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The impact of bilateral gain reduction on localization and speech perception in spatially-separated noise

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THE IMPACT OF BILATERAL GAIN REDUCTION ON LOCALIZATION AND SPEECH PERCEPTION IN SPATIALLY-SEPARATED NOISE

by

Hua Ou

An Abstract

Of a thesis submitted in partial fulfillment of the requirements for the Doctor of Philosophy degree in Speech and Hearing Science in the Graduate College of The University of Iowa

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Thesis Supervisor: Professor Ruth A. Bentler

ABSTRACT

Bilaterally independent (mismatched) hearing aids cannot replicate the natural timing and level cues between ears, and hence, may result in negative consequences for localization and speech perception in spatially-separated noise performance.

Five gain reduction patterns were used to evaluate the impact of bilaterally mismatched gain reduction schemes on localization and speech perception performance in noise, compared to an unaltered bilaterally linear time-invariant amplification scheme (reference scheme), in which audibility was optimized. The bilaterally mismatched gain reduction schemes were later matched (synchronized) between ears to explore the possibility of restoring the deteriorated performance due to the mismatched schemes. Sound quality and listening-effort ratings among different gain reduction patterns were assessed, as well as the relationship between self-reported localization ability in daily life and measured localization performance in a laboratory setting.

Twenty-four bilateral hearing aid users were enrolled in this study and tested in a virtual environment with insert earphones. The results indicated that bilaterally mismatched gain reduction schemes had a negative impact on localization, compared to the reference scheme; whereas matching gain reduction schemes between ears improved the deteriorated localization performance. In contrast, the use of bilaterally mismatched gain reductions did not negatively impact the speech perception performance in noise. Matching the gain reduction scheme between ears actually resulted in reduced speech perception performance, compared to the mismatched gain reductions. Self-reported localization abilities were not found to be strongly related to the measured localization performance in this study. Finally, these five different gain reduction patterns did not result in significantly different overall sound quality ratings and listening-effort ratings for hearing aid users. However, the use of gain reductions (mismatched or matched) reduced the perceived noise intrusiveness, compared to the use of reference schemes.

It is unclear why there was a discrepancy between the results of the localization and speech perception performance in the present study. It is likely that hearing-impaired listeners do not use binaural cues in the localization task in the same manner as in the speech perception task.

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CERTIFICATE OF APPROVAL

PH.D. THESIS

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has been approved by the Examining Committee for the thesis requirement for the Doctor of Philosophy degree in Speech and Hearing Science at the December 2010 graduation.

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LIST OF ABBREVIATIONS

ANSI	American National Standards Institute
APHAB	Abbreviated Profile of Hearing Aid Benefit
AV	Aversiveness
BN	Background Noise
BTE	Behind-The-Ear
EC	Ease of Communication
HI	Hearing-impaired
HRTF	Head Related Transfer Function
IHAFF	Independent Hearing Aid Fitting Forum
ILD	Interaural Level Differences
ITC	In-The-Canal
ITD	Interaural Time Differences
ITE	In-The-Ear
ITU	International Telecommunication Union
KEMAR	Knowles Electronics Manikin for Acoustic Research
MATLAB	MATrix LABoratory
NH	Normal-hearing
RV	Reverberation
RAU	Rationalized Arcsine Units
SAS	Statistical Analysis System
SII	Speech Intelligibility Index
SNHL	Sensorineural Hearing Loss
SNR	Signal-to-Noise Ratio
SRM	Spatial Release from Masking

CHAPTER 1

INTRODUCTION

Individuals with hearing loss often experience problems understanding speech, especially in complex acoustic environments. Even though loss of audibility can explain most of the speech perception difficulties experienced by people with mild hearing losses, factors other than audibility have been implicated in those with moderate or greater hearing losses (Ching, Dillon, & Byrne, 1998; Hogan & Turner, 1998; Lorenzi, Gilbert, Carn, Garnier, & Moore, 2006; Moore, 2007; Pavlovic & Studebaker, 1984; Pavlovic, Studebaker, & Sherbecoe, 1986; Tyler, Summerfield, Wood, & Fernandes, 1982). The first factor, reduced spectral resolution (or frequency selectivity), has been studied extensively relative to its impact on speech perception for listeners with hearing loss. In short, the better the spectral resolution, the better the speech performance (e.g.,Henry, Turner, & Behrens, 2005). Broader auditory filters (i.e., reduced frequency selectivity) typically produce a more smoothed representation of the speech spectrum than normal auditory filters (Leek, Dorman, & Summerfield, 1987; Loizou & Poroy, 2001). Therefore, hearing-impaired people with reduced spectral resolution may not make full use of spectral feature differences as normally hearing people do. In addition, hearing-impaired listeners may experience more masking in noise than normal-hearing listeners for a comparable signal-to-noise ratio (SNR) due to reduced frequency selectivity. In fact, it has been found that hearing-impaired listeners require more favorable SNRs to better understand speech than do listeners with normal hearing (Bronkhorst & Plomp, 1989; 1992; Plomp, 1978). Furthermore, hearing-impaired listeners take less advantage of the relatively silent periods (“dips”) in a fluctuating background sound than do normal-hearing listeners (Festen & Plomp, 1990; George, Festen, & Houtgast, 2006; Gustafsson & Arlinger, 1994; Lorenzi, Husson, Ardoint, & Debruille, 2006; Peters, Moore, & Baer, 1998). The ability to “listen in the background dips” is believed to be related to the ability to take advantage of temporal fine structure cues of speech sounds. Hearing impairment either worsens the precision of phase locking or hinders the ability to extract information from phase locking (Moore, 2007). Consequently, noise has a more detrimental effect on listeners with hearing loss than listeners with normal hearing. Finally, hearing-impaired listeners obtain less binaural advantage than do normal-hearing listeners (e.g., Bronkhorst & Plomp, 1989; Darwin, 2006; Dirks & Wilson, 1969; MacKeith & Coles, 1971). One way to describe binaural advantage for speech perception is to compare the difference between speech perception performance measured in spatially-separated and co-located noise (i.e., speech and noise are coming from the same direction). It has been established that speech recognition performance in noise can be improved when speech and noise sources are separated in space (e.g., Dubno, Ahistrom, & Horwitz, 2002; Freyman, Helfer, McCall, & Clifton, 1999; Hawley, Litovsky, & Culling, 2004; Hirsh, 1950). The amount of speech recognition improvement due to this spatial separation depends on the type, number, and position of noise sources. Hearing-impaired listeners have been found to benefit less from spatial separation and have less binaural advantage than normal-hearing listeners (e.g., Arbogast, Mason, & Kidd, 2005; Gelfand, Ross, & Miller, 1988). Moreover, evidence supports the notion that hearing-impaired listeners have more difficulty directing attention to a desired sound source (selective attention) and filter out other acoustical interferences than do normal-hearing listeners in complex and dynamic auditory environments (e.g. "cocktail party" environments) (Gatehouse & Akeroyd, 2006; Shinn-Cunningham & Best, 2008). As a result, people with hearing loss typically encounter more challenges when listening in noise compared to people with normal hearing.

Speech Perception in Spatially-separated Noise

The "cocktail party effect" refers to how listeners manage to understand one person while attending less to others who may be speaking at the same time (Cherry, 1953). Spatial separation of speech and noise has been shown to help listeners hear better in a "cocktail party” environment, because these listeners can use binaural cues to determine speech and attenuate other unwanted sounds. For example, the level of unwanted noise may be reduced at certain frequencies as a result of the head shadow effect, especially for the higher frequencies with shorter wavelengths (Masterton, Heffner, & Ravizza, 1969). This head shadow effect, or better-ear effect, can improve SNRs at one ear or the other (Bronkhorst & Plomp, 1988; 1992). Another possible effect might be due to the binaural auditory system taking advantage of interaural differences of received sounds. Some researchers refer to this effect as “binaural interaction” or “binaural analysis”, which predominates in the low frequencies (e.g., Bronkhorst, 2000; Levitt & Rabiner, 1967; Zurek, 1993). In addition, Bronkhorst and Plomp (1989) proposed that hearing-impaired listeners benefit less from speech-noise separation than normal-hearing listeners because of the reduced audibility of high frequencies. That is, hearing-impaired listeners cannot make use of the head shadow effect, which is most significant at high frequencies. Others have suggested that a reduced ability to make effective use of interaural differences may contribute to smaller binaural benefit (e.g., Gelfand et al., 1988). Because integrating interaural differences is also necessary for sound localization, it is of interest to know if people need to be able to localize sound in order to benefit from spatial separation to improve speech perception.

Sound Localization

Sound localization requires the integration of multiple acoustic cues (for a review, see Blauert, 1997). The ability to localize on the horizontal plane arises primarily from the successful use of binaural cues: low-frequency interaural time differences (ITD) and high-frequency interaural level differences (ILD). Spectral cues that are due to sound reflection from the pinna, head, and torso are referred to as the head-related transfer function (HRTF). The comparison of HRTFs across ears for a given source position provides the interaural spectral cues necessary for localization. It has been suggested that ITDs and ILDs are mainly useful for judgments in the left/right dimension, while spectral cues are primarily responsible for the front/back dimension and vertical sound source localization (Wightman & Kistler, 1992).

People with hearing loss have been found to have difficulties using interaural time, level or spectral cues to locate sounds (e.g., Abel & Hay, 1996; Lorenzi, Gatehouse, & Lever, 1999a; Noble, Byrne, & Lepage, 1994; Noble, Ter-Horst, & Byrne, 1995). The presence of noise can further reduce localization accuracy (e.g., Good & Gilkey, 1996; Lorenzi et al., 1999a; Lorenzi, Gatehouse, & Lever, 1999b). It has been found that the localization accuracy is monotonically related to the SNR of the environment. That is, the better the SNR, the higher the localization accuracy. Good and Gilkey (1996) found a similar correlation between SNR and localization accuracy in broadband noise for people with normal hearing.

The impact of hearing loss on localization accuracy depends on the configuration, type of hearing loss, and plane of interest. For example, Noble et al. (1994) found that listeners with conductive/mixed hearing loss had worse localization performance at the horizontal plane than listeners with sensorineural hearing loss (SNHL) when the degree of hearing loss was accounted for. These same investigators found that listeners with higher frequency hearing loss had more difficulties locating sounds from front and rear sources than listeners with lower frequency hearing loss. Listeners with poorer low or mid-frequency hearing had reduced accuracy in frontal-horizontal plane localization tasks than listeners with better hearing.

It is interesting to note that the correlation between hearing loss and localization is not strong (Byrne & Dirks, 1996; Noble et al., 1994). That is, localization performance cannot be predicted from hearing thresholds (Durlach, Thompson, & Colburn, 1981); deficits other than loss of audibility may also deteriorate localization accuracy. For example, Lorenzi et al. (1999b) reported that localization accuracy for high-passed stimuli was not affected when normal-hearing listeners had simulated high-frequency hearing loss. Similar results were found for listeners with simulated low-frequency hearing loss using low-passed stimuli.

Lorenzi et al. (1999a) investigated the relationship between the simultaneous detection of a broadband click train and localization in noise. The results indicated that detection performance was similar among different conditions until the SNR of the click train and the noise reached 0 dB. However, the detection performance declined in adverse SNRs worse than 0 dB. It was found that the detection performance was similar when noise came from either the front or the side, which implied that the decreased localization accuracy with the noise presented from the side cannot be solely attributed to reduced audibility of the targets. Although Good et al. (1997) noted that signals that are easier to detect are usually localized more accurately, it appears the detection of a signal in a free field does not strongly rely on the ability to locate the signal. In other words, it is not necessary to be aware of the location of the signal before detecting it. It is possible that localization requires more central auditory binaural processing ability than mere detection of sounds, which happens peripherally in the auditory system.

The Relationship between Speech Perception and Localization in Noise

In the early 50's, exploration of the relation between localization and the ability to separate a desired signal from other sounds coming from different directions commenced. Hirsh (1950) first proposed a possible correlation between localization and speech perception in noise. That is, he postulated that localization of speech and noise might play a role in the speech perception task. It is likely that the binaural system utilizes the interaural differences across ears to locate sound sources and separate signals based on locations. Cherry (1953) proposed that the ability to "hear out" signals may be mediated by a localization-based signal segregation. More recently, growing evidence supports the idea that localization could be one of the attributes needed to separate multiple auditory sources and help segregate competing speech sounds (e.g., Bregman, 1990; Darwin & Hukin, 2000a, 2000b; Drennan, Gatehouse, & Lever, 2003). When a signal and one or more maskers are spatially separated in a free field, it is easier to locate and attend to the signal (Yost, 1997). One could speculate that poor localization skills might lead to poor speech perception. Several studies have reported that some hearing-impaired listeners with reduced localization abilities also do poorly on speech perception tasks in noise, while others with normal localization abilities exhibit good performances on speech perception tests (Kubo et al., 1998; Sebkova & Bamford, 1981); however, the inverse is not true. That is, some listeners who have good speech perception do poorly on localization tasks. Current data indicate that the correlation is relatively poor between localization error and speech perception performance, even when loss of audibility is accounted for (Noble, Byrne, & Ter-Horst, 1997). Another investigation found that although the absolute localization of target talkers was poor, listeners could still benefit from spatial separation and had improved speech perception performance (Drullman & Bronkhorst, 2000). On the other hand, it is possible that listeners can benefit from visual cues and increased SNRs to improve speech perception in spatially-separated noise if they can locate the sound source and turn to face it.

Localization ability and the mechanism underlying the benefit from spatial separation of speech and noise may both rely on the same cues (ILD and ITD cues); still, this does not infer a causal relationship between localization and speech perception in noise. It is possible that the inability to effectively use binaural cues (e.g., hearing loss) may have a negative impact on both localization ability and speech intelligibility in spatially-separated noise. Because amplification is the primary remediation for hearing loss, the following section will discuss the effect of bilateral amplification on localization and speech perception in noise.

The Effect of Bilateral Amplification

Bilateral amplification has been recommended as a way to improve localization, sound quality and speech perception in noise (e.g., Dillon, 2001; Holmes, 2003; Ricketts, 2000; Simon, 2005). Although bilateral amplification does not always result in binaural hearing, the advantages of bilateral amplification, if they occur, may be similar to known binaural benefits. One advantage of having two ears arises from the brain being able to take advantage of differences in signals arriving at the two ears (squelch effect) and/or similarities in signals arriving at both ears (binaural summation). Another advantage of binaural hearing is that listeners can often make use of the head shadow effect or better-ear effect (Brown & Balkany, 2007).

Several studies have shown that bilateral hearing aid fittings can improve speech perception over monaural fittings in spatially-separated noise or co-located noise (e.g., Laurence, Moore, & Glasberg, 1983; Moore, Johnson, Clark, & Pluvinage, 1992), and can provide better horizontal plane localization than unilateral hearing aid fittings for moderate to severe hearing loss (e.g., Noble & Byrne, 1991; Ricketts, 2000; Van den Bogaert, Klasen, Moonen, Van Deun, & Wouters, 2006). Yet, other researchers have questioned whether hearing aids could also interfere with sound localization performance and/or speech perception in spatially-separated noise (e.g., Hausler, Colburn, & Marr, 1983; Kalluri & Edwards, 2007; Van den Bogaert et al., 2006). For example, the microphone positions of different hearing aid styles could have an impact on localization (Hausler et al., 1983; Noble & Byrne, 1990). The microphone position on behind-the-ear (BTE) hearing aids, above the pinna, could disrupt the spectral cues, which may interfere with accurate front/back discrimination. Several studies have indicated that listeners wearing BTE hearing aids had poorer localization performance than listeners wearing in-the-ear (ITE) hearing aids (Hausler et al., 1983; Orton & Preves, 1979; Westermann & Topholm, 1985). Others have suggested that hearing aid configuration had little impact on localization in the horizontal plane when unaided performance was accounted for (e.g., Noble & Byrne, 1990). It should be noted that unaided localization performance tends to be better than aided performance (e.g., Byrne & Noble, 1998), even when audibility is accounted for.

A second source of potential disruption to localization ability lies in the processing time of independently-fit hearing aids. Besides the inherent processing delay[1] in the digital signal processing, amplitude compression time constants could also vary between the ears, having further detrimental effects on binaural cues (Dillon, 2001; Kalluri & Edwards, 2007; Keidser et al., 2006; Van den Bogaert et al., 2006). For example, two hearing aids with wide dynamic range compression that work independently may give less gain for the higher-level sounds arriving at the near ear compared to the lower-level sounds at the far ear, resulting in distorted ILD cues. Despite this, two studies to-date have reported that the impact of compression on localization is trivial (Keidser et al., 2006; Musa-Shufani, Walger, von Wedel, & Meister, 2006). Another study has suggested that compression may reduce binaural advantages due to the spatial separation of speech and noise for hearing-impaired listeners (Kalluri & Edwards, 2007). Chung (2004) also proposed that the processing delays due to digital signal processing may be mismatched between bilaterally fit hearing aids. In addition, the interaural phase difference could be distorted as well. Therefore, it is possible that independently fitted devices could deteriorate localization accuracy and speech perception in spatially-separated noise. It is noteworthy that there is evidence to support that human brain can adapt to the mismatched delays within hours or days (e.g., Javer & Schwarz, 1995; King et al., 2001). It infers that if the mismatched processing time does not change too often, people may still be able to localize sounds using distorted cues.

