hear the word. All subjects absolved a training run to get used to the test and word material on each study date with no feedback given. To assess bimodal performance, we measured the effect of spatial release from masking (SRM), which is the difference in SRTs between speech signal and noise being spatially collocated vs when speech and noise being spatially separated. To measure SRM, SRTs were measured with speech and noise being spatially collocated at 0° azimuth (S_0N_0) and with the speech coming from 0° azimuth and the noise from either 90° or -90° azimuth, dependent whether subjects had their HA on the right or left ear (S_0N_{HA}). This spatial condition was chosen to assess whether the anticipated better ear listening on the CI side would be affected by the CI delay. Test and retest were measured for both spatial configurations yielding a total of four test lists per condition. Within these four test lists conditions were randomized and subjects were asked if they preferred a short break after completion of each list. SRM was calculated as the difference between the mean SRTs between test and retest measured at S_0N_0 and S_0N_{HA}.

5.2.8 Data analysis and statistical evaluation

To assess sound source localization accuracy, two metrics were calculated according to Yost et al., (2013): The root-mean-square (RMS) error, being a measurement of the precision of the subjects' judgements and the signed bias, representing the systematic error or trueness of the subjects' localization judgements. For the signed bias, the data was normalized. This was done by inverting the measured signed bias of subjects wearing the CI on their left ear. Following, a positive signed bias always represents a systematic error towards the CI of the subjects. Shapiro-Wilk tests were performed on all datasets to test for normality. Due to the consistent non-normal distribution of datapoints, non-parametric Friedman tests were used to test for differences between different conditions. In the case of significant outcomes post-hoc pairwise testing was applied via Wilcoxon signed-rank tests. To compare initial performance and improvements in sound source localization accuracy linear regression was performed. An alpha level of 0.05 was used to determine statistical significance. We performed all statistical testing in MATLAB.

5.3 Results

5.3.1 Sound localization

5.3.1.1 Overall sound localization accuracy

Sound localization accuracy in terms of RMS error and signed bias for both approaches can be found in Figure 5.2 and Figure 5.3. Friedman tests were conducted for both measures in both study approaches. For the clinical approach, a significant influence of condition was found for RMS error ($\chi2(3) = 13.69$, $p = 0.003$) and for the signed bias ($\chi2(3) = 13.36$, $p = 0.004$).

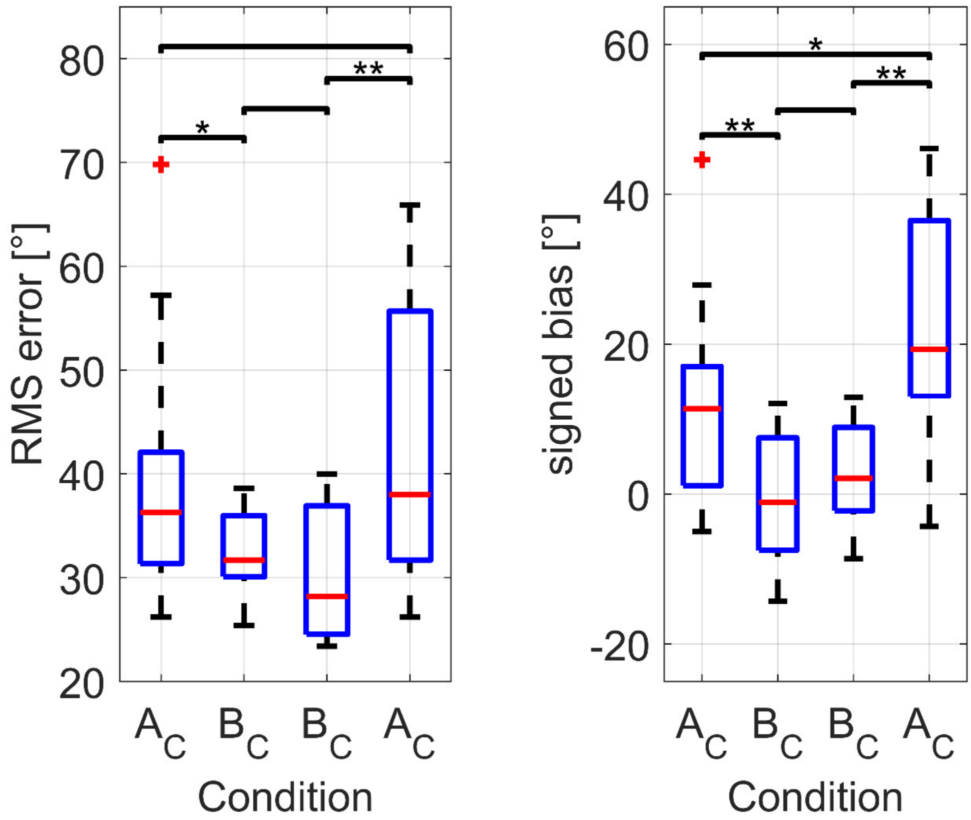

Figure 5.2**:** RMS error and signed bias between conditions for the clinical approach as boxplots (red line: median; box: 1st-3rd quartile; whiskers: minimum and maximum without outliers; outliers in red). Statistically significant differences denoted by * ($p<0.05$), ** ($p<0.01$) and *** ($p<0.001$).

Post-hoc pairwise Wicoxon signed-rank tests for the clinical approach revealed a significant improvement in sound localization accuracy between the first A_C and B_C condition in RMS error ($p = 0.04$) and signed bias ($p = 0.004$) indicating a significant improvement due to acute reduction of the device delay mismatch. After three weeks of familiarization no further significant difference could be found in RMS error and signed bias when comparing the first B_C and second B_C condition. When the CI delay was acutely set to its initial value of 1.5 ms a highly significant deterioration could be seen in RMS error (p = 0.005) and signed bias (p = 0.004). Between the first and second A_C condition the RMS error did not change significantly. However, signed bias was significantly worse in the second A_C condition (p = 0.03) compared to the first A_C condition. For the signed bias Wilcoxon signed-rank tests in the clinical approach revealed a significant difference from a zero median for the first (p < 0.001) and second A_C condition (p = 0.002), and no significant difference from zero in the first (p = 0.9) and second (p = 0.24) B_C condition.

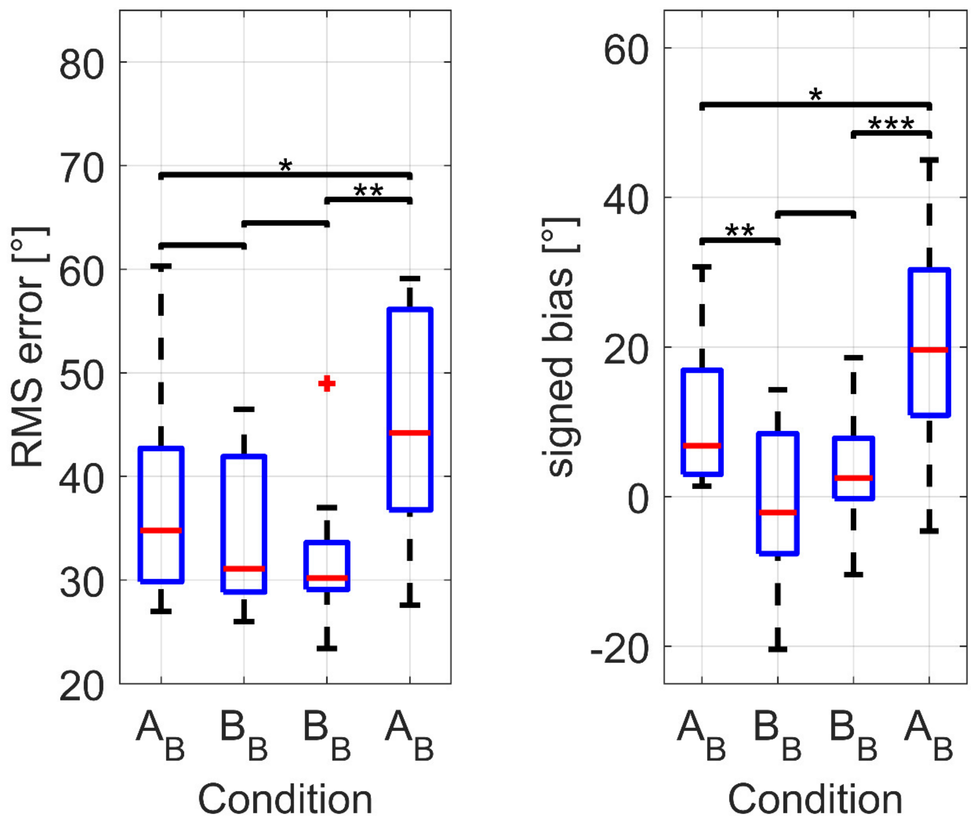

Figure 5.3: RMS error and signed bias between conditions for the basic research approach as boxplots (red line: median; box: 1st-3rd quartile; whiskers: minimum and maximum without outliers; outliers in red). Statistically significant differences denoted by * (p<0.05), ** (p<0.01) and *** (p<0.001).

In the basic research approach Friedman tests also revealed a significant influence of condition on RMS error ($\chi2(3) = 9.22$ p = 0.03) and signed bias ($\chi2(3) = 25.15$ p < 0.001). For the basic research approach, no significant improvement could be observed in RMS error, when the CI delay was set to the measured HA delay acutely. However signed bias improved significantly (p = 0.002) when testing acutely after device delay mismatch reduction. After three weeks of familiarization to the adjusted CI delay no significant differences in sound source localization accuracy could be observed. Between the second B_B and A_B condition significant deterioration of RMS error (p = 0.005) and signed bias (p

< 0.001) was observed. Between the initial A_B condition and the second A_B condition a significant deterioration could be seen in RMS error ($p = 0.04$) and signed bias ($p = 0.01$). For the signed bias Wilcoxon signed-rank tests in the clinical approach revealed a significant difference from a zero median for the first ($p = 0.014$) and second A_B condition ($p = 0.002$), thus indicating a significant bias towards the subjects CI. No significant difference from zero in the first ($p = 0.78$) and second ($p = 0.24$) B_B condition was observed.

5.3.1.2 Effects of directionality on baseline performance

To investigate the effect of HA/CI directionality on the baseline performance, Wilcoxon signed-rank tests were performed for RMS error and signed bias between the first A_C and the first A_B condition. Pairwise testing did not reveal a significant difference due to HA/CI directionality for RMS error ($p = 0.72$) and signed bias ($p = 0.85$). When comparing the second A_C and the second A_B condition, no significant difference in performance could be found in RMS error ($p = 0.72$) and in signed bias ($p = 0.56$). These results show that the microphone directionality did not influence the subjects sound localization accuracy in this study.

5.3.1.3 Effects of delay compensation method

In both the clinical approach and the basic research approach we investigated the effect of the delay compensation method acutely within the first B_C and B_B testing. This allowed us to test for acute performance differences in sound source localization accuracy within the same approach. Thus, only having the delay setting as the independent variable. Within the clinical approach a significant difference between delaying the CI stimulation by the manufacturer given delay and the delay measured in the lab could be found for RMS error ($p = 0.01$) with a mean RMS of $32.4 \pm 4.1°$ for the manufacturer delay and 35.2 ± 6 ° for the measured HA delay. For the signed bias no significant difference between the two delays could be found ($p = 0.06$). In the basic research approach also, no significant difference could be found for RMS error ($p = 0.43$) and for the signed bias ($p = 0.1$). Thus, within both approaches the delay chosen to compensate for the device delay mismatch did not consistently significantly influence localization accuracy, which is probably due to the relatively small difference between both delays (0.7 ± 0.6 ms).

5.3.1.4 Effects of initial localization performance and HA delay

Investigating the question whether the HA delay and thus the magnitude of the device delay mismatch played a role in initial sound source localization performance, linear regression was performed between HA delay and the RMS error and signed bias in the first A condition for both approaches.

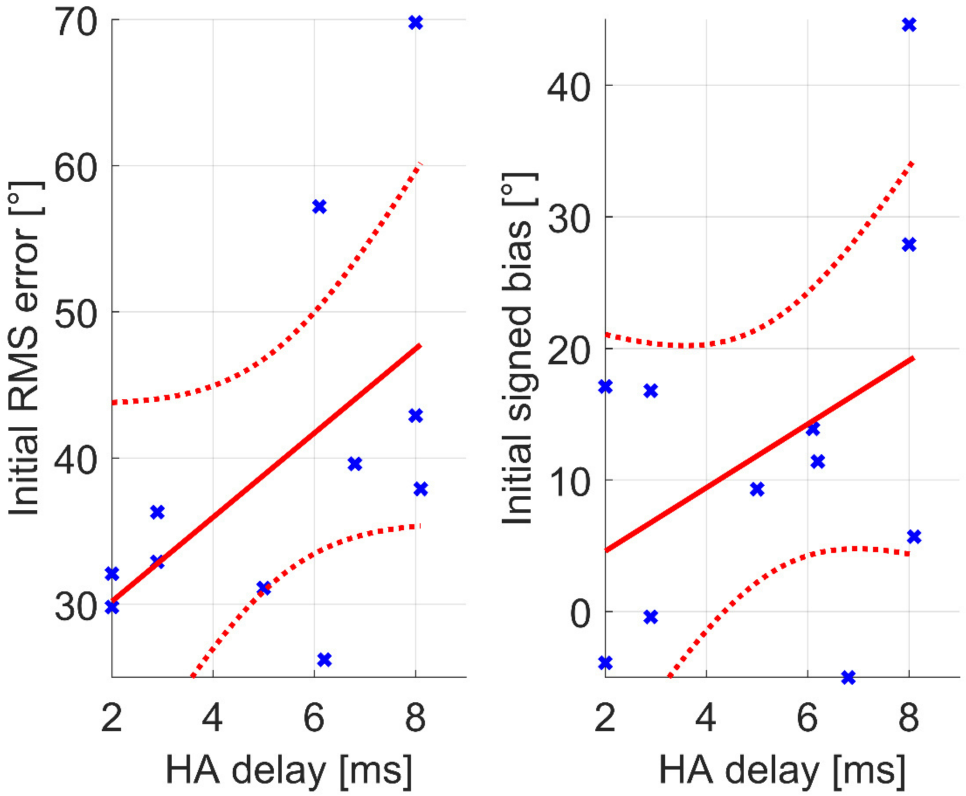

Figure 5.4**:** Linear regression between initial performance in RMS error and signed bias and HA delay in the clinical approach. Data in blue, regression in red (dotted lines represent the confidence bounds of the linear model).

In the clinical approach, no significant correlation could be found between HA delay and initial performance in RMS error ($r = 0.54$; $p = 0.09$) and signed bias ($r = 0.41$; $p = 0.22$). This means that the initial performance after familiarization to a device delay mismatch was not significantly correlated to the magnitude of this mismatch in our cohort.