Finally, it remains unclear whether bilateral hearing aids should be "coordinated" in their processing. Many current manufacturers provide the option to link the gain/output

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[1] Processing delay refers to the time delay between the input and output of the same signal throughout a hearing aid.

and/or adaptive features so that both hearing aids change given parameters simultaneously. Several investigators have reported the speech perception performance in noise to be similar whether the hearing-impaired listeners are using bilateral directional hearing aids or an omnidirectional aid in one ear and a directional aid in the other (e.g., Bentler, Egge, Tubbs, Dittberner, & Flamme, 2004; Cord, Walden, Surr, & Dittberner, 2007). Yet, other investigators have found hearing-impaired listeners exhibit superior speech perception performance in noise when using bilateral directional hearing aids as compared to a mismatched fitting (directional aid in only one ear) (Hornsby & Ricketts, 2007; Mackenzie & Lutman, 2005). It is possible that even when hearing aids are meant to work simultaneously, the various features don't actually engage simultaneously. For example, Banerjee (2008) explored the percentage of time that three digital features in bilateral hearing aids were synchronized in daily life for 10 hearing aid users over four weeks. In her study, the term "synchronization" was defined as when both hearing aids perform in the same manner for a given amount of time; that is, if a change occurs in one hearing aid, the other one would make a similar change simultaneously. These data were obtained from a data-logging feature of the hearing aids that captured the input level and digital features' status from each side. The results indicated that 78-97% of the time bilateral hearing aids were synchronized. The investigator suggested that further synchronization of bilateral hearing aids may not be necessary, because bilateral hearing aids are "naturally synchronized most of the time."

In summary, while some studies have shown that bilateral amplification can improve performance when listening in noise, others have suggested that independently fit (mismatched) bilateral hearing aids may interfere with localization and speech perception in spatially-separated noise. As a result, it remains unclear whether bilateral hearing aids should be synchronized or not for purposes of localization and speech perception. In the current investigation, the effect of Synchrony on gain reduction patterns from digital noise reduction (DNR) schemes will be assessed in terms of speech perception and localization.

Digital Noise Reduction

Significant effort has gone into the development of hearing aid processing schemes and features that both can improve speech recognition and reduce listening effort in noise. DNR algorithms are one possible way of helping people hear better in noise. While there are varying algorithms across industry, the basic concept has been to reduce output in noisy environments. With the development of modern techniques, the concept of DNR has been broadened to include several other technologies besides classic DNR (Bentler, Ricketts, & Mueller, 2010). Classic DNR algorithms take advantage of the temporal differences between speech and noise, particularly in terms of modulation rate and depth. Since noise and speech often coexist in time, frequency, and space, the accuracy of the detection phase is often unreliable. For example, it may be particularly difficult to accurately detect and reduce noise when the noise is "speech-like," such as another talker or multi-talkers. Other DNR algorithms include fast filtering (e.g., Wiener filter), wind noise reduction and impulse noise reduction. The current study only focused on the classic DNR algorithms.

More complex classic DNR algorithms are now available that employ decision rules capable of defining what constitutes noise, how much gain reduction is appropriate, and in which frequency ranges the gain reduction should be accomplished (Bentler & Chiou, 2006). Mueller and Ricketts (2005) reported that there are 100 or more models of digital hearing aids in today's market employ DNR algorithms, each with its own rules of application. Most research studies to-date support the notion that DNR schemes can reduce listening effort and improve sound quality for hearing aid use, but few suggest any improvement in speech intelligibility in noise (for a review, see Bentler & Chiou,2006; Bentler, 2005).

Classic DNR algorithms used in hearing aids generally include two stages. The first stage detects and analyzes incoming signals in each frequency channel. The second stage adjusts gain in each frequency channel. Different hearing aid manufacturers implement different detection, analysis and gain reduction rules (for a detailed review, see Chung, 2004). In general, the rules depend on the estimated environmental SNRs in each frequency channel, frequency importance functions, the degree of noise reduction selected from the fitting software, and the input levels.

The following section will discuss the potential impact of bilaterally independent DNR schemes on localization and speech performance in spatially-separated noise and explores the importance of the proposed study.

Only one study has addressed the impact of bilateral DNR as implemented in a commercial hearing aid on localization (Keidser et al., 2006). These findings supported the notion that bilaterally independent (mismatched) DNR can have a negative impact on localization in the horizontal plane. The investigators demonstrated that the ILDs were increased when the noise was coming from one side, with the DNR being activated in both hearing aids. It was noted that localization performance in the frontal-horizontal plane was poorer when the DNR was activated than when it was deactivated. It should be noted that this study only included one DNR algorithm. Our pilot data indicated that DNR algorithms differ across manufacturers, resulting in different gain reduction patterns (see Chapter 3 for details). It is possible that other gain reduction patterns may result in more or less detrimental effects on localization ability. Furthermore, it is unknown whether this deteriorated performance could be restored by coordinating, or synchronizing, the bilateral DNR.

Finally, although no published studies to-date have explored the impact of bilaterally independent (mismatched) DNR on speech perception in spatially separated noise, some early studies explored the possibility of allowing hearing aid users to take advantage of binaural cues when using DNR schemes. For example, one earlier study advocated a bilateral noise reduction scheme to help hearing-impaired listeners in noisy environments (Kollmeier, Peissig, & Hohmann, 1993). The basic idea was to maintain the binaural cues for listeners. This noise reduction scheme implemented algorithms in both ears to calculate and compare the ITDs and ILDs for each frequency component of processed stimuli to "reference" unprocessed stimuli. Any deviations of ITDs and ILDs of processed from the unprocessed stimuli were restored. The results indicated that in spatially-separated noise, speech perception performance with the processed stimuli was better than that with the unprocessed stimuli for both normal-hearing and hearing-impaired listeners. To date, this kind of DNR scheme has not been implemented in any wearable hearing device.

In short, little is known about the impact of using different gain reduction patterns from bilaterally independent (mismatched) DNR compared to bilaterally synchronized (matched) DNR on localization in noise. Even less is known about the impact of bilaterally independent DNR on speech perception performance in spatially-separated noise, and whether bilaterally synchronized (matched) DNR can restore the detrimental impact caused by bilaterally independent (mismatched) DNR. As a result, it is important to investigate the impact of bilateral DNR on localization and speech perception in spatially-separated noise. In the proposed study, the first stage of DNR (i.e. detecting and analyzing incoming signals) was not considered to be a variable. Instead, this study only focused on the second stage (i.e., reducing gain in different frequency channels).

The purpose of the study was to examine the impact of bilateral gain reduction on speech perception and localization in spatially-separated noise. Since previous studies have suggested that noise reduction can also improve sound quality and reduce listening effort, those outcomes were also assessed. Finally, the relationship between measured localization and self-reported localization performance in daily life was assessed as well.

Hypotheses and Research Questions

Hypotheses

Hypothesis 1: Hearing aid users will have inferior performance in both localization and speech perception in spatially-separated noise when given a bilaterally mismatched gain reduction scheme (gain reduction in one ear only) as compared to an unaltered bilaterally linear time-invariant amplification scheme wherein audibility is optimized (no gain reduction in either ear).

Hypothesis 2: Hearing aid users will have superior performance in both localization and speech perception in spatially-separated noise when given a bilaterally matched gain reduction scheme (same gain reduction between ears) as compared to a bilaterally mismatched gain reduction scheme (gain reduction in one ear only).

Research Questions

Localization research questions

1. Will localization in noise in the frontal-horizontal plane be negatively affected when hearing aid users are given a bilaterally mismatched gain reduction scheme (gain reduction in one ear only) as compared to an unaltered bilaterally linear time-invariant amplification scheme (no gain reduction in either ear)?
2. Will deteriorated localization performance in noise due to the bilaterally mismatched gain reduction scheme (gain reduction in one ear only) be restored by a bilaterally matched gain reduction scheme (same gain reduction between ears)?
3. Using a self-report inventory, is there any relationship between self-reported localization performance and measured localization performance?

Speech perception research questions

1. Will speech perception performance in spatially-separated noise in the frontal-horizontal plane be negatively affected when hearing aid users are given a bilaterally mismatched gain reduction scheme (gain reduction in one ear only) as compared to an unaltered bilaterally linear time-invariant amplification scheme (no gain reduction in either ear)?

2. Will deteriorated speech perception performance in noise due to the bilaterally mismatched gain reduction scheme (gain reduction in one ear only) be restored by a bilaterally matched gain reduction scheme (same gain reduction between ears)?

<u>Sound quality and listening effort research questions</u>

1. Do different gain reduction patterns result in different sound quality ratings for hearing aid users?

2. Do different gain reduction patterns result in different listening effort ratings for hearing aid users?

CHAPTER 2

LITERATURE REVIEW

Chapter 1 introduced the importance of the current study's research questions and summarized its purposes and hypotheses. This chapter provides a more thorough description of background information essential for a deeper understanding of the study. Topics include an overview of issues regarding noise reduction, sound localization, and speech perception in spatially-separated noise.

Noise Reduction

Noise is considered to be any unwanted signal received by a hearing device. This noise can be music, speech from other talkers, and/or environmental signals (e.g., automobile noise). One of the most common problems encountered by hearing aid users is the interference of background noise. Generally speaking, three types of noise are particularly damaging to speech intelligibility (Levitt, 2001). The first is noise having a spectrum similar to that of speech. The second is speech interference; for example, single-talker or multi-talker backgrounds (babble noise or cocktail-party noise). The third is room reverberation. Reverberation is the persistence of sound in a room resulting from repeated reflections from the room's boundaries. Though some reverberation can be helpful to reinforce the speech signal, too much reverberation can reduce speech intelligibility and overall sound quality in a hearing aid, particularly in the presence of other types of noise (Boothroyd, 2006).

As mentioned in Chapter 1, enormous effort has been focused on the development of a hearing aid that can improve speech recognition and lighten the need for listening effort in noise. For example, the use of directional microphones is one possible way of improving speech perception in noise (e.g., Bentler, Palmer, & Mueller, 2006; Boymans & Dreschler, 2000). Simply put, directional microphones can pick up a target signal from in front of a listener and diminish the unwanted signals from the rear or side. This technology takes advantage of the spatial separation between the signal and the noise by reducing microphone sensitivity for inputs from the side or rear azimuths. By distorting the binaural and spectral cues, a directional microphone scheme may also affect the listener's ability to localize sound (e.g., Chung, Neuman, & Higgins, 2008; Keidser et al., 2006; Van den Bogaert et al., 2006).

The employment of noise reduction algorithms is another possible way to help people hear better in noise. The main objectives of noise reduction as applied to speech communication are to improve the overall sound quality and reduce the listening effort of the hearing aid user. A noise reduction method must attenuate noise components that mask the information-bearing components of speech and, if possible, should do so without suppressing or distorting important speech components (Weiss & Newman, 1993). Otherwise, the removal of noise may also reduce speech intelligibility because important speech cues buried in noise are removed as well.

Two general principals can be applied to the problem of the effect of noise on speech intelligibility (Levitt, 2001). First, the more we know about the interaction of speech and noise, the more we can do to reduce the effects of the latter; second, the larger the differences between the two , the more we can do to reduce noise's effects on speech. However, since noise and speech often coexist at the same point in time and share the same frequency and space domains, the aim of reducing noise's negative effects on speech intelligibility is a difficult one to achieve. The most realistic goal in solving this problem is to suppress noise components that coexist with speech in regions where the speech components do not convey significant perceptual cues.

Noise reduction has been an available feature in hearing aids since the 1970s. The most commonly employed noise reduction methods include the use of filtering, amplitude compression, and gain reduction based on modulation detection. The use of directional microphone and multi-microphone methods (e.g., adaptive noise cancellation and adaptive beam forming) are not discussed here because they are related to the spatial properties of speech and noise, which are involved in other possible methods to improve speech intelligibility. In the current review, noise reduction will be discussed in regard to the time and frequency domains. (For detailed reviews, see Bentler and Chiou (2006), and Chung (2004).)

Basic Approaches

The basic approaches discussed here are based on spectral differences between speech and noise.

Fixed high-pass filters

Many types of environmental noise, which are typically steady-state or time-invariant, contain large amounts of low frequency energy. According to the Speech Intelligibility Index (SII) (ANSI S3.5- 1997) which has been used to predict speech intelligibility in noise, all frequency bands below 0.4 KHz have a negative signal to noise ratio (SNR) and make no contribution to speech intelligibility because the speech is already masked by the noise. In addition, the possible excessive upward-spread-of-masking could further reduce speech intelligibility for people with hearing loss (Gagne, 1988; Trees & Turner, 1986). It is possible to eliminate this part of noise without sacrificing speech intelligibility by employing a fixed high-pass filter. In addition, with the removal of low frequency intensive energy, overall sound quality can be improved as well. This constitutes the simplest method of reducing noise, and is a feature that can be manually switched on or off as required. Therefore, this approach was usually implemented in analog noise reduction algorithms.

Unfortunately, it is not easy to adjust a filter's frequency response to match the noise's spectral characteristics. If the cutoff frequency happens to be set to include the positive SNR part, some loss of intelligibility could result (Levitt, 2001; Weiss & Newman, 1993). Another problem is that the frequency spectra of everyday noise are seldom significantly different from that of speech and are time invariant. Thus, a fixed high-pass filter cannot effectively eliminate most of the noise without reducing speech intelligibility at the same time (Neuman & Schwander, 1987).

Adaptive filters

If a reasonably accurate estimate of the noise spectrum as it varies over time can be obtained, it is possible to use adaptive filtering to reduce noise levels without negatively influencing speech intelligibility. One example that was used in a commercial hearing aid is the Zeta Noise Blocker (Intellitech Inc Corp). It employed overlapping bandpass analog filtering whose cutoff frequencies and gains could be adjusted to suppress noise across the hearing aid's full bandwidth when the noise level exceeds that of speech (Graupe, Grosspietsch, & Taylor, 1986).

Another example of the adaptive filter is the Wiener filter (Wiener, 1949). The transfer function of this filter is defined by the power spectrums of the clean signal, Ws, and of the noise, Wn, and is given as Ws/(Ws+Wn). The aim of Wiener filtering is to enhance the SNR when the characteristics of signal and noise are known and are stationary in a statistical sense. This requirement is rarely met in real-world situations. However, because the speech spectrum varies relatively slowly in time for many speech sounds as does the short-term noise spectrum, it is possible to use the Wiener filter for short time periods. This approach is referred to as an adaptive Wiener filter (Bentler & Chiou, 2006; Weiss & Newman, 1993). Levitt et al. reported that some hearing-impaired subjects experienced benefits when using short-term Wiener filtering.

Spectral subtraction

Spectral subtraction, a digital scheme, is based on an estimate of the noise spectrum of noisy speech (Loizou, 2007). The noise spectrum can be obtained when speech is absent and thus can be subtracted from the speech-plus-noise spectrum when the speech is again present. Therefore, spectral subtraction can be used to generate an enhanced speech signal. Although this method can improve the SNR for many commonly encountered ambient noises, speech intelligibility is not improved in all cases because the processed signal can reveals audible distortions (processing noise) and speech information is occasionally eliminated. On the other hand, most studies found that the sound quality was improved using this approach (e.g., Boll, 1979; Hu & Loizou, 2006).

Sinewave modeling

In this approach, the major peaks in the speech-plus-noise spectrum are obtained. These peaks are frequently located at the harmonics of voiced speech sounds and consist mostly of speech with a small amount of noise. Conversely, the spectral components between these peaks, which are mostly of noise, are discarded. The spectral peaks are then converted back to a time waveform, resulting in an improved SNR with some audible distortion (Kates, 1994).

In summary, the approaches described above essentially yield the same results in terms of improved sound quality ratings but none have been shown to provide a significant change in speech intelligibility (e.g., Dillon & Lovegrove, 1993; Weiss & Newman, 1993). Because of the time-variant nature of the world in which we live and because speech has known temporal patterns, called modulations, noise reduction algorithms have changed from analog filtering of targeted frequency regions to digital filtering based on the temporal characteristics of the environmental signals (Bentler & Chiou, 2006). It should be noted that some of these basic approaches currently implements in digital noise reduction algorithms, such as Wiener filtering.

Digital Noise Reduction (DNR)

The so-called classic DNR algorithms (Bentler, Ricketts, & Mueller, 2010) are based on modulation detection, which differentiates speech from noise based mainly on amplitude changes over a short time period (modulations). The temporal modulations relevant to speech occur usually from 0.1 to 40Hz. Drullman, Festen and Plomp (1994) found that temporal modulations below 4Hz do not impact speech intelligibility for normal-hearing listeners. The most important temporal modulation frequencies for speech are located from 4 to 16Hz whereas most noise has a higher modulation rate than speech. Another of modulation's parameters is called modulation depth (in dB), which is a measurement of the amplitude from peak to valley. For clean speech modulation, the modulation depth is about 30-50dB. Table 1 displays a comparison of modulation characteristics among clean speech, speech babble, and jet noise (Bentler & Chiou, 2006). The jet noise exhibits the highest modulation frequency and least modulation depth among these three sounds. Determination of the amount of gain reduction for classic DNR schemes is typically based on these two parameters. That is, gain reduction may be implemented when the modulation frequency is above 10Hz and/or the modulation depth is relatively low.

Another kind of DNR algorithm is based on comodulation detection, which is also referred to as synchronous morphology. Speech contains energy in different frequency regions that is synchronized in periodic bursts by the opening and closing of the vocal folds, which is called comodulation. Noise or non-speech sounds rarely have comodulation. Gain reduction happens when there is an absence of harmonic structure (Bentler & Chiou, 2006; Mueller & Ricketts, 2005). The criteria for determining gain reduction (either modulation or comodulation detection) can be applied to any or all channels of the digital hearing aid. Table 2 presents the DNR schemes across seven different manufacturers. Six of them use the classic DNR based primarily on modulation detection. Only one manufacturer among six implements Wiener filtering plus modulation detection. One of seven uses comodulation detection (synchronous morphology). Other DNR algorithms used in commercial hearing aids deal with some specific type of noise, such as wind noise reduction or impulse noise reduction. The current study focuses only on the classic DNR algorithms.

<u>The effectiveness of DNR</u>

The term “effectiveness” relates to how well hearing aids could work for hearing-impaired listeners in practice. It is usually evaluated based on clinical trials followed self-reported outcome measurements obtained in the real world. Cox (2003) pointed out that many aspects of real-life outcomes cannot be assessed in the laboratory because traditional laboratory measurements (e.g., speech recognition in noise) cannot reflect the individualized hearing difficulties that hearing impaired listeners might have. The best way to quantify problems and outcomes is to use self-reported data. Another reason to consider self-reported outcome measurements is that laboratory outcome measurements do not resemble hearing-impaired listeners’ impressions of real-life outcomes despite attempts to make laboratory simulations as accurate to real-world conditions as possible.