Influence of longer familiarization periods on the effects of device delay mismatch reduction in bimodal subjects

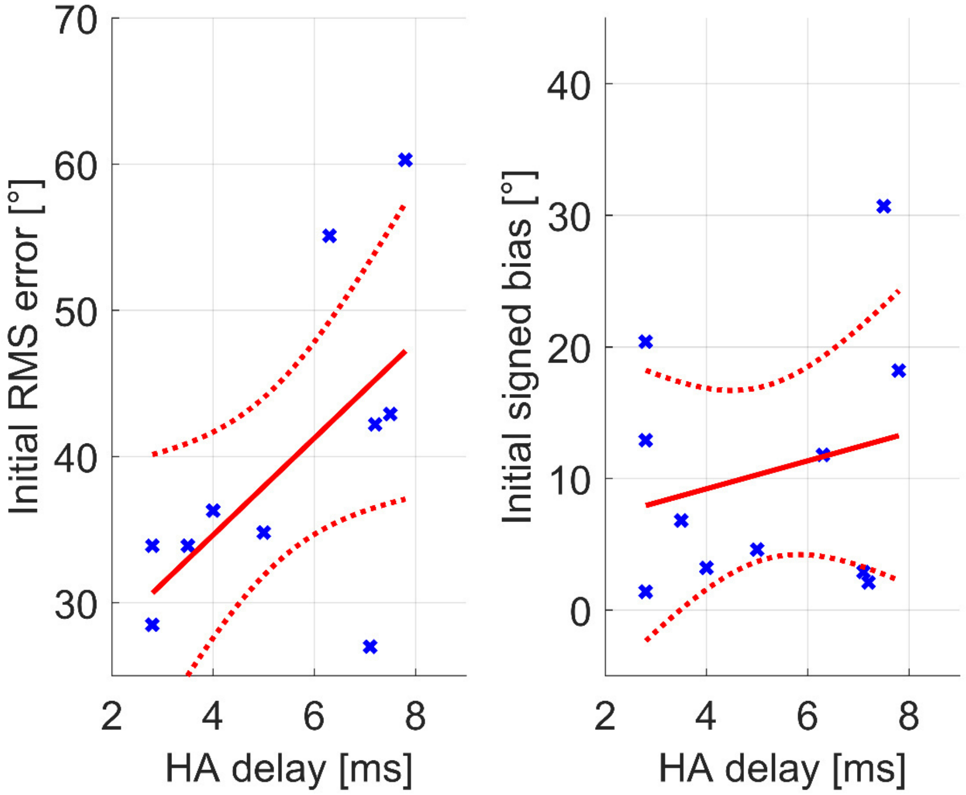

Figure 5.5: Linear regression between initial performance in RMS error and signed bias and HA delay in the basic research approach. Data in blue, regression in red (dotted lines represent the confidence bounds of the linear model).

For the basic research approach, the correlation between initial performance and RMS error was significant ($r = 0.63$; $p = 0.04$), but not for the signed bias ($r = 0.23$; $p = 0.5$).

Influence of longer familiarization periods on the effects of device delay mismatch reduction in bimodal subjects

5.3.1.5 Effects of compensated delay magnitude

To assess the effects of the magnitude of the corrected delay to the increase in performance in bimodal subjects, linear regression between HA delay and acute performance increase between the first A and B conditions for both approaches were performed. The results can be found in Figure 5.6 and Figure 5.7.

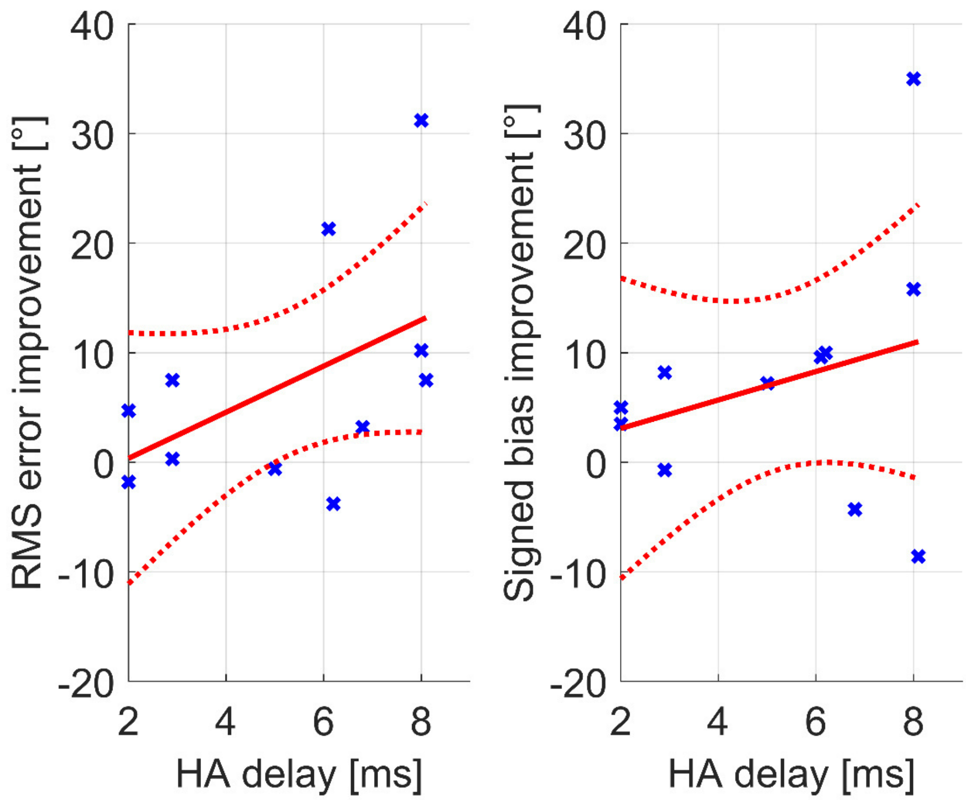

Figure 5.6: Linear regression between performance improvement due to device delay mismatch reduction in RMS error and signed bias and HA delay in the clinical approach. Data in blue, regression in red (dotted lines represent the confidence bounds of the linear model).

No significant correlation between HA delay and improvement in sound source localization in RMS error (r = 0.5; p = 0.13) or signed bias (r = 0.28; p = 0.41) could be found in the clinical approach.

Influence of longer familiarization periods on the effects of device delay mismatch reduction in bimodal subjects