In general, the results from studies of DNR suggest no speech intelligibility improvement using DNR in laboratory settings (Alcantara, Moore, Kuhnel, & Launer, 2003; Bentler, Wu, Kettel, & Hurtig, 2008; e.g., Boymans & Dreschler, 2000; Walden, Surr, Cord, Edwards, & Olson, 2000). However, most studies have reported that the use of DNR can reduce listening effort or improve ease of listening. For example, Bentler et al. (2008) found that the ratings for ease of listening and listening comfort were significantly higher when DNR was switched on than when it was off, regardless of the time constants of the algorithm. In contrast, the results of sound quality rating comparisons between with the activation of DNR are varied. Some researchers have reported a significant and strong preference for DNR in noisy contexts (e.g., Chung, Tufts, & Nelson, 2009; Ricketts & Hornsby, 2005). Others did not find any significant differences between the DNR on and off in terms of sound quality ratings (e.g., Alcantara et al., 2003; Bentler et al., 2008).

One of the most popular self-reported outcome metrics for hearing aid users, the Abbreviated Profile of Hearing Aid Benefit (APHAB), has been used in numerous studies related to DNR. The APHAB, developed by Cox and Alexander (1995), has four subscales:

Ease of Communication (EC), Reverberation (RV), Background Noise (BN), and Aversiveness (AV). Using this tool, evidence supports the notion that the use of DNR is beneficial to hearing aids users regarding perceived aversiveness. For example, Boymans and Dreschler (2000) found three items on the aversiveness subscale supported the usefulness of DNR: speech recognition in car noise, sudden loud sounds, and traffic noises. Bentler et al. (2008) reported that the perceived aversivness in the field trial was not different between the DNR-on and pre-fitting conditions. However, perceived aversivness was significantly greater with DNR-off than during the pre-fitting. This suggests that hearing aids users might experience less aversiveness when DNR is activated. In another study, however, there was no change in perceived aversiveness towards noise compared to normal-hearing listeners (Palmer, Bentler, & Mueller, 2006).

In summary, DNR in hearing aids has not been shown to result in improved speech understanding in noise. Yet, most studies support the use of DNR to improve sound quality and enhance ease of listening and listening comfort.

The following sections briefly review key characteristics of spatial hearing and speech hearing in spatially-separated noise. A brief analysis of the relationship between these two aspects is also provided.

Spatial Hearing - Sound Localization

The phenomenon of sound localization has been extensively studied for many years (see reviews by Blauert, 1997; Gilkey & Anderson, 1997). Sound localization is important in real life because of the role it plays in helping people avoid danger. For example, the safety of the workers in a high noise environment is dependent upon good localization abilities (Morata et al., 2001; Morata et al., 2005). Another important practical function of the ability to localize sounds is that it allows listeners to constantly adjust to sound targets as conversations switch between speakers. As a result, listeners can take advantage of both auditory and visual cues to improve communication in difficult listening environments.

As sound travels from its source to a listener's ears, its characteristics are changed by the interactions between the sound and the listener's pinnae, head, and body (Brungart & Rabinowitz, 1999). Spatial hearing refers to the ability to use these changes as acoustic cues to determine the position of a sound source in space. The two components of spatial hearing are the perceived direction of a sound source and the perceived distance between the listener and the sound source. Kopco (2003, p.5) indicated that "The basis of spatial hearing is that the listener's auditory system extracts cues about the localization of the sound source from the sounds received at the ears and the listeners use the cues to perform localization and detection of sounds." In short, spatial hearing can help listeners locate sound sources and improve their perception of signals masked by other spatially separated sounds.

Localization Cues

The pinnae, head, and torso reflect sounds and produce a direction-dependent filter for sounds from all directions (e.g., Moore, 2007). The listener's head filters the sound, so that the spectrum at the location of the tympanic membrane would be different across the frequencies if the head were not present. The ratio or dB difference between these two conditions (i.e. presence or absence of the head) is referred to as the head-related transfer function (HRTF) or the free-field transfer function.

The HRTF forms three acoustic cues which contribute to the ability to localize a sound: interaural time differences (ITD), interaural level differences (ILD), and spectral cues, respectively (Middlebrooks & Green, 1991; Wightman & Kistler, 1997). The phase of the HRTF ratio is related to the ITDs and the amplitude of the HRTF ratio is related to the ILDs (Wightman & Kistler, 1997). The filtering action of the pinna generates the monaural spectral cues. Simply put, if a sound is not located directly in front of or behind the listener's head, it will travel at different distances and be diffracted by the head and pinna. The ability to localize sounds in the horizontal plane comes mainly from the ability to compare the sounds that arrive at two ears at different time intervals, intensity levels, and spectral shapes. It has been widely accepted that ITD cues are dominant at low frequencies, while ILD and spectral cues are dominant at high frequencies. This concept is often referred to as the duplex theory (Rayleigh, 1907).

The ability to detect small changes in interaural differences is related to a listener's resolving ability in the process of localizing sounds. Humans can distinguish auditory angles to as small as 1-2 degrees (Mills, 1958, 1972). This is often referred to as minimal audible angle, the smallest detectable change in angular position when stimuli are presented through loudspeakers in a free-field.

ITD cues

The arrival time differs across ears, resulting in ITDs. Different sound source angles create different ITDs. For example, the ITD for a sound straight ahead is zero μs while it could be as high as 690μs for a sound at a 90° azimuth (Moore, 2003). The amount of ITD varies as a function of frequency if the sound source position is fixed (Kuhn, 1977). Listeners can detect 10-15 μs ITDs from the median plane. These 10-15 μs ITDs can result in a difference in azimuth of 1-5 degrees (Blauert, 1997).

The ITD is the primary cue for localizing low-frequency sounds (< ~1500Hz) (Wightman & Kistler, 1992). ITDs can also be coded for the low-frequency envelope of high frequency stimuli and be used to localize a sound (e.g., Lorenzi et al., 1999b; Yost, Wightman, & Green, 1971). However, ITDs in a particular frequency region in reverberant environments may not be reliable (Shinn-Cunningham, Kopco, & Martin, 2005). In general, ITDs have been reported to be the dominant cues in sound localization in the horizontal plane (Wightman & Kistler, 1992).

ILD cues

The difference in sound pressure level between two ears produces ILDs. The head produces a barrier to the transmission of high frequency sounds. Therefore, acoustical energy builds up at one side of the head facing the sound source whereas it is blocked from reaching the other side of the head. This baffle effect lessens the sound pressure level at the shadowed ear, resulting in the ILDs. The baffle effect is passive and acts simultaneously on all signals in the environment. The ear that receives the more intense signals delivers the information to the brain to help localize the sound as coming from the left or right side.

The ILD is commonly considered useful for sound localization in high frequency areas (> ~ 1500Hz). The amount of ILD could reach 20 dB at high frequencies (Moore, 2003) or even reach 35dB at 10 kHz (Middlebrooks, Makous, & Green, 1989). The low-frequency ILD is usually very small and may be used for locating nearby sources (Brungart, 1999; Brungart, Durlach, & Rabinowitz, 1999). Frequency-dependent ILDs are considered more useful than a single overall ILD.

Spectral cues

The filtering function of each ear's pinna provides direction-dependent spectral notches and peaks, which form the spectral cues (e.g., Butler, Humanski, & Musicant, 1990). In effect, spectral cues are the product of the HRTF. Since spectral cues do not require the interaural comparison, they are often referred to as "monaural" cues. Wightman and Kistler (1997) pointed out that spectral cues were "highly idiosyncratic." In other words, each person has his or her unique HRFTs. In addition, the spectral cues only occur in the high frequencies (>~ 5000Hz) due to the small dimensions of the pinna. Spectral cues are considered to be most useful for judging sound source elevation and to help resolve front-back confusions (Musicant & Butler, 1984; Roffler & Butler, 1968). Wightman and Kistler (1997) also noted that listeners should be able to take advantage of monaural spectral cues as long as the listener has adequate high frequency hearing and the sound source has sufficient high-frequency content. Some studies have indicated that the spectral information between 2000- 5000 Hz is important for front-back discrimination (e.g., Butler, 1986).

Because each ear has its own HRTF, if a listener compares the HRTF across ears for a given source position which is not in the median plane, it can form interaural spectral cues necessary for localization (Blauert, 1997; Butler, 1987). These interaural spectral cues are eventually identical to frequency-dependent ILDs.

Generally speaking, ITD and ILD cues are usually considered the most useful for sound localization in the horizontal plane whereas the spectral cues are the primary cues in the vertical plane (Wightman et al., 1989; Wightman & Kistler, 1992).

Wightman and Kistler (1992) evaluated the relative importance of ITDs, ILDs, and spectral cues in an experiment employing headphones. They measured nine subjects' HRTF and programmed these individualized HRTFs to the stimuli to simulate the natural acoustic cues in a free field. Then they manipulated the ITDs in the stimuli by altering the phase components of the filters without affecting the ILDs and spectral cues. Thus, the ILDs and spectral cues were maintained in the same manner as naturally occurring cues. Three fixed ITDs were used. The results indicated that the fixed ITDs overrode the influence of ILDs and spectral cues on the horizontal plane. The perceived azimuth was determined by the fixed ITDs. The interesting finding was that the perceived elevation was not affected by the manipulation of the ITDs. Wightman and Kistler further found that the ILDs and spectral cues were dominant for the high-passed stimuli important for localization when the low-frequency ITDs were removed from the stimuli. When the high-pass cut-off frequency was lowered to the low end of frequency regions, the fixed ITDs again overrode the ILDs and spectral cues. Wightman and Kistler concluded that low-frequency ITDs are the dominant cues in establishing possible location, whereas the ILDs and spectral cues are the secondary cues to determine source direction, and most likely these two cues can help resolve confusion. In addition, Blauert (1997) discussed the equivalence of time difference to a particular level difference, which is referred as “trading ratio.” In the stimuli area between 1500Hz and 3000Hz, listeners cannot use the ITD or ILD cues as efficiently as they can in other frequency areas. However, it should be noted that another study conducted by Carlile, Delaney, and Corderoy (1999) found that high-frequency ILDs might be the dominant cues in the left-right localization.

In summary, the ability to localize sounds in the frontal-horizontal plane is based mainly on the analysis of ITDs and ILDs. Low-frequency ITDs are helpful in localizing broadband sounds, whereas high-frequency ILDs and spectral cues are dominant cues in localizing high-frequency sounds. The ability to localize sounds in the elevated plane is based primarily on spectral cues. This suggests that ITDs and ILDs are useful mainly for judgments in the left/right dimension while spectral cues are responsible primarily for the front/back dimension (Wightman & Kistler, 1992).

The Impact of Noise on Localization

The presence of noise can deteriorate the accuracy of localization. The impact of noise on the use of ITDs, ILDs, and spectral cues is varied. Most early free-field studies of spatial hearing did not directly evaluate the impact of noise on localization; rather, they examined the detection or the recognition of signals in noise (e.g., Bronkhorst & Plomp, 1988; Saberi, Dostal, Sadralodabai, Bull, & Perrott, 1991). Other studies measured the just noticeable difference (jnd’s) of ITDs and ILDs in the presence of noise using headphones. Those results indicated that when noise is presented diotically, the worse the signal/noise ratios (SNRs), the bigger the ITD and ILD jnd’s (e.g., Stern, Slocum, & Phillips, 1983).

In an experiment directly evaluating the impact of noise and different SNRs on sound localization, Good and Gilkey (1996) instructed three normal hearing subjects to make absolute judgments of sound locations in a free-field. The SNRs were relative to each individual's detection threshold when the signal and masker were both presented from the same position in front of the subject. The results indicated that localization accuracy in noise decreased monotonically in both the horizontal and vertical planes as the SNRs decreased. It is interesting to note that accuracy was not severely impacted by the noise as long as the SNRs remained sufficiently far from the detection threshold. The performance in the left/right and up/down dimensions was less influenced by the reduction in SNRs than that in the front/back dimensions. Good and Gilkey further calculated the systematic bias due to the masker. They found that the worse the SNR, the more likely the systematic bias were shifted toward the masker. Butler and Naunton (1964) reported this phenomenon as a "pulling effect," while Blauert (1997) named it "the phantom image."

Lorenzi et al. (1999b) later adopted the methodology from Good and Gilkey (1996) to investigate the relative influence of noise on ITD, ILD, and spectral cues in four normal-hearing listeners. The results indicated that the localization accuracy was not severely impacted when the SNR was at 0 dB or better regardless of where the masker was. When the SNR dropped below 0 dB, the localization accuracy started to decrease monotonically for all low-pass, high-pass, and broadband conditions. The masker at either side had a more negative influence than when located straight ahead. In fact, the pulling or pushing effect due to the masker was more distinct when the masker was at either side.

Simply put, the presence of noise has a minor impact on localization accuracy unless the SNRs are relatively poor. In those conditions, the localization performance would be reduced monotonically with the SNRs. Noise location has some differential influence on localization accuracy. Localization performance decreases more rapidly when the noise comes from the listener's sides than that when it is from in front. In addition, the response can be drawn towards or against the location of the noise masker, which is called the "pulling effect" or "pushing effect."

The Impact of Hearing Impairment on Localization

Hearing-impaired listeners have been found to have difficulties using interaural time, level, or spectral cues to locate sounds (e.g., Abel & Hay, 1994; Noble et al., 1994; 1995; Lorenzi et al., 1999a). For example, high-frequency hearing loss may prevent hearing-impaired listeners from using high-frequency dominated ILDs and spectral cues.

As mentioned in Chapter 1, the impact of hearing loss on localization accuracy depends on the configuration, type of hearing loss, and plane of interest. In general, listeners with conductive/mixed hearing loss showed worse localization performance on the horizontal plane than did listeners with sensorineural hearing loss (SNHL) when the degree of hearing loss was matched. Listeners with higher frequency hearing loss had more difficulty locating sounds from front and rear sources. Listeners with worse low or mid-frequency hearing showed less accuracy in frontal - horizontal plane localization (Noble et al., 1994).

It should be noted that the correlation between hearing thresholds and localization performance is relatively weak (e.g., Noble et al., 1994; Byrne & Dirks, 1996). That is, localization performance cannot be predicted solely based on hearing thresholds (Durlach et al., 1981). Lorenzi et al. (1999b) reported that localization accuracy for high-passed stimuli was not affected when normal hearing listeners had simulated high-frequency hearing loss. Similar results were found using low-passed stimuli in listeners with simulated low-frequency hearing loss.

It is interesting that no agreement exists concerning the ability of listeners with SNHL to use localization cues (Durlach et al., 1981) because large intersubject variability in binaural performance was found for those who had similar configuration, type, and degree of hearing loss (e.g., Gabriel, Koehnke, & Colburn, 1992; Simon & Aleksandrovsky, 1997). As well, these studies used a variety of methodologies, which might explain the differences of the results.

The presence of noise could further reduce the localization accuracy of hearing-impaired listeners (e.g., Good & Gilkey, 1996; Lorenzi et al., 1999a). Lorenzi et al. (1999a) explored localization accuracy on a group of high-frequency hearing-impaired listeners. The results indicated that the localization accuracy for broadband pulse trains did not deteriorate with the presence of noise until the SNR was below 0 dB. The impact of noise at +/- 90° on localization abilities was greater than that at 0°. In effect, localization accuracy exhibited a monotonical relationship with the SNRs of the test environment. That is, the poorer the SNRs, the poorer the localization accuracy.

In the same study of Lorenzi et al. (1999a), they also investigated the relationship between the simultaneous detection of a broadband click train and localization in noise. The results indicated that the detection performance was similar among different conditions until the SNR reached 0 dB. The performance was more reduced for adverse SNRs when the target and noise were co-located (i.e., from the same position) compared to when the target and noise were spatially separated. Because detection performances were similar when noise came from either the front or the side, the decreased localization accuracies regarding side-presented noise cannot be explained solely by the reduced audibility of the targets. The detection of a signal in the free field does not strongly rely on the ability to locate the signal, although other researchers found that signals that are more easily detected are usually localized more accurately (Good & Gilkey, 1996). It is possible that localization requires a more central auditory binaural processing ability than does mere sound detection, which happens peripherally in the auditory system. It has been found that localization accuracy is related to other factors, such as head movement cues, visual cues, or cognitive cues (Sayers & Cherry, 1957; Shelton & Searle, 1980).

In summary, localization performance is deteriorated in hearing-impaired listeners mainly because of the loss of audibility. The presence of noise may further decrease localization accuracy in hearing-impaired listeners.

The Impact of Hearing Aids on Localization

Because hearing impairment can negatively affect localization performance, it follows that the use of hearing aids can reduce the impact of audibility deficits. Over the past 50 years a number of investigators have studied this question (e.g., Byrne, Noble, & LePage, 1992; Dermody & Byrne, 1975; DiCarlo & Brown, 1960; Noble & Byrne, 1990, 1991).

One concern relevant to hearing aids' role in improving localization accuracy is related to the fact that the microphone position is often at the top of the pinna (e.g., behind-the-ear style) and might distort spectral cues. Another factor is that hearing aids do not typically amplify high frequency sounds above 4-6 KHz. The inability to amplify high-frequency sounds might negatively impact the use of spectral cues. More recently, the use of independent hearing aids (i.e., each hearing aid responds independently to its environment) has raised the concern that the interaural-difference cues could be disrupted. For example, bilateral hearing aids might delay the transmission of the sound to the tympanic membrane differently, such as different signal processing delays between ears, which may result in distorted timing cues. As a result, it is inconclusive whether two hearing aids can result in better localization performance than one. A number of studies have found that a bilateral fitting can result in better localization than unilateral fitting (e.g., DiCarlo & Brown, 1960; Byrne et al., 1992). However, it should be noted that bilateral fitting of hearing aids does not always restore natural binaural cues. In fact, some hearing aid users employing unilateral fittings can achieve localization accuracy similar that obtained with a bilateral fitting if the stimulus level is high enough (For a detailed review on the impact of bilateral amplification on localization, see Simon (2005).)

Other factors play a role in a hearing aid user's ability to localize sound. The following sections will discuss the effects of hearing aid styles, earmold types, bilateral fitting, and bilaterally independent signal processing on sound localization.

The impact of hearing aid styles

In a series of two classic studies undertaken by Noble and Byrne (1990, 1991), the impact of different hearing aid styles on localization performance was investigated. These investigators studied localization performance in three groups of hearing aid users using bilateral behind-the-ear (BTE), in-the-ear (ITE), or in-the-canal (ITC) devices in both the horizontal and vertical planes. Each group was tested wearing personal hearing aids as well as two other styles. In addition, a group of normal-hearing listeners participated in the study as a control group. The normal-hearing group showed good localization accuracy in both the horizontal and vertical planes without wearing hearing aids, that localization accuracy was severely decreased in the vertical plane when wearing hearing aids, and that horizontal localization performance was mildly deteriorated when using hearing aids. The three groups of hearing aid users performed similarly in the unaided condition. The ITC group performed poorer than the ITE and BTE groups when tested bilaterally aided using personal aids. Although the ITE group showed better localization accuracy than the BTE group, there was no significant difference between them. Other studies found that ITE users had better localization performance than BTE users (Orton & Preves, 1979; Westermann & Topholm, 1985). In addition, because hearing aids could interrupt the spectral cues from the pinna transformation and vertical localization relies heavily on the spectral cues, none of the hearing-impaired listeners in the Noble and Byrne study performed well when attempting to locate a sound in the vertical plane.

In summary, hearing aids have been shown to negatively influence localization performance in both the horizontal and vertical planes, regardless of hearing aid styles. The negative impact is most prominent in the vertical plane. Generally speaking, the localization performance with the ITE style is better than that of BTE or ITC.

<u>The impact of earmold types</u>

It is possible that earmolds can reduce interaural differences, which further deteriorates localization performance. Byrne et al. (1995) compared the localization performance between open and closed earmolds in two groups of bilateral BTE users with SNHL. One group had near-normal low frequency hearing, and the other group had better hearing in high frequency range (6000-8000Hz). The results did not support the notion that the open earmold can result in better localization performance than the closed one. Only two subjects with better high frequency hearing (<30 dBHL) showed improved vertical localization performance when using open earmolds. In addition, Byrne and colleagues investigated the impact of three types of earmolds on localization performance in a group of conductive/mixed hearing aid users. No significant differences were found across those three types of earmolds. Noble, Sinclair, and Byrne (1998) further specifically evaluated the impact of open earmolds on the localization ability of listeners with bilateral high-frequency hearing loss. Three kinds of experimental earmolds (closed, open, and sleeve) were coupled to the subjects' own hearing aids. Although the results did not indicate any significant differences in localization accuracy among those three earmolds, the closed earmold condition resulted in noticeably more errors than the other two open earmold conditions. The localization performance under the closed earmold condition was worse than the unaided condition.

In brief, the evidence from these two studies supports the notion that the impact of different earmold types on localization performance in hearing aid users is not significant, although the closed earmold tends to result in more localization errors compared to other kind of earmolds.

The impact of bilateral fitting

Noble and Byrne (1991) compared localization performance between unilateral and bilateral hearing aid fittings for the three groups previously studied by Noble and Byrne (1990). It is interesting that the localization performance for the BTE group was similar between unilateral and bilateral fitting conditions. In contrast, the ITE group showed significantly worse performance when fit unilaterally than bilaterally. The ITC group's results were somewhat inconsistent. The normal-hearing group, which served as a control, showed significantly worse localization performance under the unilateral fitting condition than under the bilateral fitting condition. In general, the performance resulting from unilateral fitting tends to be worse than that of bilateral fitting. This finding is consistent with other studies (DiCarlo & Brown, 1960; Dermody & Byrne, 1975; Sebkova & Bamford, 1981).

Byrne et al. (1992) adapted the Noble and Byrne methodology (1990) to investigate the long-term effects of bilateral and unilateral hearing aids on localization performance. All 87 participants used personal hearing aids, and 83 out of 87 subjects had symmetrical hearing loss. The results indicated that hearing aid listeners with moderate to severe hearing loss performed better when using bilateral fitting than unilateral fitting. In contrast, hearing aid users with mild-to-moderate hearing loss indicated a similar localization performance between unilateral fitting and bilateral fitting devices. This is consistent with Stephens's et al. (1991) finding that the improved localization ability of two hearing aids tends to be more prominent on those individuals with severe-to-profound hearing impairment.

In summary, the localization performance of a unilateral hearing aid fitting tends to be worse than that of bilateral fitting, especially for those with severe hearing loss. In contrast, the difference of effects between unilateral and bilateral fitting is minor for people with mild-to-moderate hearing loss.

The impact of bilaterally independent signal processing

Bilaterally independent signal processing can interfere with localization cues in several ways. First, the ITDs may be altered due to delays between the hearing aids' respective processing of the signal. Secondly, the ILDs may be disrupted because of differences in the amplification schemes in the two hearing aids. Consequently, any signal processing that distorts the ITD or the ILD could also deteriorate localization performance.

Although relatively little research has been devoted to the effect of bilateral processing on localization, directional microphones have been the subject of many studies. For example, Van den Bogaert et al. (2006) compared the unaided localization performance and aided localization performance with omnidirectional and adaptive directional microphones for a group of BTE hearing aid users. The results indicated that the adaptive directional microphone decreased localization accuracy compared to the omnidirectional microphone. Furthermore, localization performance under the adaptive directional microphone condition was worse than that under the unaided condition. Keidser et al. (2006) measured the localization performance of individuals using BTE hearing aids equipped with directional microphones. They examined four different directional microphone combination patterns for this group of hearing aid users besides unaided condition: bilateral omnidirectional microphone, bilateral directional microphone with cardioid patterns, bilateral directional microphone with cardioid in one ear and dipole in the other, and directional microphone with cardioid pattern in one ear and omnidirectional microphone in the other. The results showed that the localization accuracy in the frontal horizontal plane decreased when the microphone patterns were not matched between ears. Chung et al. (2008) did a similar study to evaluate eight hearing impaired listeners' localization performance in a virtual environment. The results indicated that the use of bilaterally mismatched directional microphones did not negatively impact localization performance in the frontal horizontal plane. Overall, however, these studies suggest that localization performance could be reduced when the directional microphone pattern is not matched between ears.

Relative to the issue of different signal processing in the bilaterally fit hearing aids, Keidser et al. (2006) and Musa-Shufani et al. (2006) found that the use of amplitude compression in hearing aids can influence the bilateral ILDs, though the impact on localization performance is trivial.

As mentioned in Chapter 1, studies on DNR and its effect on localization are limited. To the author's knowledge, the study by Keidser et al. (2006) is the only contemporary one that has systematically investigated the impact of three types of signal processing (compression, DNR, and directional microphones) on localization performance in the horizontal plane. They found that bilaterally independent DNR significantly reduced localization accuracy, but the impact was too small to be considered clinically significant.

Recently, there has been a trend toward combining bilateral signal processing and the preservation of localization cues for the purpose of improving localization performance or even speech perception. Although these techniques have not been utilized in commercial hearing aids, the related studies have shown some promising results. For example, Van den Bogaert et al. (2008) compared the localization performance of two new DNR techniques (binaural multichannel Wiener filtering and binaural multichannel Wiener filtering with partial noise estimation) to that of adaptive directional microphones. Both new DNR techniques were based on a statistical Wiener filter approach. They differed in terms of whether or not part of the unprocessed signal was added to the processed signal (through Wiener filtering). The task for the subjects was to identify the position of both speech and noise in the frontal horizontal plane. The results indicated good localization performance for the speech signal using binaural multichannel Wiener filtering, and a good localization performance for both speech and noise using binaural multichannel Wiener filtering with a partial noise estimate. However, the localization performance of the adaptive directional microphone was significantly worse than those of the new DNR techniques.

In summary, bilaterally independent signal processing can negatively interfere with localization performance, although it is unclear if the magnitude of the effect is clinically significant. It is possible that future bilateral signal processing could preserve localization cues and help improve localization performance for hearing aid users.

Speech Hearing – Speech Perception in Spatially-separated Noise

As early as the 1950's, Hirsh described a phenomenon in which it was easier to listen to speech when the noise source was spatially separated from the speech signal as compared with when the noise and speech were from the same location (i.e., co-located). Since then, a large body of research has shown that benefits from separation exist in varied situations including when the competing noise is a second talker or multi- talkers (e.g., Hawley, Litovsky, & Colburn, 1999; Yost, Dye, & Sheft, 1996).

Cherry (1953) advocated that listeners can make use of differences in the spatial locations of sources to help separate a speech signal from competing noises. The head shadow effect was implicated as one of main reasons for the beneficial effect of separation when there was only one single noise source (Bronkhorst & Plomp, 1988). The benefit due to separation could be reduced if the multiple competing noise sources are symmetrically positioned in opposite directions from a speech signal. However, the symmetrical separation of speech and noise can still benefit listeners due to momentary fluctuations of the SNR in a particular ear (Bronkhorst & Plomp, 1992; 1993).

One piece of evidence to support the notion that people can benefit from spatial separation is the binaural masking level difference (BMLD). BLMD implies that the signal detection can be improved in the dichotic (different stimulus between ears) listening condition compared to the diotic (same stimulus between ears) condition. If the interferer has a different phase or level relationship between ears than the target signal, the detection of the target can be enhanced. If the BMLD occurs in the free field, it is referred to as “spatial release from masking (SRM).” That is, the perception of the target speech is improved when the target signal and interferer are spatially separated in the free field.

It has been found that listeners can take advantage of binaural cues (ITDs and ILDs) from separated sound sources and realize this SRM. The magnitude of SRM is usually calculated as the difference in speech reception thresholds between spatially-separated and co-located maskers, which can be as high as 16dB (e.g., Good et al., 1997). The term “spatial benefit” has also been used to refer to the SRM (e.g., Dubno, Ahlstrom, & Horwitz, 2002; Hawley, Litovsky, & Culling, 2004).

Benefit from Spatial Separation

Three factors contribute to the benefit of spatial separation: binaural summation, binaural squelch, and better ear effect. Listeners can benefit from binaural summation if the auditory system can combine (or sum) similar information from two ears. The redundancy can only be useful if sounds are audible in both ears. “Binaural squelch” effect refers to the way the binaural auditory system makes use of interaural differences among received sounds. The brain has the ability to combine the sounds from two ears and build a central binaural spectrum. In effect, the brain compares the central binaural spectrum to the spectrum at each ear and subtracts the “unwanted” noise from the combined spectrum, resulting in a better SNR of the combined binaural spectrum. Binaural summation and binaural squelch are dominant at low-frequency regions, and these two effects are often combined and referred to as binaural analysis. Finally, listeners can take advantage of head shadow and benefit from better-ear effect. Because the head produces a barrier to the transmission of high frequency sounds, the acoustical energy builds up at the side of the head facing the sound source whereas it is blocked from reaching the other side of the head. The level of unwanted noise may be reduced at certain frequencies as a result of the head shadow, which could further improve the SNR at one ear more than the other. Thus, listeners can make use of the head shadow by ‘listening’ at each ear, deciding which ear has the better SNR, and ignoring the other ear. This so-called “better-ear” effect is dominant in the high frequency area. It has been shown that the effect of head shadow starts at 1500 Hz and peaks between 2000 to 5000Hz (e.g., Festern & Plomp, 1986). It should be noted that listeners with monaural hearing can benefit from the better-ear effect as well.

Hawley, Litovsky, and Culling (2004) evaluated the benefit of spatial separation in a virtual cocktail party environment for a group of normal-hearing listeners. Four kinds of background noise were utilized: speech, reversed speech, speech-shaped noise, and speech-shaped modulated noise. Each noise was either co-located or spatially separated from the target sentences at 0° azimuth. Speech performance was tested bilaterally and unilaterally in the left ear under the condition of each background noise. The total spatial benefit was defined as the speech performance difference between the bilaterally separated and co-located conditions and consisted of the advantages due to the better-ear effect and binaural analysis. The advantage due to the better-ear effect was defined as the difference in speech performance between the unilaterally separated and co-located conditions for the left ear. The advantage due to the binaural analysis was calculated by subtracting the better-ear advantage from the total spatial benefit. The results showed that the advantage due to the binaural analysis was about 2-4 dB when only one source of background noise was presented regardless of the type or the location of the masker. The same result was found for the speech-shaped noise or speech-shaped modulated noise when two- or three-source maskers were presented. The advantage due to the binaural analysis reached 6-7dB when the masker was speech or reversed speech for the multiple- masker conditions. In addition, the advantage due to the better-ear effect was found to be similar to that due to the binaural analysis.

However, this better-ear advantage could reduce or disappear if the multi-source maskers are distributed on both left and right side of the target.

Because binaural cues (ITDs and ILDs) provide the basis for people to benefit from separated sound sources and receive SRM, it would be interesting to investigate the relative contribution of these two cues. Bronkhorst and Plomp (1988) examined the independent contributions of ITD and ILD cues to speech recognition in noise for a group of normal-hearing listeners in a virtual environment. The speech signal was presented from a 0 degrees azimuth and the noise was varied from a 0 to 180 degrees azimuth. Three kinds of noise were used: “dT” noise (provided ITD cues but no ILD cues), “dL” noise (provided ILD cues but no ITD cues), and “FF” noise (contained both ITD and ILD cues). The spatial benefits were measured for each noise condition. When listening in the “FF” noise from a 90 degree azimuth, the spatial benefit was around 10.1dB; when listening in the “dL” noise from a 90 degree azimuth, the spatial benefit was around 7.8dB; when listening in the “dT” noise from a 90 degree azimuth, the spatial benefit was around 5.0dB. The authors concluded that ILD cues were dominant for deriving the spatial benefit but the effects of using ILD and ITD cues seemed not additive.

It should be noted that a number of researchers have advocated that SRM results not only from the better-ear effect or from binaural analysis. Higher-level processing may be involved as well, such as selective attention (e.g., Gatehouse & Noble, 2004; Kidd, Mason, Rohtla, & Deliwala, 1998; Marrone, Mason, & Kidd, 2008a). Hearing-impaired listeners might have more problems listening in a complex environment when selective attention is needed. For example, Gatehouse and Noble (2004) reported that “having a conversation with one person when many people are talking” or “talking with one person and following TV” were the most difficult situations and most correlated with handicap due to hearing loss. Therefore it is vital to explore the relative importance of attention focused on a point in space (selective attention). Better-ear effect and binaural analysis may sufficiently explain the spatial benefit when the background noise brings energetic masking. However, an additional masking, i.e., informational masking, occurs when the speech signal and the masker (competing speech) are both audible but the listener cannot isolate the signal from the masker. One factor that could influence multi-talker speech perception is the similarity between the target and the masker, which includes the gender of the talker and the masker. Therefore, when the background noise is another talker, selective attention may play a more important role than the better-ear effect and/or binaural analysis. That is, if the listener has a priori knowledge of the target and knows where to direct his or her attention, the listener may show improved hearing performance in noisy contexts (e.g., Kidd, Arbogast, Mason, & Gallun, 2005).

Knowing the location of the target has been shown to enable listeners to improve their speech performance in a cocktail party environment (e.g., Arbogast & Kidd, 2000; Ericson, Brungart, & Simpson, 2004; Kidd, Arbogast et al., 2005; Shinn-Cunningham & Ihlefeld, 2004). Kidd et al. (2005) tested four normal-hearing listeners in a sound field. The stimuli were from the Coordinate Response Measure (CRM) corpus (Bolia, Nelson, Ericson, & Simpson, 2000), and were presented through three loudspeakers (0°, +/- 60°). Three sentences were presented simultaneously but separately from three separated loudspeakers. Only one sentence was assigned as the target sentence and two other sentences were considered maskers. The probability of the location of the target sentence was pre-determined and the subjects were notified before each trial about the probability of the target location. The "call sign" was the key to determine which sentence was the target sentence. It was given in two ways: One was given one second before the stimulus presentation (call sign before), and the other was given after the stimulus presentation (call sign after). The "call-sign before" indicated that subjects had prior knowledge about the target sentence whereas the "call-sign after" did not. The results indicate that the accuracy was greater with the "call sign before" than that with the "call-sign after". However, when the location probability was high (~ 1.0), the performance was similar between the two “call sign” conditions. Accuracy decreased for both “call sign” conditions when the location probability was lowered. Singh, Pichora-Fuller, and Schneider (2008) found a similar advantage from knowing the target location for both young and old listener groups except that the old group indicated poorer performance compared to the young group.

In contrast, Jones and Litovsky (2008) investigated the impact of masker predictability (not target predictability) on speech perception performance in a cocktail party environment for a group of normal-hearing listeners. All the listeners were informed of the location of the speech target in advance, but not the exact number or locations of the maskers during the experiment. Rather, the listeners were instructed regarding the probability of the masker configuration being presented in each trail. No significant differences in speech performance were found between the varying predictability of masker configurations. In other words, the results suggested that knowing the maskers’ number or locations beforehand did not influence the outcome of the speech test when the location of the target was known. Fan, Streeter, and Durlach (2008) also found that the spatial uncertainty of maskers had a relatively small effect on non-speech detection in noise. Interestingly, Brungart and Simpson (2007) found that varying the predictability of both target and masker locations simultaneously had a significant effect on speech performance.

In real life, listeners may have to switch their attention back and forth if they are talking to a group of people. Therefore, localization could be one of the attributes needed to separate multiple auditory sources and help attend to target sounds (e.g., Bregman, 1990; Drennan, Gatehouse, & Lever, 2003; Darwin & Hukin, 2000a, 2000b).