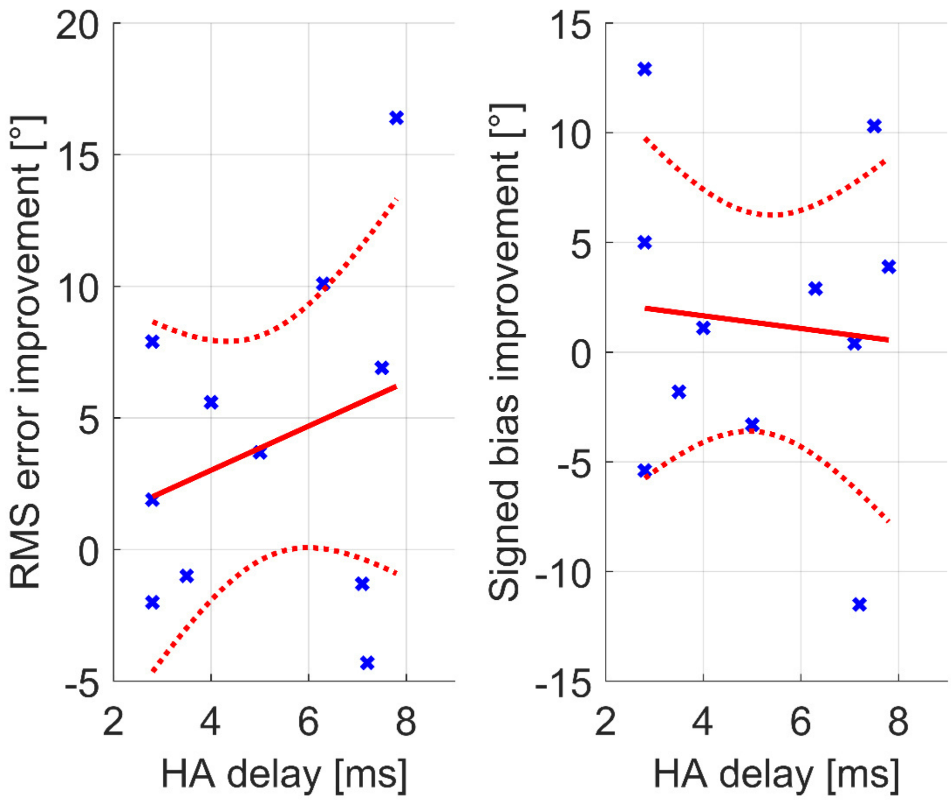

Figure 5.7: Linear regression between performance improvement due to device delay mismatch reduction in RMS error and signed bias and HA delay in the basic research approach. Data in blue, regression in red (dotted lines represent the confidence bounds of the linear model)

Also, in the basic research approach, linear regression did not reveal a significant correlation between HA delay and improvement in sound source localization accuracy in neither RMS error ($r = 0.28$; $p = 0.4$) nor signed bias ($r = 0.001$; $p = 0.97$).

Influence of longer familiarization periods on the effects of device delay mismatch reduction in bimodal subjects

5.3.1.6 Effects of initial performance

Figure 5.8 shows the linear regression between initial localization performance and improvement after acutely reducing the device delay mismatch. For the RMS error, initial performance and improvement of RMS error by device delay mismatch reduction correlated highly significantly ($r = 0.96$, $p < 0.001$). Also, for the signed bias, the correlation was highly significant ($r = 0.89$; $p < 0.001$).

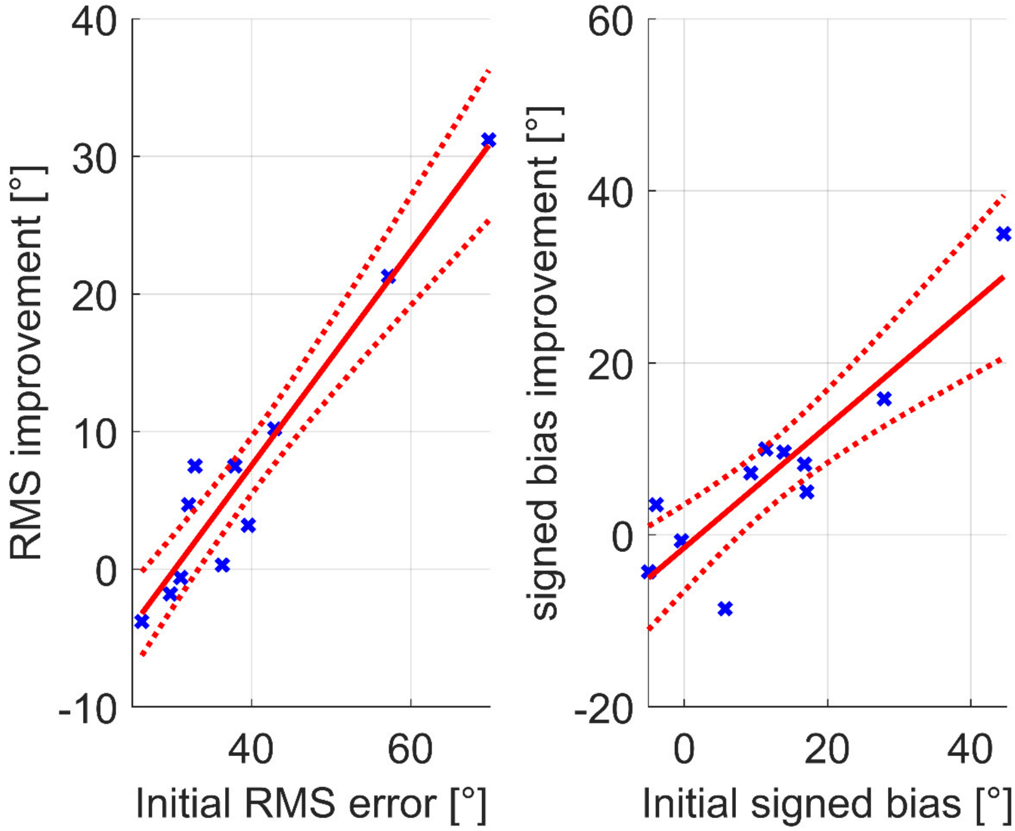

Figure 5.8: Linear regression between initial performance and acute improvement for RMS error and signed bias in the clinical approach. Data in blue, regression in red (dotted lines represent the confidence bounds of the linear model).

In the basic research approach (Figure 5.9), RMS improvement also correlated highly significantly with baseline RMS error ($r = 0.75$; $p = 0.008$). The same correlation was found between baseline signed bias and its improvement ($r = 0.83$; $p = 0.001$). It must be noted that the improvement in signed bias was calculated as the decrease in distance to a signed bias of 0° since this is the optimal case. This was calculated as the difference in absolute signed bias of the first A and B condition. These correlations revealed that subjects performing worse initially benefitted more from the reduction of the device delay mismatch than the subjects with good sound source localization accuracy without device delay mismatch compensation.

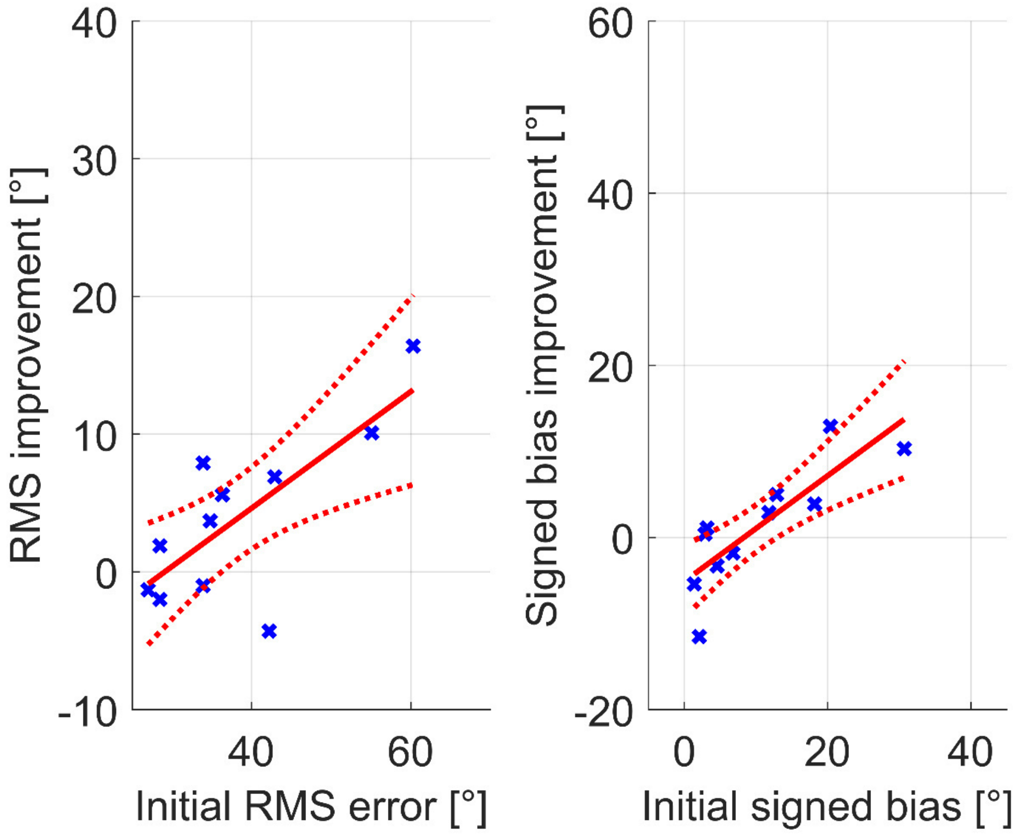

Figure 5.9: Linear regression between initial performance and acute improvement for RMS error and signed bias in the basic research approach. Data in blue, regression in red (dotted lines represent the confidence bounds of the linear model).