The Impact of Hearing Impairment on Spatial Benefit

It is well-documented that hearing impairment can hinder people benefiting from the spatial separation of speech and noise. The loss of audibility, reduced spectral and temporal resolution, or a wider auditory filter may all affect the ability of hearing-impaired listeners to make effective use of cues to segregate the target from noise. As a result, hearing-impaired listeners may experience smaller spatial benefit than normal-hearing listeners.

Bronkhorst and Plomp (1989) explored the independent contribution of ITD and ILD cues to spatial benefits and evaluated the impact of the degree and configuration of hearing loss on spatial benefit in a virtual environment. A similar methodology was adapted from Bronkhorst and Plomp (1988). The results showed that the spatial benefit of the hearing-impaired group with symmetrical hearing loss was 2.7- 4.1 dB less than that of the normal hearing group. The group with asymmetrical loss was found to have 2.2-7.2 dB less spatial benefit compared to the normal-hearing group. The spatial benefit due to the ITD was insensitive to the hearing impairment, which suggests that hearing-impaired listeners can almost always benefit from timing cues if the signal level is above hearing thresholds. Furthermore, the authors concluded that the spatial benefit results mainly from using the ILD cues. The loss of audibility may be interfered with the use of ILDs for hearing-impaired listeners. It was found that the hearing threshold at 4 kHz was positively related to spatial benefit. That is, the worse the hearing threshold at 4 kHz, the less the binaural advantage. Later, Bronkhorst and Plomp (1992) evaluated the effect of spatially separated multiple speech-like maskers on binaural speech recognition in a virtual environment. A similar result was found in which hearing-impaired listeners benefited less from the spatial separation of speech and noise than normal-hearing listeners. The small spatial benefit observed in hearing-impaired listeners, especially in those with high frequency hearing loss, was believed to be due mainly to the inability to make effective use of the ILD cues.

Recently, Dubno et al. (2002) investigated the relative contributions of different frequency regions to spatial benefit as a function of speech and noise's high-pass and low-pass cutoff frequency in three groups of listeners: young normal-hearing, old normal-hearing, and old hearing-impaired. The results indicated that the spatial benefits for full-frequency range stimuli were 6.1dB, 4.9dB, and 2.7dB for the respective groups. When listening in the low-pass filtered stimuli, the spatial benefit for all three groups was slightly decreased compared to unfiltered stimuli. The spatial benefit was continually reduced to zero in the old hearing-impaired group when the low-pass cutoff frequency was continually lowered, whereas the benefit in the two normal-hearing groups was relatively stable as the cutoff frequency was lowered. When listening in the high-pass filtered stimuli, the spatial benefit in these three groups were slightly increased compared to the unfiltered stimuli. When only high-frequency information was available, the spatial benefit was increased in both two normal-hearing groups but not in the group with hearing-impairment. Dubno et al. (2002) concluded that the lack of spatial benefit in the high frequency region for the hearing-impaired group was a result of the high-frequency hearing loss. The hearing-impaired group could still benefit from low frequency information although the spatial benefit for the low-pass filtered stimuli were much lower than that for the high-pass filtered stimuli. Moreover, the authors suggested that there might be a trade-off point in terms of the contributions of timing and level cues when the mid-frequency information was being removed. The high-frequency dominated ILD cues were found to account for bigger spatial benefits than the low-frequency dominated ITD cues. When the available spectral information was in the mid-frequency range, the results of the spatial benefits were varied. In addition, the age effect was believed to play a role in the spatial benefit. The spatial benefit in the old normal-hearing group was less than that in the young normal-hearing group. Similar age effects have been reported in other studies as well (e.g., Grose, Poth, & Peters, 1994; Pichora-Fuller & Schneider, 1992).

The studies mentioned above generally support the notion that the reduced spatial benefit or small SRM for hearing-impaired listeners is due mainly to their reduced ability to take advantage of better-ear effect. However, because the spatial release from informational masking cannot be predicted based on lower-level processing, such as better-ear effect or binaural analysis, recent studies have focused on higher-level factors in more complex listening situations for hearing-impaired listeners. For example, Marrone et al. (2008a) evaluated the impact of hearing loss, age, and reverberation on spatial benefit in a complex multi-talker background. The authors specifically limited the possibility of using the better-ear effect by placing the speech maskers symmetrically around the target position. Highly similar target and maskers (same-sex talkers) were used to generate informational masking rather than energetic masking. The results suggested that all participants, including age matched normal-hearing and hearing-impaired listeners, benefit from the spatial separation of speech and background noise. The younger normal-hearing group showed the greatest spatial benefit while the older hearing-impaired listeners had the least. The authors concluded that the SRM resulted mainly from a reduction in informational masking when the better-ear effect was alleviated. Therefore, it was postulated that hearing-impaired listeners had more adverse effects from the energetic masking due to their wider auditory filters (Arbogast et al., 2005) and benefited less from informational masking release. In addition, the results showed a negative correlation between speech perception thresholds in quiet and spatial benefits. This suggested that the more severe the hearing impairment, the less the spatial benefit that could be expected. In addition, the existence of reverberation was found to reduce spatial benefit for all listeners. It was interesting that the correlation between spatial benefit and age was relatively weak in their study. These investigators pointed out the possibility that old listeners may have difficulty selectively attending to one talker at the presence of competing interferences or ignoring irrelevant stimuli. (For more information regarding selective attention, see the review by Shinn-Cunningham and Best (2008).)

In summary, hearing-impaired listeners benefit less from spatial separation than normal hearing listeners. This lack of spatial release is believed to be related to the inability to take advantage of better-ear listening due to high frequency hearing loss. Furthermore, other deficits besides loss of audibility, such as suprathreshold deficits, selective attention or even age effect, could help explain the smaller spatial benefit in hearing-impaired listeners compared to normal-hearing listeners.

Because the use of a hearing aid is one remediation option for hearing-impairment, the following part will briefly discuss the advantage of bilateral fitting and the impact of bilateral hearing aids on spatial benefits.

The Advantage of Bilateral Fitting

Although fitting hearing aids bilaterally has been recommended for hearing-impaired listeners for decades (e.g., Balfour & Hawkins, 1992; Dillon, 2001; Kobler & Rosenhall, 2002; Markides, 1980; Noble, 2006), it is unclear as to how much binaural advantage can be assumed from this fitting approach. In fact, the issue of whether two hearing aids (bilateral fitting) are better than one (unilateral fitting) has been a source of debate for over fifty years (e.g., Carhart, 1946). It could be assumed that hearing-impaired listeners who retain some degree of functionality in both ears would benefit from two hearing aids. Yet, Kochkin (2009) reported that only 74.3% hearing aid fittings in the U.S. are bilateral. It is thus possible that some people do not realize binaural benefits from their bilateral fitting.

Considerable research has been focused on the comparison of unilateral and bilateral fittings. For example, Kobler and Rosenhall (2002) evaluated the advantages and disadvantages of bilateral fitting compared to unilateral fitting in 19 hearing impaired listeners with moderate hearing loss. The participants were instructed to repeat sentences back in noise and identify which loudspeaker presented the signal. The level of speech perception performance was significantly improved in those who were fitted bilaterally compared to those with unilateral fittings. Bilateral fitting was seen to be better than unilateral fitting as well in regard to localization results. In another study (Noble & Gatehouse, 2006), hearing aid users reported that wearing two hearing aids resulted in lowered listening effort compared to the experience of wearing only one. Later, Noble (2006) reviewed 14 studies which comparing bilateral versus unilateral fitting and focused on the self-reported outcome measurements. The results from the reviewed studies were highly varied due to differences in the sample populations and methodologies being used. However, Noble found no strong evidence to support the self-reported advantage of bilateral fitting when listening in noise, especially for those who had mild to moderate hearing loss. Some investigators found that older hearing aid users may not benefit from two hearing aids compared to one due to higher-level deficits (e.g., Henkin, Waldman, & Kishon-Rabin, 2007; Walden & Walden, 2005). In addition, Gatehouse and Akeroyd (2006) point out that measurements of binaural hearing benefit taken in a traditional laboratory setting are essentially those reflecting a static environment. The results may not represent the experiences in daily life, which could explain why there is no relationship between self-reported disabilities and laboratory-measured binaural hearing capabilities. Furthermore, the phenomenon of “binaural interference” exists for some people with hearing loss, especially those with asymmetrical hearing loss. It has been found that speech perception can be worse when listening with both ears than when listening with one ear for these listeners (e.g., Allen, Schwab, Cranford, & Carpenter, 2000; Dillon, 2001). The mechanism behind this binaural interference so far has remained unclear. But it could one of the reasons that some hearing aid users cannot benefit from bilateral fitting.

Of course, whether or not a bilateral fitting is successful cannot be predicted based solely on audiometric measurements. It depends on many factors, such as personal needs, expectations from hearing aids, the existence of binaural interference, or even the impact of listening environments. Although there is no evidence supporting the absolute benefit of bilateral fitting over unilateral fitting, the trend toward employing two hearing aids is rising (Kochkin, 2009). Therefore, it is vital to discuss the impact of bilateral hearing aids on spatial benefit.

The Impact of Bilateral Hearing Aids on Spatial Benefit

Because bilateral hearing aids typically operate independently, timing and level cues could be distorted by the independent signal processing applied (e.g., Dillon, 2001; 2003; Keidser et al., 2006, Van Den Bogaert et al., 2006; Kalluri & Edwards, 2007). Levitt (1987) pointed out that the hearing aids' signal processing can make the phase responses different between ears, which could alter the ITD cues for hearing aid users. Therefore, hearing aid users may not make effective use of distorted spatial cues, which is crucial in the SRM. In other words, hearing aids could interfere with spatial cues and result in reduced spatial benefit for hearing aids users. On the other hand, the amplification which compensates for the loss of audibility may help hearing-impaired listeners benefit from the better-ear effect (Ahlstrom, Horwitz, & Dubno, 2009). So far, very few studies have directly investigated the impact of bilateral hearing aids on spatial benefit (Festen & Plomp, 1986; Kalluri & Edwards, 2007; Marrone, Mason, & Kidd, 2008b).

Marron et al. (2008b) evaluated the spatial benefits for a group of bilateral hearing aid users and age-matched normal-hearing listeners in two reverberant conditions. The target was from the front, and the maskers were either co-located (i.e., from the same location) in relation to the target or symmetrically situated on the left and right sides of the target. Listeners were tested in the unaided condition and wearing their personal hearing aids, both in the bilateral modes. The presentation level was arbitrarily chosen at 30dB SL for all conditions for the hearing aid users. The results showed that the unaided spatial benefit was about 1~2 dB better than in either of the aided ones. The spatial benefit was in average 1dB more in the bilateral aided than in the unilateral aided conditions. The younger listeners tended to benefit more from the spatial separation than the older listeners. The difference between unaided, bilateral aided, and unilateral aided conditions for the older listeners was therefore less robust than that for the younger listeners. In addition, it was found that the reverberation had a negative impact on the spatial benefit. Although the spatial benefit was smaller in the unilateral aided than in the bilateral-aided conditions, it suggested that listeners can still benefit from the unaided ear to receive the spatial release.

The results from another study that implemented bilaterally independent compression algorithms for a group of normal-hearing listeners indicated that the spatial benefit was reduced in a condition containing only ILD cues (Kalluri & Edwards, 2007). Nevertheless, compression had little impact on spatial benefit when both ITD and ILD cues were available. This finding suggests that bilaterally independent compression might not interfere with the ITD cues. Therefore, listeners can rely on ITD cues to benefit from spatial release.

Because bilateral hearing aids have the potential to distort binaural cues, it is natural that more and more investigators have an urge to develop a system for bilateral signal processing in hearing aids that preserves binaural cues (e.g., Drennan, Gatehouse, Howell, Van Tasell, & Lund, 2005; Kollmeier et al., 1993; Van den Bogaert et al., 2008; Van den Bogaert, Doclo, Wouters, & Moonen, 2009). Some of studies have shown promising results. Drennan et al. (2005) found that phase-preserving amplification slightly improved individuals' speech performance in spatially separated noise. Most recently, Van den Bogaert et al. (2009) found that a combination of a contralateral microphone signal and an ipsilateral microphone signal in a bilateral multichannel Wiener filtering noise reduction algorithm resulted in improved spatial benefits in normal-hearing listeners.

In summary, hearing-aid users can benefit from spatial separation of the target and masker. Although bilateral signal processing's effect on spatial benefit remains inconclusive, it seems that one of the future aims of the use bilateral hearing aids is to preserve natural spatial cues as much as possible.

The Relationship between Localization and Speech Perception

Since Hirsh (1950) started exploring the relationship between localization and speech perception in spatially separated noise, it remains unclear as to whether listeners need good localization abilities in order to attend to the target source in a cocktail party environment. It has been suggested that the use of spatial locations is one way to form auditory images and allow an individual to segregate the target talker from others and direct the attention to it (e.g., Marron et al., 2008a). However, Drullman and Bronkhorst (2000) pointed out that the effect of spatial separation was not related to absolute localization ability. Spatial separation can contribute significantly to speech perception even when the absolute localization of targets is rather poor. It is possible that spatial benefit does not require intact spatial cues to extract useful information. At the same time, localization accuracy depends mainly on how well listeners can make use of spatial cues.

Most studies of speech perception in spatially separated noise have not measured localization performance. Very few studies have directly addressed the relationship between spatial hearing and speech hearing (e.g., Noble et al., 1997; Noble et al., 1995). Noble et al. (1997) performed a comparison of hearing speech in noise, detection of spatial separateness (i.e., judgments of whether a pair of stimuli were generated from different sources or at the same sources), and localization in a group of subjects with hearing loss. A modest relationship was found between spatial benefit and the detection of spatial separateness. However, no relationship was suggested between absolute signal localization ability and spatial benefit. Furthermore, Noble and colleagues found that once the effect of audibility was controlled, listeners with poorer localization ability tended to require a more positive SNR to achieve 50% speech intelligibility. In addition, Good et al. (1997) provided evidence that detection of a non-speech signal in noise and localization in noise were not strongly related.

In another study, Noble et al. (1995) used an inventory to evaluate the relationship between self-reported speech hearing and localization disability. The partial correlations after controlling for the loss of audibility between the localization and speech hearing subscale scores were significant although the range of correlation was wide (.32~.51). In contrast, Noble et al. (1997) found a weak negative correlation between localization error and spatial benefit, even when the loss of audibility was accounted for. As mentioned in Chapter 1, several other studies have indicated that some hearing-impaired listeners with reduced localization abilities also did poorly on speech perception tasks in noise, and others with normal localization abilities exhibited good performances on speech perception tests. In contrast, some listeners who had good speech perception were shown to have poor localization performance (Sebkova & Bamford, 1981; Kubo et al., 1998). Additionally, Hawley, Litovsky, and Colburn (1999) found that in normal-hearing listeners speech perception performance was not impacted by localization ability. Their findings support the notion that listeners may take advantage of any cues besides spatial cues to have good speech perception, these cues including the differences in sound level between talkers or the fundamental frequency of the talker's voice.

Given the results of the above studies, at least some data support the idea that a modest relationship between localization and speech "hearing" does exist. However, the connection between localization and speech perception ability remains unclear. No evidence is available from which to infer a causal relationship between localization and speech perception in noise in spite of the same cues (ILD and ITD cues) being required for these two tasks.

Table 1. Modulation frequency (Hz) and modulation depth (dB) comparison among clean speech, speech babble, and jet noise

	Modulation frequency (Hz)	Modulation depth (dB)
Clean speech	4-8	35-50
Speech babble		15-20
Jet Noise	>30	5

Table 2. Noise reduction schemes across seven hearing aid manufactures

Hearing aid manufacturers	Noise reduction schemes
Widex	modulation detection
Starkey	modulation detection
GNResound	modulation detection
Sonic Innovations	modulation detection
Siemens	modulation detection
	Wiener filter
Phonak	modulation detection
Oticon	comodulation detection

CHAPTER 3

METHODS

In this study, three major research questions were pursued: 1) Will speech perception and/or localization performance in spatially-separated noise in the frontal-horizontal plane be negatively affected when hearing aid users are given a bilaterally mismatched gain reduction scheme as compared to an unaltered bilaterally linear time-invariant amplification scheme? 2) Will the deteriorated speech perception and/or localization performance in noise due to the bilaterally mismatched gain reduction scheme be restored by a bilaterally matched gain reduction scheme? 3) Do different gain reduction patterns result in different sound quality and/or listening effort ratings for hearing aid users? To answer these questions, a group of hearing aid users were tested in a virtual environment with insert earphones. All stimuli were pre-programmed using the head-related-transfer-function (HRTF) of the Knowles Electronics Manikin for Acoustic Research (KEMAR) to simulate a virtual environment without hearing aids. The purpose of this programming was to include natural head shadow effects resulting in natural alteration in timing and spectrum between two ears. The pre-programmed stimuli were pre-filtered based on the pre-determined gain reduction patterns measured in our lab. The hearing loss of each listener was compensated for using a simulated linear amplification through MATLAB 7. Data were gathered for 1) localization in noise in the frontal-horizontal plane, 2) speech perception in spatially-separated noise, and 3) subjective listening-effort ratings and subjective sound quality ratings. The impact of the different bilateral gain reduction patterns was tested in two modes: mismatched (gain reduction in one ear only) and matched (same gain reduction between ears). The mismatched results of localization and speech perception were compared to those obtained with a bilaterally-fit, linear time-invariant amplification scheme (i.e., a reference scheme) wherein audibility was optimized. The mismatched results were also compared to the matched results. Various gain reduction patterns with two different background noises were studied to determine the optimal gain reduction pattern in terms of sound quality and listening-effort ratings. In addition, we studied the relationship between localization and speech recognition results. Finally, self-reported hearing deficits were assessed using an inventory when wearing personal hearing aids in daily life.

Subjects

Ninety six hearing aid users were recruited for the study; 24 (14 females; 10 males) qualified and participated in the study with the following inclusion criteria: 1) adult bilateral hearing aid users with age older than 20; 2) at least one year experience with hearing aids; 3) bilateral sensorineural hearing loss (air-bone gap < 10 dB); 4) hearing symmetry (i.e., an interaural threshold difference in hearing level of less than 15dB across the frequencies 0.5, 1, 2 and 4kHz; 5) hearing threshold levels no better than 20 dB HL at 500 Hz and no worse than 75 dB HL at 3000 Hz (re: ANSI, 1996); 6) normal tympanogram; 7) normal cognitive function; 8) normal reading visual acuity after correction; 9) native speaker of English. The exclusion criteria were: 1) unilateral hearing aid users; 2) rising or severe, sharply sloping configurations of hearing loss. Mean thresholds are displayed in Figure 1. Ages ranged from 45 to 81 years with the median age of 63 years old. Subjects were paid for their participation. Ten subjects with normal hearing (5 females; 5 males) participated in the localization experiment as well. The age range for the normal hearing group was 23-65 years old. The hearing thresholds were better than 20 dB HL from 250- 8000Hz (re: ANSI 1996).

Gain Reduction Patterns Determination

To determine the typical gain reduction patterns used in current DNR algorithms, preliminary data from seven commercially available behind-the-ear (BTE) hearing aids from seven leading manufacturers were gathered (Table 3). Each model represented the company’s premier hearing aid model at the time. All models were multichannel digital hearing aids. They were programmed to fit a flat 50 dB HL hearing loss (250 through 8000Hz) using the

National Acoustical Laboratory- Nonlinear 1 (NAL-NL1) prescriptive method (Dillon, 2001). All adaptive features other than digital noise reduction were disabled. The compression parameters were enabled based on the manufacture's default, or first-fit setting. A single passage (nine sentences) from the Connected Speech Test (CST) (Cox, Alexander, & Gilmore, 1987; Cox, Alexander, Gilmore, & Pusakulich, 1988) was used as the input stimulus. Background noises consisted of two types: 1) six-talker babble and 2) steady-state-speech-shaped noise. To determine the impact of the DNR algorithms, the SNR of the speech and noise was set to four pre-determined conditions: +5, 0, -5, and -10dB, respectively. The overall input levels of the speech were fixed at 65, 75, and 85 dB SPL and the noise was varied to achieve those SNRs. A Bruel and Kjaer (B&K) test box via a 2cc coupler was utilized to obtain a gain versus frequency response curve for each hearing aid. The outputs were compared between two conditions of DNR (on and off). From those measures we calculated the difference. Because no level effect (i.e., same results for different levels of input) was observed during pilot testing, except for one hearing aid that showed more low frequency gain reduction at higher input levels, the gain frequency response for 65dB SPL input was chosen to be used to determine the gain reduction patterns for this study.

We determined that the gain reduction patterns could be grouped into four categories when the background noise was multi-talker babble: no change[2], low-frequency (<500Hz-1000Hz) gain reduction, both low- and high-frequency gain reduction (particularly in the low frequency range), and low-frequency gain reduction with high-frequency gain boost (Figure 2, A-D). The gain reduction patterns for the steady-state-speech-shaped noise could be grouped into two categories: both low- and high-frequency gain reduction by the same amount, and more low- and less high-frequency gain reduction (Figure 3, A-B). The

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[2] Because this gain reduction pattern does not have any gain changes across frequencies, it was not used as a filter in the present study.

subsequent gain alterations used in the present study were based on these values, and these gain patterns were later achieved through the following MATLAB programming:

First, the values of the gain reduction patterns were averaged from each grouping (Table 4). Second, seven to eight points of the average gain reduction values were chosen from the Table 4 and applied in the MATLAB 7 to generate filters which simulate those gain reduction patterns. The values between points were interpolated. Any points below 125Hz or above 8000Hz were maintained the same as the ending point. After the stimuli were programmed through the MATLAB, the measurement was taken again through the B&K test box via a 2cc coupler to confirm that the desired gain reduction was achieved.

Stimuli Preparation

Because the pinnae, head, and torso reflect sounds and produce a direction-dependent filter for sounds from all directions, it is possible to present HRTF-filtered stimuli over headphones to produce the illusion that sound sources are "externalized" (i.e., outside the listeners' heads) in actual space (e.g., Begault & Wenzel, 1993; Wenzel, Arruda, Kistler, & Wightman, 1993; Wightman & Kistler, 1989a; 1989b). Because this method of sound delivery simulates free-field presentation, it can create a "virtual" environment for listeners. An obvious advantage of this use is that one can easily manipulate acoustical cues under desired conditions.

Several studies have compared the difference between sound field and virtual environments. For example, Wenzel et al. (1993) found that participants' localization accuracy was similar for the horizontal plane, regardless of whether stimuli presentation was from a true free field over loudspeakers or over headphones. However, more errors were observed in distinguishing front from back when using headphones than free field listening. Middlebrooks (1999) reported that fewer-than-average localization errors were observed when individual HRTFs were used whereas Kawaura et al. (1991 as cited in (Moore, 2003)) found that the performance was not significantly different for the two conditions. Drullman and Bronkhorst (2000) evaluated both speech perception and localization performance between individualized HRTFs and general HRTFs when the stimuli were band-limited to below 4KHz, and no significances were found. Because all the experiments took place with stimuli presented in the frontal horizontal plane in the present study, we determined that non-individualized HRTFs would not significantly (or negatively) impact the outcome measures.

In the present study, the stimuli were synthesized with the purpose of preserving all of the natural acoustic cues for speech perception and sound localization that are present with free-field stimuli. A group of HRTF electronic files measured by Gardner and Martin (1994; 1995) from a KEMAR head under anechoic conditions were applied in the present study to simulate the spatial locations. The HRTF files consist of left/right responses from a distance of 1.4 meter in the horizontal plane. There are 128-point impulse responses in total. The sampling rate was 44.1 KHz. Applying the HRTF corresponding to the proposed direction can bring in the natural combination of ITDs and ILDs into that stimulus.

Before all the stimuli were convolved with the KEMAR's HRTFs, they were programmed through MATLAB to imitate the pre-determined gain reduction patterns. Twelve conditions of test were generated (Table 5).

Simulated Amplification and Fitting Procedures

The simulated hearing aid amplification used in this study was achieved through MATLAB programming. The prescriptive formula of NAL-RP (National Acoustic Laboratory - Revised, Profound) (Dillon, 2001) was applied to provide the linear amplification for subjects. To make sure the output met the NAL-RP target, probe microphone measures of the ear-canal output were obtained for each subject. The participant sat in front of a probe microphone system (Verifit system 3.4.16) and faced the speaker of the system, at a distance of 90cm. The probe microphone tube was placed into the open ear canal, with the reference microphone faced outwards. Pink noise from the probe microphone system was used as the stimulus and presented at 65dB SPL. The unaided values at the eardrum were measured and entered into a pre-designed Excel file to calculate the "target" output for each subject.

A pair of ER-2 insert earphones with flat frequency responses within +/- 1~ 2dB in both ears (see Appendix A) was used to transduce the signal. The "dandelion passage" (#76) from the CST was presented to the subject through the insert earphones from a Dell personal computer. This passage had been filtered and amplified through MATLAB so that the gain and frequency response was shaped to compensate for the hearing loss for each individual subject. The equipment set-up is shown in Figure 4 except that the participant was sitting outside of the soundbooth during the fitting stage. Due to the output limitation of the ER-2 insert earphones, the gain at 3000 Hz was restricted; consequently, the target gain could only be achieved for hearing loss up to 70 dB HL. After filtering for gain, the speech passage was played back to the subject through the insert earphones. The output was measured using the probe microphone system. The measured output was compared to the "target" output (NAL-RP); if the root-mean-square (rms) error across frequency range between 500 Hz to 4000Hz was within 5dB, the initial fitting for one ear was considered to be appropriate. The same steps were carried out in the opposite ear. If the rms error was more than 5dB, further MATLAB programming was used to adjust the output. Finally, a stimulus set made up of five everyday sounds (alarm, bell, child laughing, guitar, and telephone ringing) was used to verify that the sound level at the eardrum (as measured by the probe microphone) was appropriate for the localization experiment as well.

When the output targets were met for both the "dandelion passage" and the everyday sounds, the stimuli were again presented back to the subject bilaterally. The subject was asked to rate the loudness comfort using the Independent Hearing Aid Fitting Forum (IHAFF) loudness scale (Valente & Van Vliet, 1997) for each ear. The overall level was adjusted, if necessary, according to the loudness comfort judgment using the audiometer dial. If the subject responded that any stimulus was “too loud”, the overall level was reduced by 1dB per step in each ear until the sound was rated comfortable and equally loud in both ears. Multiple steps of fine tuning were carried out, as needed. The output was reduced 3dB for both ears because of the bilateral fitting rule for the NAL-RP formula. At the completion of the fitting stage, each subject rated the volume (separately across ears) to be between 4 (comfortable) and 5 (comfortable but loud). For detailed fitting steps, please refer to Appendix B.

The fitting for the normal-hearing group was pre-determined, and based on a flat 50dB HL hearing loss. The same final tuning steps were carried out for this group to make sure that each stimulus was rated comfortable and equally loud in both ears.

Outcome Measures

In order to investigate the impact of bilateral gain reduction on speech perception and localization in spatially-separated noise, one speech recognition test and one localization test were administered to each subject in each of the test conditions. In addition, the sound quality ratings and listening-effort ratings were collected during speech recognition testing. A questionnaire to evaluate the self-reported hearing deficits in daily life with personal hearing aids was administered as well.

Speech Recognition Test – Connected Speech Test

The Connected Speech Test (CST) was designed to simulate everyday speech with contextual information (Cox et al., 1987; Cox et al., 1988). The CST is based on fixed presentation levels and SNRs. This test includes 24 pairs of speech test passages produced by a female talker. The task of the listener is to repeat the sentences as heard. Each passage has seven to ten sentences, with a total of 25 key words. The final score is based on how many key words are correctly repeated. In the present study, two pairs of test passages (i.e., 100 key words) were used for each condition.

The SNR chosen for testing was set at -5 dB for two reasons: 1) most noise reduction algorithms take effect when the SNR is -5dB; 2) Pearsons, Bennett, and Fidell (1976) reported the real-world SNRs in relatively noisy environments to be +4 dB to – 1dB. This -5 dB SNR was slightly poorer than what Persons et al. reported. The SNR conditions in the present study were achieved by varying the background noise levels with a fixed 65 dB SPL speech level.

Each subject was placed in the center of a double-wall, sound-treated IAC booth at 0° azimuth to the center of an arc hanging inside the booth (refer to Figure 4). The diameter of the arc was 1m; the height of the arc was ear level for the seated subject. The speech and noise signals were pre-mixed and played from a Dell personal computer with a LynxTwo sound card, routed via a GSI60 audiometer and a SAMSON Servo 120 amplifier (Figure 4), and then routed back to the audiometer and presented to the subject from the ER-2 insert ear phones. In the present study, the speech signal was always from the front (0° azimuth) and the background noise was always from the right side (+90° azimuth).

To decrease the negative impact of learning, and to minimize the order effect, a Latin Square Design was used in which each CST test set was paired with each condition only once and each CST test set occupied a particular place in the order only once (Table 6).

Listening-effort Rating

The Borg Category Ratio scale (Borg-CR10) was developed to measure the perceived effort, exertion, or pain in physical work (Borg, 1998). In the present study, it was adapted to measure the perceived listening effort after each speech task. The Borg-CR10 scale is a continuous rating scale which combines verbal descriptors and numbers. Listeners can use any number from 0 (i.e., no perceived effort at all) to 10 (i.e., extremely strong perceived effort) to rate the listening effort that they had experienced during the CST test. If the participants perceived more effort and wanted to use a higher number than 10, they could rate the effort outside of that range, e.g., 11 or higher. Refer to Appendix C for the detailed instruction. Each participant was given a sheet of paper with the scale and asked to orally state the number corresponding to the listening effort after each condition.

Sound Quality Rating

When listeners judge sound quality of some signals in background noise, they often have uncertainty about whether the reduced quality of the sound results from signal distortions, background noise intrusiveness, or both (International Telecommunication Union, 2003). This uncertainty may add additional error variance to the ratings, which could compromise the reliability. The International Telecommunication Union (ITU) addressed this issue and developed the ITU-T P.835 method (2003) to evaluate the sound quality based on speech alone, noise alone, and overall effect. The first two ratings in the tool use a five-point scale based on either the signal distortion (SIG) or the background intrusiveness (BAK). The last rating uses a Mean Opinion Score (OVRL) to rate the overall effect. This method was adopted in the present study. Refer to the Appendix D for detailed instructions.

After finishing the CST test, each participant was asked to rate the sound quality of a speech sample for each filter condition. Each sample was comprised of three sentences from the CST test passages. The listeners were asked to rate one sentence a time based on the speech signal only, the background noise only, and the overall effect. The test order of the "speech signal only" and the "background noise only" was counterbalanced. Finally, they were asked to rate the last sentence based on the "overall effect."

Localization in Noise Test

For the localization task, five everyday sounds were chosen from the original 16 stimuli from Dunn, Tyler, and Witt (2005). They were: telephone ring (Phone), buzzer

(Alarm), guitar playing (Guitar), train-crossing warning (Bell), and child laughing (Child). For each condition, nine loudspeakers emitted five stimuli with two repetitions. The total repetitions were 90 (9*5*2). That is, each loudspeaker would emit five stimuli twice (5*2) and each stimulus would be repeated 18 times (9*2). Randomization of order was based on 90 repetitions. The five stimuli were defined on four parameters (Warner & Bentler, 2002): low cutoff frequency, high cutoff frequency, frequency of the primary peak, and number of peaks (Table 7). The low cutoff frequency is defined as the frequency which the amplitude is 30dB down from the maximum on the 1/3 octave band spectrum toward the low-frequency end whereas the high cutoff frequency is measured at 30dB down from the maximum amplitude toward the high-frequency end. The frequency of the primary peak is measured at the maximum amplitude, and the peak is defined as the place where the amplitude is 10dB higher than the encircling troughs. The five stimuli were chosen to be representative of a wide range of these acoustic parameters (see Appendix E for spectrum and time waveform graphs of these five stimuli).

The subject was placed in the center of a double-walled, sound-treated IAC booth at 0° azimuth to the center of an arc placed inside the booth. The diameter of the arc was 1m; the height of the arc was ear level for the seated subject. Because a virtual environment was used for the localization experiment, no real loudspeakers were required. Instead, nine paper symbols of numbers were used to indicate the position of loudspeakers from -60° azimuth to +60° azimuth on the arc. Each everyday sound could come from any of the nine loudspeakers, while the background noise was always from +90° azimuth. Everyday sounds and noise were pre-programmed and played from a Dell personal computer with a LynxTwo sound card, routed via a GSI60 audiometer and a SAMSON Servo 120 amplifier, and then routed back to the audiometer and presented through a pair of ER-2 insert earphones. The subject was seated in front of a touch screen and asked to pinpoint the location of the loudspeaker with a stylus. The subject was asked to face 0° azimuth and not to move his/her head before the sound was playing. A web camera inside the booth was utilized to monitor any head movement. The order of the stimuli and the location of the loudspeaker were both randomized within a session. Before the localization-in-noise test was administered, each subject was tested in quiet for four to five sessions in order to be familiarized with the task. The data from the final practice session were also used as a baseline for each subject in terms of localization performance in quiet. The normal-hearing subjects were tested in quiet to provide normative data for this localization experiment as well.

Because the present study was focused only on the frontal horizontal plane, the target and response azimuths were only in the left/right dimension (-90° to +90°) (Good & Gilkey, 1996). There are various error measures applied in the localization studies (e.g., Noble & Byrne, 1990; Good & Gilkey, 1996; Lorenzi et al., 1999a, b; Keidser et al., 2006; Van den Bogaert et al., 2006). Three measures of localization performance were used in the present study:

1) The root-mean-square (rms) error

$$\text{rms error} = \sqrt{\frac{\sum_{i=1}^{n} \left|\text{Response}_i - \text{Target}_i\right|^2}{n}}\ .$$

"n" in the formulae corresponds to the number of presentation per loudspeaker used in the study. The rms error indicates the difference between the perceived angles and target angles. The average rms error is the rms error averaged across loudspeakers. The possible range of the average rms error in the present study was varied from 0° (perfect localization) to 55.43° (the chance level due to random guessing). The average rms error reveals the overall localization accuracy. The higher the rms error, the worse the localization accuracy. The systematic bias contributes to the average rms error as well as the random variability in listener's responses.

2) The mean signed error

$$\text{mean signed error} = \frac{\sum_{i=1}^{n}\left(\text{Response}_i - \text{Target}_i\right)}{n}.$$

The mean signed error suggests the direction of the systematic bias. A negative error indicates a bias toward the left and a positive error indicates a bias toward the right. The error could equal to 0 if the listener displays perfect localization or random guessing.

3) The proportion of variance accounted for (r^2)

Good and Gilkey (1996) first introduced the proportion of variance accounted for (r^2) as an outcome measure for localization studies. The r^2 is defined as the proportion of variance in perceived loudspeaker azimuth that is accounted for by the best-fitting linear relationship with the target loudspeaker azimuth. Lorenzi et al. (1999b) recommended using r^2 to indicate the variability of subjects' responses, which is a measure of localization consistency for confirmatory purposes. Values from 0 to 1 can occur; the higher the value, the better the localization consistency.

Speech Spatial Qualities (SSQ) Questionnaire

The Speech, Spatial and Qualities of hearing Scale (SSQ) was designed to evaluate three domains of self-reported hearing deficits, including "speech, spatial, and other qualities of hearing" (Noble & Gatehouse, 2004). The three domains comprise 50 items. The "speech hearing" domain has 14 items; the "spatial hearing" domain has 17 items; the "other qualities" domain has 19 items. The score for each item is obtained using a scale from 0 to 10. The higher the score, the greater the ability. Refer to Appendix F for the SSQ items. Since the purpose of this test administration was primarily to understand real-world localization abilities using personal hearing aids, the subjects were asked to fill in the SSQ questionnaire at home and to turn it in when they returned. We used a pencil-to-paper form in the present study. Subjects had the option to indicate if they did not use hearing aids in the situations described in the questionnaire.