5.3.1.7 Effects of acute desynchronization of CI an HA stimulation versus HA delay

Since a highly significant deterioration between the first and second A condition and between the second B and second A condition in both approaches could be observed, linear regression between the HA delay and the deterioration between the second B and second A condition in both approaches was performed. This addressed the question whether this acute decline in sound source localization accuracy was related to the magnitude of device delay mismatch. This could not be proven before, when subjects were already familiarized to the device delay mismatch.

Figure 5.10 shows the linear regression between HA delay and performance decline for RMS error and signed bias for the clinical approach of this study. Note that for the following figures a positive decline means a higher decline between B and A.

Influence of longer familiarization periods on the effects of device delay mismatch reduction in bimodal subjects

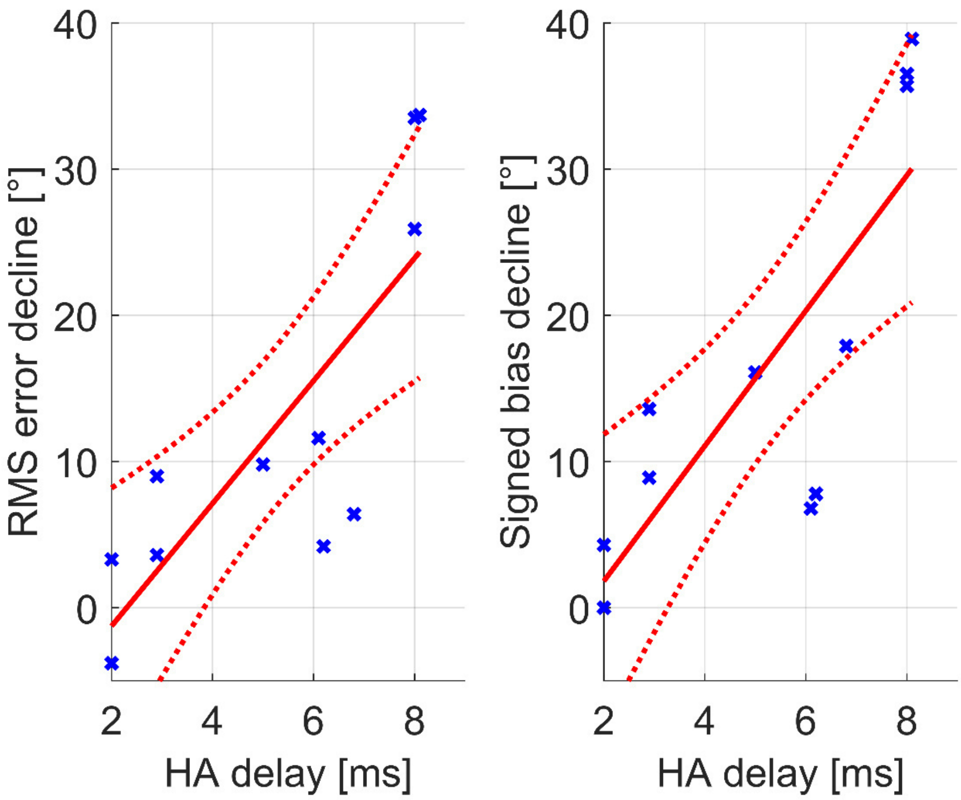

Figure 5.10: Linear regression between HA delay and acute performance decline for RMS error and signed bias in the clinical approach. Data in blue, regression in red (dotted lines represent the confidence bounds of the linear model).

There was a significant correlation for RMS error decline and HA delay (r = 0.8; p = 0.003) and between signed bias decline and HA delay (r = 0.81; p = 0.002). For the basic research approach linear regression results between HA delay and decline in RMS error and signed bias are shown in Figure 5.11.

In the basic research condition, the HA delay and RMS error decline showed a significant positive correlation with a Pearson's correlation coefficient of r = 0.71 and a p-value of p = 0.02. For the signed bias decline, correlation with the HA delay did not reach significant correlation (r = 0.55; p = 0.08).

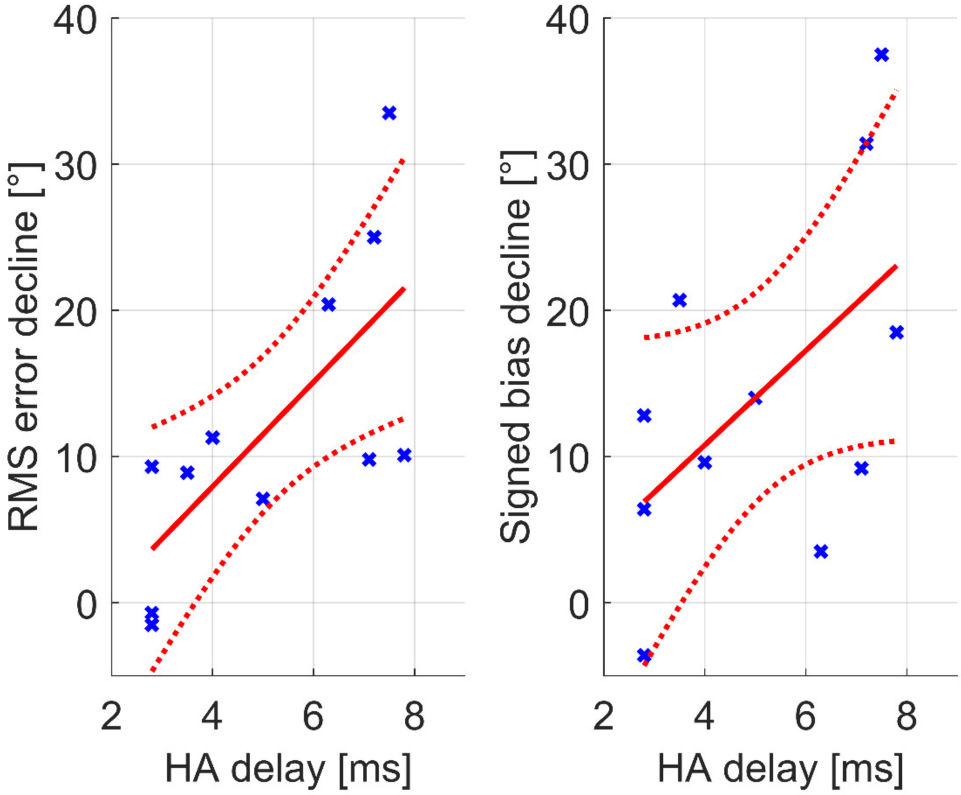

Figure 5.11: Linear regression between HA delay and acute performance decline for RMS error and signed bias in the basic research approach. Data in blue, regression in red (dotted lines represent the confidence bounds of the linear model)

5.3.2 Speech tests

5.3.2.1 Spatial release from masking

Figure 5.12 shows the measured SRM for each delay condition for the clinical approach as well as for the basic research approach. In each combination of device delay mismatch and study approach, subjects showed SRM significantly different from zero. Friedman tests showed no significant difference in SRM due to programmed device delay in the clinical approach ($\chi2(3) = 4.42$; $p = 0.2$) and in the basic research approach ($\chi2(3) = 7.04$; $p = 0.07$). Furthermore, by pairwise comparison via Wilcoxon signed-rank test of the first A conditions of the clinical approach vs the basic approach, a significant difference could be found ($p < 0.001$). Since no changes to the device delay mismatch were made in these conditions, this difference was an effect of the differences in HA/CI directionality.