Procedures

Once the subject agreed to participate in the study, a total of three visits were scheduled:

Visit 1

After the subject consented to participate in the study, hearing status was measured by pure-tone audiometry and tympanometry. Cognitive function was screened by the Mini Mental State Exam (Crum, Anthony, Bassett, & Folstein, 1993). Visual acuity was screened by a Snellen eye chart. In addition, the information of date of birth and native language was obtained. If the subject was eligible to enroll the study, the fitting stage commenced. At the end of the visit, the subject was given the SSQ questionnaire to fill in at home and was asked to turn it in during the following visit. Before the subject returned for the next visit, all the stimuli needed for the experiments were pre-programmed.

Visits 2 and 3

The order for the speech recognition test and localization test was counterbalanced across the 24 subjects with hearing loss. The normal-hearing group (n = 10) did only the localization experiment.

For the speech recognition test, the subject was seated in a double-walled sound booth and listened to the stimuli through a pair of ER-2 insert earphones. The task of the subject was to repeat sentences back as accurately as possible. Practice CST passages were given prior to the actual data gathering to let the subject get familiar with the process. Once the subject felt comfortable with the testing, the formal test began. The subject could take as many breaks as he or she wanted during the experiment. After finishing each condition of the CST experiment, the subject was asked to give the rating corresponding to the listening effort for that condition using the Borg-CR10 scale. When all the test conditions were completed, the subject was asked to give the sound quality rating for each condition using the ITU-T P.835 method. This visit took approximately 2-3 hours.

For the localization in noise task, the subject was seated in front of an arc from -60° azimuth to +60° azimuth in the center of the same sound booth. The arc was at the ear level of the subject. A touch screen was also placed in front of the subject. Each stimulus was presented to the subject using the ER-2 insert earphones. The subject needed to indentify which loudspeaker the sounds were coming from by using the touch screen. The subject was given four to five practice sessions (without noise) to get familiar with the task. Once the subject felt ready, the formal test began. Subjects were given breaks as needed. This task took approximately 2-3 hours.

Once the subject finished visit 2, visit 3 was scheduled. All visits were scheduled on different dates depending on the subject’s schedule. The total time needed to finish all visits ranged from one to four weeks. The subjects were paid for their participation.

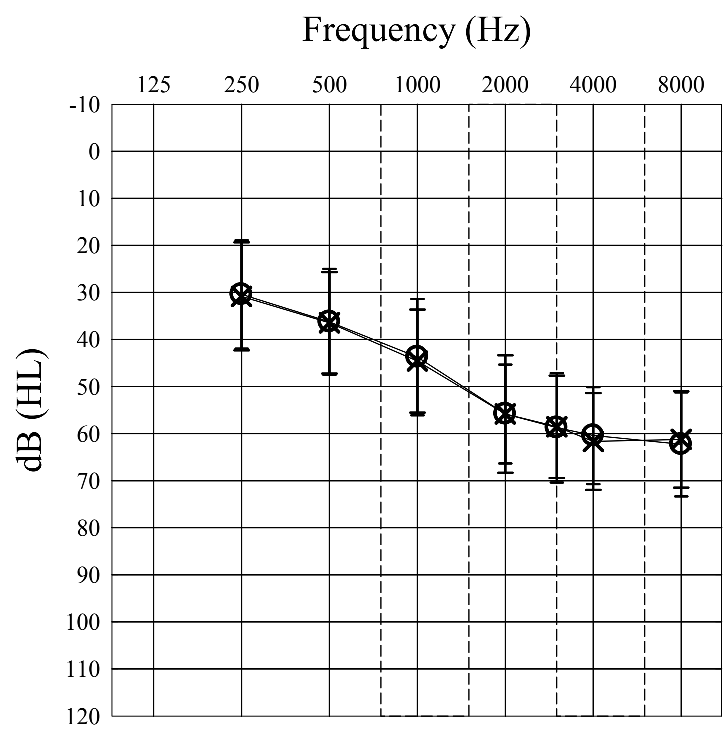

Figure 1. Mean thresholds for the left and right ears. The error bar stands for one standard deviation.

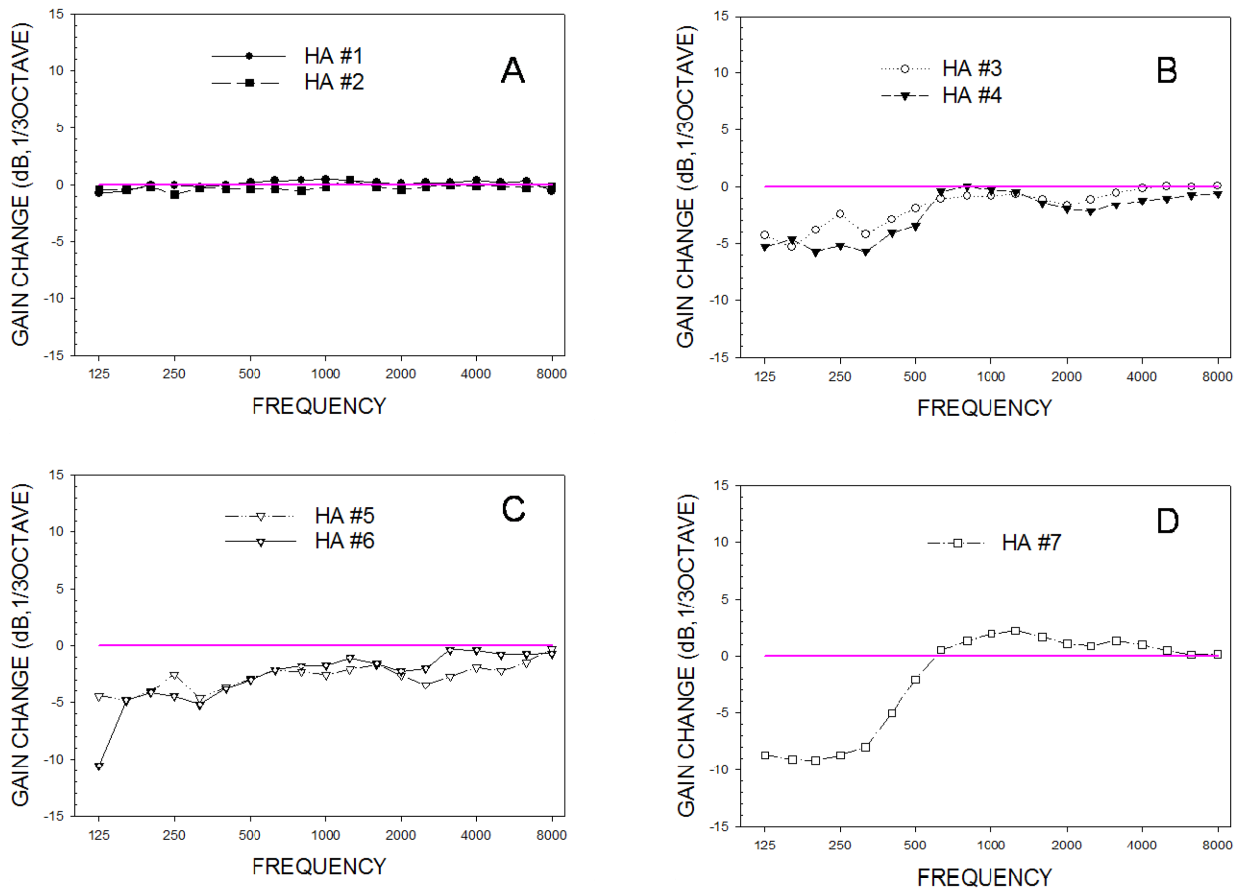

Figure 2. Panels A-D show four different gain reduction patterns when the background noise was multi-talker babble. Gain change refers to the differences between digital noise reduction being activated and deactivated. Panel A indicates no measurable gain change; panel B indicates low-frequency (<500Hz-1000Hz) gain reduction; panel C shows both low- and high-frequency gain reduction (particularly in the low frequency range); panel D indicates low-frequency gain reduction with high-frequency gain boost.

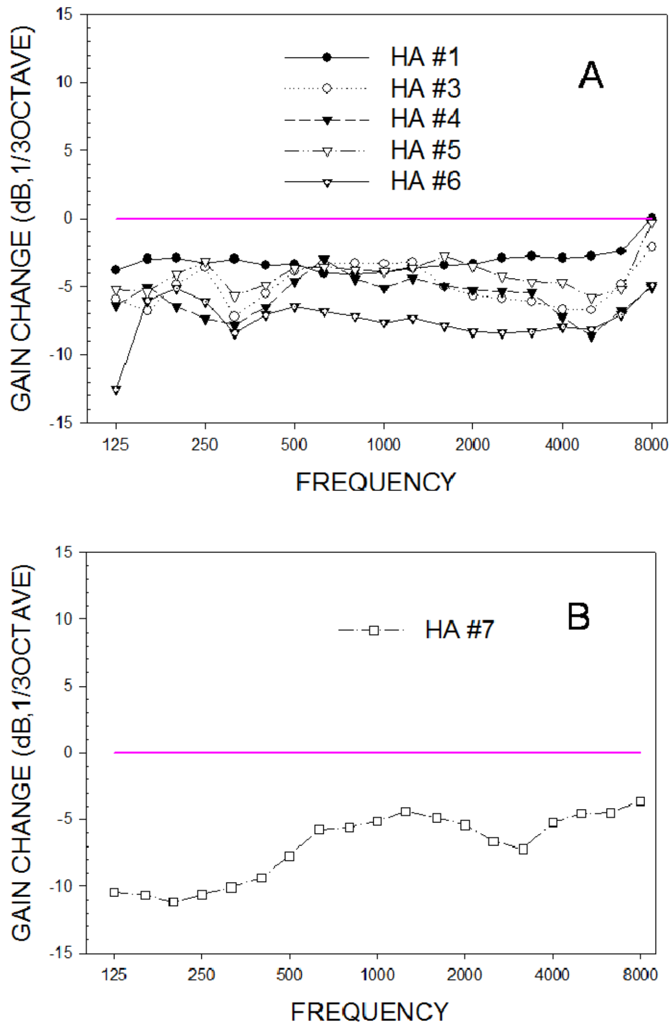

Figure 3. Panels A and B show the gain reduction patterns for the steady-state-speech-shaped noise. Gain change refers to the gain differences between digital noise reduction being activated and deactivated. Panel A shows that the amount of gain reduction is similar between low- and high-frequency regions; Panel B indicates the gain reduction exists in both low- and high-frequency regions (particularly in the low frequency range).

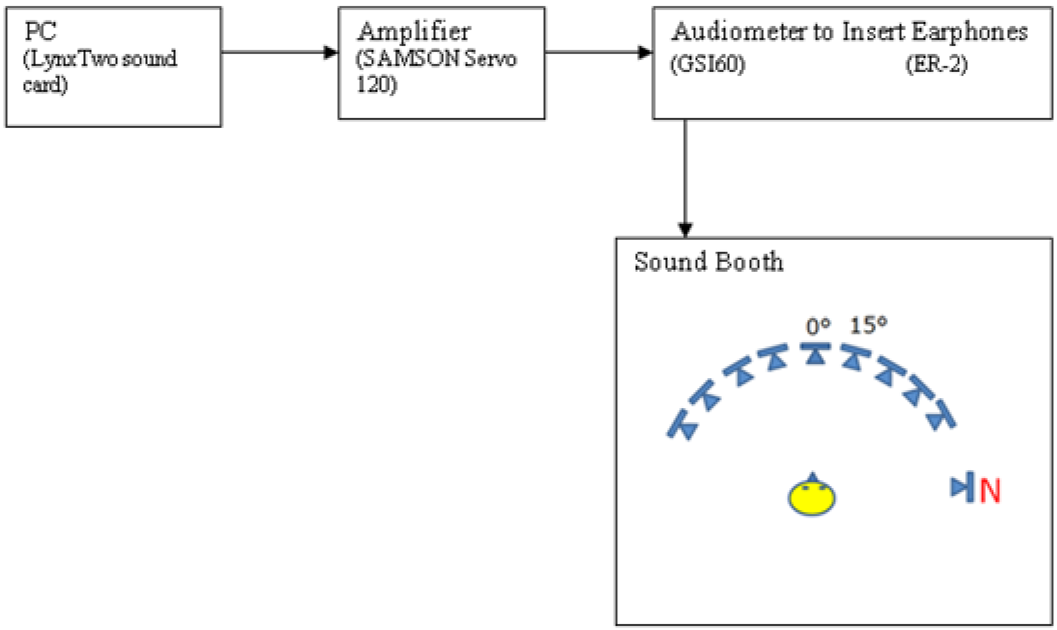

Figure 4. Experimental set-up diagram.

Table 3. Hearing aid make and model for derived gain reduction patterns

Hearing Aid #	Make and Model
1	Oticon Syncro BTE
2	Phonak Savia Art 211 dSZ BTE
3	Siemens Centra S BTE
4	GN ReSound Metrix MX70-D
5	Starkey Destiny 1200 DaVinci 13 BTE
6	Widex Aikia AK-19 BTE
7	Sonic Velocity BTE

Table 4. Summary of the gain reduction values used in the MATLAB

	Gain Reduction (dB)				
	Babble	Babble	Babble	SSN	SSN
Frequency (Hz)	Ptn1	Ptn2	Ptn3	Ptn4	Ptn5
125	-5	-7	-9	-7	-10
160	-5	-5	-9	-5	-11
200	-5	-4	-9	-5	-11
250	-4	-4	-9	-5	-11
315	-5	-5	-8	-7	-10
400	-3	-4	-5	-6	-9
500	-3	-3	-2	-5	-8
630	-1	-2	1	-5	-6
800	0	-2	1	-5	-6
1000	-1	-2	2	-5	-5
1250	0	-2	2	-5	-4
1600	-1	-2	2	-5	-5
2000	-2	-2	1	-5	-5
2500	-2	-3	1	-5	-7
3150	-1	-2	1	-5	-7
4000	-1	-1	1	-5	-5
5000	0	-1	0	-5	-5
6300	0	-1	0	-4	-4
8000	0	-1	0	-2	-4

Note: SSN stands for the speech-shaped noise. “Ptn” refers to the pattern.

Table 5. Descriptions for all conditions

Condition #	Name	Description	Folder Name
C1	Linear_Babble	Linear amplification in multi-talker babble (no gain reduction exists)	filter0_babble
C2	Linear_SSN	Linear amplification in speech-shaped-noise (no gain reduction exists)	filter0_ssn
C3	Mismatched_Pattern1_Babble	Only right side has low frequency gain reduction, in multi-talker babble	filter1~m
C4	Mismatched_Pattern2_Babble	Only right side has gain reduction (less gain reduction in low frequency but more in high frequency area), in multi-talker babble	filter2~m
C5	Mismatched_Pattern3_Babble	Only right side has gain reduction (low-frequency gain reduction with high frequency gain boost), in multi-talker babble	filter3~m
C6	Mismatched_Pattern4_SSN	Only right side has gain reduction (same amount across the whole frequency range), in speech-shaped noise	filter4~m
C7	Mismatched_Pattern5_SSN	Only right side has gain reduction (less gain reduction in low frequency but more in high frequency area), in speech-shaped noise	filter5~m
C8	Matched_Pattern1_Babble	Both sides have low frequency gain reduction, in multi-talker babble	filter1
C9	Matched_Pattern2_Babble	Both sides have gain reduction (less gain reduction in low frequency but more in high frequency area), in multi-talker babble	filter2
C10	Matched_Pattern3_Babble	Both sides have gain reduction (low-frequency gain reduction with high frequency gain boost), in multi-talker babble	filter3
C11	Matched_Pattern4_SSN	Both sides have gain reduction (same amount across the whole frequency range), in speech-shaped noise	filter4
C12	Matched_Pattern5_SSN	Both sides have gain reduction (less gain reduction in low frequency but more in high frequency area), in speech-shaped noise	filter5

Note: The Folder name column displays the name used in the MATLAB programming.

Table 6. The Latin square design of the CST test

	S1	S2	S3	S4	S5	S6	S7	S8	S9	S10	S11	S12
G1 (Sub2&16)	C1	C2	C3	C4	C5	C6	C7	C8	C9	C10	C11	C12
G2 (Sub4&15)	C2	C3	C4	C5	C6	C7	C8	C9	C10	C11	C12	C1
G3 (Sub11&18)	C3	C4	C5	C6	C7	C8	C9	C10	C11	C12	C1	C2
G4 (Sub9&21)	C4	C5	C6	C7	C8	C9	C10	C11	C12	C1	C2	C3
G5 (Sub7&23)	C5	C6	C7	C8	C9	C10	C11	C12	C1	C2	C3	C4
G6 (Sub12&19)	C6	C7	C8	C9	C10	C11	C12	C1	C2	C3	C4	C5
G7 (Sub8&22)	C7	C8	C9	C10	C11	C12	C1	C2	C3	C4	C5	C6
G8 (Sub10&14)	C8	C9	C10	C11	C12	C1	C2	C3	C4	C5	C6	C7
G9 (Sub1&20)	C9	C10	C11	C12	C1	C2	C3	C4	C5	C6	C7	C8
G10 (Sub6&17)	C10	C11	C12	C1	C2	C3	C4	C5	C6	C7	C8	C9
G11 (Sub3&24)	C11	C12	C1	C2	C3	C4	C5	C6	C7	C8	C9	C10
G12 (Sub5&13)	C12	C1	C2	C3	C4	C5	C6	C7	C8	C9	C10	C11

Note: The "S1" stands for the first set of the CST. The "G1" stands for the first subgroup which has subjects #2 and #16.

Table 7. The properties of five everyday sounds

	LowFreqCutoff (Hz)	HighFreqCutoff (Hz)	FreqPrimPeak (Hz)	Npeaks
Guitar	80	5000	200	2
Child	250	5000	800	1
Phone	250	6300	1250	2
Alarm	200	12500	1600	5
Bell	80	10000	2500	3

Note: Refer to the text for the details.