Influence of longer familiarization periods on the effects of device delay mismatch reduction in bimodal subjects

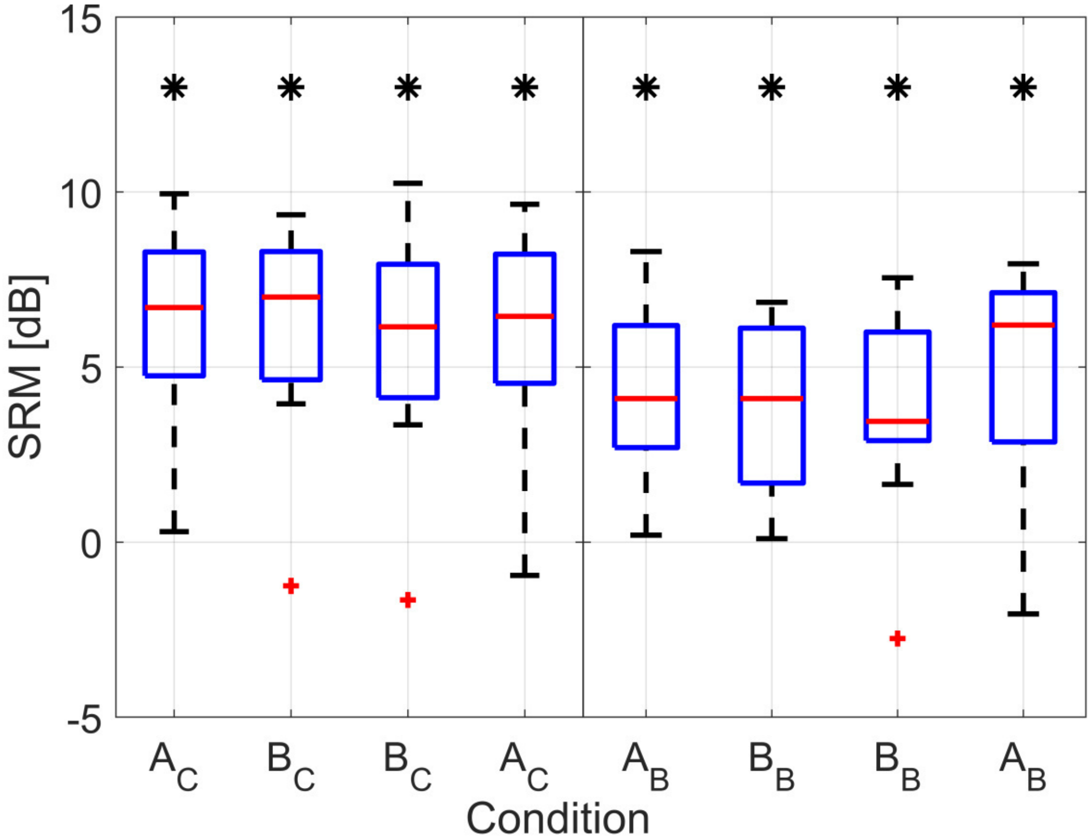

Figure 5.12: Spatial release from masking (SRM) for all conditions in the clinical approach and in the basic research approach as boxplots (red line: median; box: 1st-3rd quartile; whiskers: minimum and maximum without outliers; outliers in red). Stars above boxes indicate significant difference from a distribution with a median of 0° azimuth (p<0.05).

5.4 Discussion

The presented study investigated the effect of compensation of a mismatch in stimulation timing, a so-called device delay mismatch, in bimodal CI/HA users using MED-EL CIs. The primary goal of this study was to investigate, whether longer familiarization periods to changes in device delay mismatch as long as three weeks lead to further improvement in sound source localization accuracy than a shorter time span (1 hr) as reported in chapter 3. As MED-EL has implemented the possibility to delay the CI stimulation in their latest generation of speech processors via the fitting software, such a familiarization was possible.

The final question of this study was, if this new feature can be readily used by CI fitting specialists without access to a measurement setup to quantify HA delays by using the HA delays given by the manufacturers and without further alteration to the CI fitting used by the patients. For this reason, the study was split into two approaches, the clinical approach, and the basic research approach. The clinical approach used manufacturer values for the HA delay and no changes to the individual CI pre-processing settings were made. The basic research approach used the measured HA delays and the same CI pre-processing settings were used for the subjects.

From the current data, several questions concerning the use of device delay mismatch compensation by delaying CI stimulation in bimodal MED-EL CI users can be addressed.

5.4.1 Sound localization

The overall results for sound source localization accuracy for both the clinical condition and the basic research condition were well in line with previous findings reported in chapter 3. A significant improvement in sound localization accuracy occurred in signed bias and RMS error (but only in the clinical condition) when the device delay mismatch was reduced acutely. While previous studies only applied familiarization periods of an hour, the non-significant difference in signed bias and RMS error between the first and second B conditions proved that the positive effects of device delay mismatch reduction are in fact acute and do not improve further after familiarization of three weeks. For the signed bias, reduction of the device delay mismatch through delayed CI stimulation led to test results that did not differ significantly from a distribution with an “optimal” signed bias (i.e., 0°) as its median, as Wilcoxon signed-rank tests revealed. As in previous studies, acute reintroduction of a device delay mismatch led to highly significant deterioration of localization accuracy in terms of signed bias and RMS error. An interesting difference to the previous study with short familiarization times was the significant difference between the two conditions without reduction of a device delay mismatch (first and second A condition), which could be seen in the signed bias for the clinical approach and in signed bias and RMS error in the basic research approach. Such a difference could not be reported in chapter 3. This is probably due to the longer familiarization to a device delay mismatch. Within the first A conditions, subjects had the time to familiarize to their new setting for at least 3 weeks. When acutely switching from reduced to non-reduced device delay mismatch in the second A condition the subjects were familiarized to the optimized stimulation timing and thus performed significantly worse. This effect probably did not occur in the previous study since only an hour of familiarization was applied, which was not sufficient for the subjects to "unlearn" the compensation mechanisms for their device delay mismatch. This effect will be discussed further in the following.

The best median performance in our subject group with an RMS error around 30° was well in line with best localization performance without compensation of the device delay mismatch seen in other studies investigating sound localization in bimodal listeners (Ching et al., 2004; Dunn, Perreau, Gantz, & Tyler, 2010; Dunn, Tyler, & Witt, 2005; Veugen, Hendrikse, et al., 2016).

An additional finding was, that CI/HA directionality between both study approaches did not significantly influence localization accuracy. This is in line with findings by Dorman et al. (2018), who investigated the influence of CI directionality algorithms and could not show a significant difference in RMS error for between omnidirectional and “natural” (i.e., mimicking the directionality characteristics of the human pinna) directionality settings for bilateral users of MED-EL CIs.

By testing both delay methods acutely in the first B_C and B_B condition, the effects of the selected delay within the same directionality setting were investigated. Only within the clinical approach, RMS errors differed significantly but the absolute difference was rather low with a mean difference of 2.8°. Comparisons of signed bias and RMS error in the basic research approach did not reveal significant differences. Combined with the findings that the HA delays measured in the lab and the HA delays provided by the HA manufacturers differed minimally, this showed that the clinical approach is feasible way for CI technicians to reduce the device delay mismatch and improve sound localization performance.

Linear regression between initial localization accuracy showed no significant correlation with the HA delay and thus with device delay mismatch between CI and HA stimulation (only a significant correlation was found for the RMS error in the basic research condition). This might indicate that bimodal subjects are able to compensate for a device delay mismatch to some extent, and this compensation seems to not be dependent on the magnitude of this mismatch. Further evidence for this assumption came from correlating the HA delay with the initial improvement in sound localization accuracy. In neither the clinical approach nor in the basic research approach correlations between performance improvement after acute device delay mismatch reduction and the magnitude of the device delay mismatch could be observed. These findings might have their roots in a somewhat limited ability to compensate the effects of a device delay mismatch and a lower precision of sound localization in bimodal listeners, with both these boundaries limiting the range of localization accuracy in bimodal subjects. Limited ability of the auditory system to compensate for delay offsets between both ears over the course of several days has been shown by Trapeau & Schönwiesner (2015) in normal-hearing subjects. When a static delay of 625 µs was introduced to one ear via a constantly worn earplug, the mean signed error in sound localization of the subjects acutely deteriorated from 0° to ~25° towards the ear provided with the earplug. Interestingly, after one day of familiarization mean signed errors got better by roughly 12°. On the following days of testing no further improvements due to familiarization were reported. This ceiling effect of plasticity might also be seen in the bimodal subjects in this study and seems not directly connected to the magnitude of the device delay mismatch. Further evidence of a limitation of this plasticity could be seen in the signed bias being significantly different from zero in the initial A_C and A_B conditions and localization judgements being biased towards the faster modality. With the improvement in localization accuracy not being correlated with the magnitude of reduced device delay mismatch, this might have its roots in an upper precision limit for sound source localization in bimodal subjects of around 30° RMS error for best performances (Ching et al., 2004; Dunn et al., 2010, 2005; Veugen, Hendrikse, et al., 2016). Only one subject in one study by Seeber et al. (2004) has been reported to localize almost as precise as normal-hearing listeners but can be considered an outlier in the broader context of the literature. This limited localization precision is probably due to the lack of spectral overlap in bimodal subjects, with the CI not stimulating low frequencies sufficiently and the hearing loss on the HA side in the high frequency regions further limiting access to ILD cues and also missing ITD cues, which dominate sound localization in normal-hearing listeners (Seeber et al., 2004).

When comparing baseline performance in sound source localization and acute improvement due to device delay mismatch reduction highly significant correlations could be found for the RMS error and signed bias in both the clinical approach and the basic research approach. This means that in our sample of subjects, bimodal CI/HA users with worse sound source localization accuracy profited more from the compensation of the device delay mismatch. This might also be connected to the upper performance limit in terms of RMS error in bimodal subjects discussed above. When subjects were already close to the limit of achievable localization accuracy this might limit further improvement through device delay mismatch reduction.

When the reduction of the device delay mismatch was removed acutely a significant correlation between HA delay and thus magnitude of device delay mismatch and decline in sound localization accuracy could be observed for both the clinical as well as the basic research approach. This further supports the hypothesis, that the non-significant correlation between baseline performance and benefit through device delay mismatch reduction has its roots in familiarization to desynchronization between CI and HA. When subjects were used to a reduced device delay mismatch a higher magnitude of device delay mismatch did mean a higher decay in sound localization accuracy. This implies that bimodal subjects can familiarize to changes in device delay mismatch to some extent.

To further understand the familiarization processes to a device delay mismatch studies with shorter time intervals between tests could be conducted with possibly daily testing of sound source localization accuracy over the course of several days as done by Trapeau & Schönwiesner (2015)

5.4.2 Speech tests

SRM was assessed by comparing SRTs between two spatial conditions with speech and noise being collocated at 0° azimuth (S_0N_0) and with speech coming from the front and noise coming from the +/- 90° azimuth on the HA ear (S_0N_{HA}).
For SRM, no significant effect of device delay mismatch could be found for both approaches, whereas this group of subjects did show significant SRM for all tested conditions. The reason for this might be that bimodal subjects do not utilize binaural cues when it comes to SRM, but rather monaural head shadow cues. This is well in line with previous studies reporting SRM in bimodal HA/CI subjects being only moderated by monaural head shadow and not by binaural unmasking based on ITD (Dieudonné & Francart, 2020; Williges et al., 2019). Only if sufficient ITD coding and sensitivity could be achieved in bimodal subjects in the future, a device delay mismatch might impair SRM. A significant difference in SRM was found when comparing both approaches in the initial condition, thus only comparing effects of microphone directionality on speech understanding. Subjects performed worse when CI and HA microphones were set to an omnidirectional setting. This is not surprising given the attenuation through the pinna mimicking directionality for signals coming from +/-90° azimuth, as it was the case for our noise masker in the S_0N_{HA} condition, compared to no directional attenuation in the omnidirectional setting resulting in a lower SNR on the ear contralateral to the noise source. Similar findings on the influence of directional microphones for bilateral users of MED-EL CI, between omnidirectional and natural microphone directionalities have been reported in the past by Dorman et al. (2018).

5.5 Conclusions

From the presented data, several conclusions can be drawn on the influence of a reduced device delay mismatch in bimodal CI/HA users.

The first conclusion arising from the comparison of the clinical approach and the basic research approach, is that positive results achieved do not rely on experimental conditions considering microphone directionality. It can be achieved with the subjects everyday fitting, using manufacturer-provided HA delays without the need to measure HA delays individually. This is most relevant to the broader use of the reduction of device delay mismatch as a fitting parameter in one manufacturer's current CI programming software. Further, we conclude that the positive effects seen in sound localization accuracy are in fact acute and do not further improve after familiarization.
The magnitude of HA delay shows no direct correlation with sound localization accuracy when the device delay mismatch is present, nor on improvement after reduction of device delay mismatch. This is thought to have its roots in familiarization to the device delay mismatch and an upper performance level in bimodal sound source localization accuracy, independent of device delay mismatch.
Even with improvements in RMS error being limited to some constraints, the signed bias

in bimodal subjects can be reduced to roughly 0°. This means that a reduction of a device delay mismatch can eliminate the systematic shift in localization judgements towards the faster device and effectively re-center the perceived origin locations of sounds.
Without prior familiarization, detrimental effects of a device delay mismatch on sound localization correlated significantly with the magnitude of the device delay mismatch.
SRM was not affected by changes in device delay mismatch, which indicates that bimodal subjects rely on monaural better ear listening in this task. As soon as future CI sound coding strategies allow for the access to ITD cues for this task the effect of a device delay mismatch on spatial unmasking should be reinvestigated. Finally, bimodal subjects with poor localization performance benefit most from the reduction of a device delay mismatch.

6

Summary and Conclusions

The aim of this thesis was to investigate the effects of static interaural offsets in stimulation timing on binaural hearing. Such offsets often occur in hearing-impaired individuals who have an asymmetrical hearing loss and therefore require asymmetrical treatment of that hearing loss. In this thesis, a two-pronged approach was taken to investigate the effects of static interaural offsets in stimulation timing. Improvements on binaural hearing through reduction of this offset were measured in bimodal CI/HA users to validate the clinical relevance of a compensation of delay mismatches. Further, the effect of static interaural mismatches in stimulation timing was investigated in normal-hearing subjects. Since the latter subject group is more homogenous than hearing-impaired listeners, generalizations on the effects of temporal mismatches on binaural hearing are easier to make even with a small group size. In the following, a brief overview over the most important findings presented in the chapters of this thesis is given, followed by a comprehensive discussion connecting the presented results and its limitations. Finally, conclusions are drawn and a brief outlook on what future work on this topic could contain is given.

In the third chapter of this thesis, the effects of a reduced device delay mismatch were investigated in an A-B-B-A test design in nine adult bimodal CI/HA users. Within this design, sound localization accuracy was measured without a reduction of device delay mismatch (A) and with an acutely reduced device delay mismatch (B). Further, effects of a one-hour familiarization period on localization accuracy was measured (second B) and acute effects after reintroduction of a device delay mismatch (second A). The device delay mismatch reduction was performed via a battery-powered portable delay line (DL) based on a microcontroller worn around the neck of the subjects that delayed the CI stimulation by the HA processing latency (τ_{HA}). Further, $\tau_{HA} \pm 1$ ms was used as a CI delay to assess whether a temporal match of low- or high-frequency regions of the CI stimulation yielded a better sound localization accuracy (see Figure 3.1). The results showed that a reduction of the device delay mismatch in bimodal listeners led to a significant improvement in localization accuracy in terms of RMS error and singed bias. This improvement measured directly after reducing the device delay mismatch did not further improve after an hour of familiarization. When reintroducing the device delay mismatch, subjects acutely performed significantly worse again in the sound localization tests. Due to the non-randomizable study design, this was an important finding to clarify that the poor results measured within the first A condition were not due to a lack of training to the localization task. The three different delays additionally applied in a subgroup of seven subjects revealed that $\tau_{HA} + 1$ ms seemed to yield the best localization results but no statistical significance could be reported since the three results with τ_{HA}, τ_{HA} -1 ms and τ_{HA} +1 only varied minimally. This chapter addressed the shortcomings of a previous study by Zirn et al. (2019) by also evaluating the bias of the subjects' judgements that showed a clear shift of localization judgements towards the CI, which was the faster modality. This shift was reduced by the reduction of the device delay mismatch. However, the familiarization period one hour was rather short in this experiment. Furthermore, effects of magnitude of device delay mismatches on spatial hearing and interactions of the device delay mismatch reduction with the usage of directional microphone settings were not investigated.

In the fourth chapter of this thesis, the effects of a reference ITD was investigated on speech perception. Humans have the exceptional ability to use spatial hearing to improve SRTs, when the spatial positions of the target and the masker differ. However, this effect was not investigated in the presence of a reference ITD so far. In this chapter, spatial release from masking (SRM) was measured in the presence of a reference ITD of up to 7 ms in ten normal-hearing subjects. Since it is of great interest how different binaural cues and their processing are affected by a reference ITD, this effect was investigated with only offering ITD, ILD or with both cues combined via manipulation of the head-related impulse responses (HRIRs). Further, a blind equalization-cancelation model was applied to investigate the accuracy of such a model in predicting SRM in the presence of a reference ITD and to extend the results obtained experimentally. The experimental results show a significant influence of a reference ITD on spatial release from masking when both ITD and ILD are present in the signal. This behavior is shown to be driven by ITD based SRM being negatively affected by a reference ITD whereas ILD based SRM did not diminish due to reference ITD, because monaural head shadow dominates the effect in this task. The experimental findings could be accurately modelled. To assess, whether these effects are as drastic in a more ecologically valid environment, modeling was also performed in a virtual environment with reverberation present. With reverberation, the SRM through ITDs was generally lower as in an anechoic setting, since ITD cues are distorted by reverberation. Therefore, the detrimental effects of a reference ITD were not as pronounced as in an anechoic setting. SRM obtained by head shadow was not affected by a reference ITD in this setting. Limitations of this chapter are that such more ecologically valid room conditions were not tested with normal-hearing subjects and only modeled. Although, these modeled results are in line with the available literature on SRM in reverberation when no reference ITD is present.

In the fifth chapter, the effects of a device delay mismatch on sound localization in bimodal CI/HA users was investigated again, trying to answer remaining questions that could not be answered in chapter 3. These experiments were only possible after MED-EL had implemented a function to reduce the device delay mismatch in their latest generation of speech processors. With this new feature, familiarization periods of several weeks instead of one hour as with our self-constructed DL were possible. It was found that a reduction in device delay mismatch via the novel fitting parameter yielded similar improvements in sound source localization accuracy in bimodal listeners as were reported in chapter 3. Longer familiarization time did not yield further improvements in sound localization accuracy, confirming that the effects reported in chapter 3 are in fact acute in nature. Further, it was revealed that bimodal CI/HA subjects can familiarize to a device delay mismatch although neural plasticity was not able to fully reverse the detrimental effects on binaural hearing. In this familiarized state, the HA delay and thus the magnitude of device delay mismatch did not correlate with the localization accuracy of the subjects or with the improvements due to device delay mismatch reduction. It is hypothesized that this is due to familiarization in combination with a precision limit seen in bimodal subjects' sound localization accuracy. This limit probably has its roots in different factors than device delay mismatch. Such factors could be a frequency mismatch between both devices and the loss of fine structure ITDs through CI processing. Another connection between the findings of chapter 4 and chapter 5 was revealed when the SRM was investigated. No effect of device delay mismatch on SRM was found in chapter 5. The reason is that presumably the monaural head shadow effects dominate SRM in bimodal subjects. In chapter 4, it was shown that a reference ITD of several milliseconds did not diminish SRM when ILD and thus only head shadow cues were present in the stimuli in normal-hearing subjects. This solidifies the assumption that bimodal subjects purely rely on head shadow for SRM since they show no effect of device delay mismatch just as normal-hearing listeners when only using ILDs as reported in chapter 4.

To conclude, this thesis investigated and resolved a problem in the treatment of hearing loss, which is a static interaural offset in stimulation timing of several milliseconds.
The results of chapter 3 showed that the reduction of a device delay mismatch in bimodal CI/HA users leads to significant benefits in sound localization accuracy.
The results of chapter 4 showed that spatial unmasking is severely impacted by asymmetric stimulation timing. The usage of a verified modeling approach on the data showed, that a conventional model for SRM predicted outcomes accurately when a reference ITD of several milliseconds is present. This leads to the conclusion that such a model can be utilized to predict treatment outcomes in the future.
The relevance of the presented topic to fitting in bimodal subjects is underlined by one CI manufacturer, who has already implemented a device delay mismatch reduction in their fitting software. The first systematic study on this new feature presented in chapter 5 of this thesis provides valuable insights how this fitting parameter can be used in a clinical setting.
In summary, this thesis provided a solid basis for future investigations into mismatches present in bimodal CI/HA users. Especially interactions between device delay mismatch and frequency mismatch in bimodal subjects are highly interesting and could lead to further improvements of binaural effects in these subjects. Furthermore, measurements of SRM in more ecologically valid conditions in normal-hearing and bimodal subjects could give important insights on how asymmetries in stimulation timing affect spatial unmasking and how fitting parameters can be improved to further optimize speech understanding in real-life situations.

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