CHAPTER 4

RESULTS

Recall that we had two hypotheses in this study: 1) Hearing aid users will have inferior performance in both localization and speech perception in spatially-separated noise for a bilaterally mismatched gain reduction scheme as compared to a reference scheme (an unaltered bilaterally linear time-invariant amplification scheme without any gain reduction); and 2) Hearing aid users will have superior performance in both localization and speech perception in spatially-separated noise for a bilaterally matched gain reduction scheme (same gain reduction between ears) as compared to a bilaterally mismatched gain reduction scheme (gain reduction in the right ear only). The results from the speech perception experiment did not support either hypothesis. In fact, the opposite was found: the hearing aid users had deteriorated speech performance for the bilaterally matched gain reduction compared to the bilaterally mismatched gain reduction. The results from the localization experiment did support both hypotheses. That is, the localization accuracy with bilaterally mismatched gain reduction schemes was worse than that with the reference schemes whereas the deteriorated localization accuracy due to bilaterally mismatched gain reduction schemes was restored using the bilaterally matched gain reduction schemes. The results of the sound quality ratings indicated that gain reduction (matched or mismatched) can help reduce the perception of background noise intrusiveness for listeners with hearing loss. However, the use of neither gain reduction scheme (matched or mismatched) reduced listening effort for listeners with hearing loss compared to the reference schemes.

Speech Perception in Spatially-separated Noise

One purpose of this study was to evaluate the effect of gain-reduction synchrony (mismatched and matched) and gain reduction patterns (filters) on speech perception in spatially-separated noise. Twelve simulated hearing aid conditions (2 levels of Synchrony x

5 levels of Filter + 2 filter-off referents) were tested. Two types of background noise (multi-talker babble and speech-shaped noise) were embedded in the filters as those were the backgrounds in which the filter effect was measured. That is, Filters 1, 2, and 3 were tested in the multi-talker babble whereas Filters 4 and 5 were tested in the speech-shaped noise. As a result, the type of background noise was not used as a variable in the data analysis. The two filter-off conditions referred to the conditions using the reference schemes (i.e., the linear amplification conditions without gain reduction in either ear), and served as a measure of baseline for both background noises. The other ten conditions included five mismatched filter-on conditions (Filters 1, 2, 3, 4, and 5 activated only in the right ear) and five counterpart matched filter-on conditions (Filters 1, 2, 3, 4, and 5 activated in both ears).

All 24 hearing-impaired subjects were tested under the 12 simulated hearing aid conditions mentioned above. Recall that a Latin square design was applied to decrease the negative impact of learning, and to minimize the order effect. Each CST test set was paired with each condition only once and each CST test set occupied a particular place in the order only once. The measurement of the CST raw scores (in percent correct) was repeated, since all subjects were exposed to each condition in turn.

The raw scores (in percent correct) of each condition for each subject are displayed in Figure 5. The raw scores were transformed into rationalized arcsine units (rau) to homogenize the variance (Studebaker, 1985). Subsequently, the CST score (rau) change from the filter-off to filter-on conditions was calculated. The CST score changes were analyzed using a repeated measures analysis of variance (ANOVA) to determine the effects of Synchrony and Filter (SAS 9.1.3). Follow-up contrast tests were conducted and two adjustment methods were chosen to control the Type I error rate (Bonferroni correction and Tukey-Kramer correction) for multiple comparisons in this study, as appropriate. The CST test set was included in the analysis as a covariate so that the variance due to the effect of the test set was not treated as a random error. In addition, a t-test was applied to test if the CST score (rau) differences between two filter-off conditions were significantly different from zero. The results indicated that subjects performed worse in the multi-talker babble than in the speech-shaped noise with a difference of -8.78 rau (t_{23} = -4.27, $p < .0001$).

Effect of Synchrony

Statistical analysis indicated a significant main effect for Synchrony ($F_{1, 200}$ = 5.50; p = .02). That is, the CST score (rau) change from the filter-off conditions to the mismatched filter-on conditions was significantly different from the change from the filter-off to the matched filter-on conditions. Because the interaction between the Synchrony and Filter was not significant ($F_{4, 196}$ = 1.10; p =.36), the interaction was not included in the final analysis. The non-significant interaction suggests that the CST score (rau) change between the mismatched and matched conditions was not different when using different filters.

Because the main effect of Synchrony was significant, a series of follow-up analyses was performed to test the difference between the filter-on and corresponding filter-off conditions. Figure 6 shows the CST rau differences between the filter-on (mismatched) and filter-off conditions, as well as the differences between the filter-on (matched) and filter-off conditions. No significant difference was found between the mismatched and filter-off conditions (t_{200} = -0.58; adjusted $p > .99$) or between the matched and filter-off conditions (t_{200} = -2.85; adjusted p = .056). A Bonferroni correction was applied here to adjust the p values. However, because the Bonferroni correction is known to be conservative (e.g., Perneger, 1998), the adjusted p value ".056" may underestimate the significance. That is, it is possible that the speech performance under the matched conditions was actually worse than that under the filter-off conditions. Indeed, further analysis revealed that the use of Filters 3, 4 and 5 in the matched conditions resulted in poorer speech performance than that of the filter-off conditions (t_{200} = -2.99 with adjusted p = .038, t_{200} = -2.92 with adjusted p = .045, and t_{200} = -4.08 with adjusted p = .0008 respectively). In summary, the results imply that the speech performance of the mismatched conditions was better than that of the matched conditions.

Effect of Filter

There was a significant main effect of Filter on the CST rau change ($F_{4, 200} = 5.61$; $p = .0003$). A post hoc analysis indicated that the CST rau change for Filter 1 was significantly different than that for Filter 5 ($t_{200} = 3.17$; adjusted $p = .02$). The CST rau change for Filter 2 was also significantly different than those for Filters 3, 4, and 5 ($t_{200} = 3.02$ with adjusted $p = .02$, $t_{200} = 2.96$ with adjusted $p = .03$, and $t_{200} = 4.01$ with adjusted $p = .0008$, respectively). Other post hoc comparisons were not significant. The method for adjusting p-values to control the Type I error rate was the Tukey-Kramer adjustment method. The results suggest that the CST change for both Filter 1 and 2 was greater than that for Filter 5. (Note: We cannot say the *performance* of Filter 1 or 2 was better than that of Filter 5 because the baseline performance was different.)

In summary, the findings did not support either Hypothesis #1 or #2 in terms of impacting speech perception in spatially-separated noise. Relative to Hypothesis #1, the subjects did not show poorer speech perception performance for a bilaterally mismatched gain reduction scheme, as compared to the reference scheme. Relative to Hypothesis #2, the results support the opposite of this hypothesis. That is, the performance was reduced for a bilaterally matched gain reduction scheme as compared to a bilaterally mismatched gain reduction scheme. In other words, the use of bilaterally mismatched gain reduction did not result in a worse or better speech performance for listeners with hearing loss, compared to the performance with the reference scheme. At the same time, matching the gain reduction scheme between ears resulted in deteriorated speech performance, compared to both the reference scheme and the mismatched gain reduction scheme.

Sound Localization

Localization performance was quantified by three statistics: 1) rms errors; 2) the proportion of variance accounted for (r^2); and 3) mean signed errors. Recall 24 hearing aid users participated in the study, and 10 listeners with normal hearing served as a control group to ensure that the virtual localization task was valid. Subjects were instructed to locate the loudspeaker from which the sound was presented. All the listeners finished the experiment under 12 simulated hearing aid listening conditions as used in the CST experiment, each of which was accompanied by five types of everyday sounds (Alarm, Bell, Child, Phone, and Guitar) as stimuli, and two kinds of background noises (multi-talker babble and speech-shaped noise) from the +90° azimuth. A repeated measures ANOVA was applied to analyze the main effect of Synchrony, Filter, and Stimulus, as well as the interaction between these factors for the rms error measurements.

Hearing-impaired Group

Response patterns

The individual baseline response patterns (tested in quiet) for subjects with hearing loss, are shown in Figure 7. The response azimuth is plotted as a function of the target azimuth. The area of each filled circle is proportional to the number of responses made at that azimuth for any given target azimuths by the subjects. Any point lying on the diagonal line indicates an ideal localization performance. It is noted that all subjects performed well below the chance level (55.43°) in this study, although the individual response patterns suggest large between-subject variability. For example, some subjects (e.g., Subjects #1 and #13) have responses close to the diagonal line; others (e.g., Subjects #2, #3, and #15) have responses that diverged from the diagonal line. Some response patterns (e.g., Subjects #4, #6, and #16) have an "S" shape, which indicated that these subjects responded at further locations when target signals were moving away from 0°.

Figure 8 depicts the response patterns collapsed across 24 hearing-impaired subjects for each of the five test conditions in the speech-shaped noise, as well as the normative data (in quiet) from 10 normal-hearing subjects. The data for the conditions tested in the multi-talker babble were not displayed, because no significant differences were found among these conditions (see below). The area of each filled circle is proportional to the number of responses made at that azimuth, for any given target azimuths, by the subjects. The normative data obtained from the subjects with normal hearing show that most responses are close to the major diagonal of the panel, which indicates relatively accurate localization performance. The response patterns of the mismatched schemes (middle two panels) indicate a bias toward the left side (-60°), compared to those of the reference scheme (left lower panel) and matched schemes (right two panels).

<u>rms errors</u>

The rms error of each condition for each subject in the hearing-impaired group is shown in Figure 9. All the subjects performed well below the chance level (55.43°) in the present study. The rms error change from the filter-off to filter-on conditions was calculated, as well, and a repeated measures ANOVA was applied to test the main effects of Synchrony, Filter, and Stimuli, as well as the interaction between these variables. The results indicated that the main effects of Synchrony and Filter were both significant ($F_{1,\,1147} = 24.73$ with $p <$ .0001, and $F_{4,\,1147} = 5.56$ with $p = .0002$, respectively), whereas the main effect of Stimulus was not significant ($F_{4,\,1147} = 1.39$ with $p = .23$). The interaction between Filter and Synchrony was significant ($F_{4,\,1147} = 7.67$; $p < .0001$), as well as the interaction between Filter and Stimulus ($F_{16,\,1147} = 2.30$; $p = .0025$). Another t-test was applied which found there was no significant difference between two reference schemes ($t_{23} = 0.90$ with $p =.38$). The follow-up analyses with the Bonferroni correction to adjust the p-values were applied to answer specific questions, since significant main effects were found.

The results of the follow-up analyses indicated that the rms error under the mismatched conditions with Filters 4 and 5 being activated, was significantly different than that under filter-off conditions ($t_{1147} = 3.49$ with adjusted $p = .0061$, and $t_{1147} = 5.81$ with adjusted $p < .0001$, respectively). The results suggest that the localization accuracy of the mismatched conditions with Filters 4 or 5 on, was worse than that of filter-off conditions. No significant differences were found between the matched and filter-off conditions (all adjusted p values were greater than .05). Furthermore, the rms error under the matched conditions with Filters 4 and 5 was significantly different than that under the mismatched conditions ($t_{1147} = -3.96$ with adjusted $p = .0009$, and $t_{1147} = -6.19$ with adjusted $p < .0001$, respectively).

The results indicated that the localization accuracy of the matched conditions was better than that of the mismatched conditions. Figure 10 (right panel) shows the mean rms error difference between the mismatched and filter-off conditions, as well as the difference between the matched and filter-off conditions, for each filter. As far as the comparison between filters is concerned, no significant differences were found within matched conditions (all adjusted p values were greater than .99). Table 8 displays the rms error difference comparisons across filters within mismatched conditions. The rms errors with Filters 4 and 5 were significantly different than those with other filters (all adjusted p values were less than .05); no difference was found between Filter 4 and 5 (adjusted p value = .56). The rest of the follow-up tests did not indicate any significant differences (all adjusted p values were greater than .05).

Because the interaction of Filter and Stimulus was significant, the average rms error for each stimulus was plotted as a function of filter (Figure 11, Panel A). Although the main effect of Stimulus was not significant, the stimulus “Guitar” appeared to result in the poorest localization accuracy across all test conditions, while the stimulus “Alarm” seemed to result in the best localization accuracy in most test conditions.

In summary, the results suggest that using the bilaterally mismatched gain reduction schemes can reduce localization accuracy compared to the reference schemes. The deteriorated localization accuracy due to the bilaterally mismatched gain reduction schemes was restored, when given bilaterally matched gain reduction schemes.

The proportion of variance accounted for (r^2)

We calculated r^2, the proportion of variance, in perceived loudspeaker azimuth that is accounted for by the best-fitting linear relationship with the target loudspeaker azimuth. The r^2 was calculated per subject for each condition using linear regression. The use of r^2 in this study was for confirmatory purposes to indicate localization consistency while the use of rms error was the primary measure of localization accuracy.

These r^2 values for the hearing-impaired group ranged from 0.53 to 0.93. A value of zero indicates random guessing while a value of one indicates perfect localization (Note: Under some circumstances, the values could be one while the localization performance is not perfect). Figure 12 shows the r^2 as a function of filter for each level of Synchrony, which reveals large between-subject variability. Generally speaking, the localization consistency under the mismatched conditions is less than that under the matched conditions within subjects. Furthermore, the results suggest that the listeners who have better localization accuracy (lower rms errors) also have greater localization consistency (higher r^2 values). For example, Figure 13 and 14 display the results of r^2 and rms error for two individual listeners. The localization accuracy and consistency of Subject #5 were both worse than those of Subject #16. In addition, the localization accuracy and consistency of the mismatched conditions were worse than those of the matched conditions for both subjects.

Mean signed errors

Recall that the mean signed error suggests the direction of systematic bias. A negative error indicates a bias toward the left and a positive error indicates a bias toward the right. The error could equal to 0 if the listener displays a perfect localization or random guessing.

The data of the mean signed error for each condition are shown in Figure 15. It was noted that the direction of the systematic bias under all conditions was toward the left side, except one mismatched condition for Filter 3 and one matched condition for Filter 4. Because the interfering noise was always on the right side, the results indicated that most responses are biased away from the location of the interfering noise. In addition, it was found that most systematic bias of the matched conditions was "pulled back" to the right ear (i.e., less bias toward the left), compared to the mismatched conditions.

<u>Correlations between the localization accuracy and hearing thresholds</u>

Pearson correlations were calculated between the rms errors and hearing thresholds. A one-way ANOVA was applied and indicated there were no significant differences in hearing thresholds between ears ($F_{1,\,328} = 0.03$, $p = .8628$). Therefore, both better ear hearing thresholds and right ear hearing thresholds were used for the correlation calculation. The values of r were all near zero, indicating localization accuracy was weakly correlated with hearing impairment.

<u>Correlations between the localization accuracy and the speech recognition performance</u>

Pearson correlations were calculated between the rms errors and CST rau for all 12 conditions. All the values of r were less than 0.4 and none of them revealed any statistical significance (all p values were less than .05). The results indicated that the speech recognition performance was only weakly correlated with the localization accuracy in the current study.

SSQ

Subjects were asked to rate their daily listening experiences with their own hearing aids using the three domains of the Speech, Spatial and Qualities of Hearing Scale (SSQ): speech hearing, spatial hearing, and qualities of hearing, respectively. Subjects could indicate if they did not use hearing aids in the situations described in the questionnaire. The score ranged from 0 (“minimal ability”) to 10 (“complete ability”) (Gatehouse & Noble, 2004). Higher scores correlated with better self-reported ability.

The mean scores and standard deviations of SSQ items are shown in Table 9. The scores are ordered from the lowest to highest within each domain. First, the scores in the speech hearing domain were generally lower than in other domains. The lowest rating was for talking with a group of people without visual cues. The highest rating was for talking to one person in a quiet environment. Second, the lowest rating in the spatial hearing domain was distance perception of voice/footsteps or vehicles. To the contrary, subjects were able to lateralize a talker from the left or the right with the least difficulty (i.e., highest rating). Finally, subjects reported that they needed to concentrate hard and spent a lot of effort in conversation (lowest ratings) in the qualities of hearing domain. The subjects reported that they were able to easily distinguish familiar music or to judge another person’s mood from voice (highest ratings).

Correlations between the SSQ
and the laboratory test results

Because the SSQ scores were not normally distributed, the ranks in the Spearman's rho were used to get the Pearson partial correlation coefficients with localization test results and better ear hearing thresholds. Only the localization results from the filter-off conditions were used to calculate the correlation, because, although the Synchrony status of bilateral hearing aids in daily life for the subjects was unknown, we assumed it was not likely engaged. The analysis indicated weak correlations between filter-off conditions and the spatial hearing scores ($r = 0.36$ for the babble; $r = 0.50$ for the speech-shaped noise with both p values greater than .05), and no significance was found. The results suggest that the correlation between self-reported localization ability and measured localization performance is relatively weak.

A better ear four frequency average (4FA) was also derived, and its relationship to SSQ items was considered. The Pearson partial correlation between speech hearing items and better 4FA was weak, and no significance was found ($p > .05$). However, 10 out of 17 spatial hearing items showed relatively strong correlations with significance to better ear 4FA (Table 10). The results suggest that listeners with greater hearing impairment generally reported greater disability on the items in the spatial hearing domain.

Normal-hearing Group

The results indicated that normal-hearing listeners listening through a simulated hearing aid processing had more localization accuracy and consistency than the hearing aid users; their patterns of performance were similar to those of hearing aid users in the current study.

rms errors

The rms error of each condition for each subject in the normal-hearing group is shown in Figure 16. All listeners with normal hearing performed well below the chance level. In terms of rms error in localization test, a repeated measures ANOVA was applied which found that the main effect of Synchrony, Filter, and Stimulus were all significant ($F_{1, 461} = 48.96$ with $p < .0001$, $F_{4, 461} = 7.73$ with $p < .0001$, and $F_{4, 461} = 3.82$ with $p = .0045$, respectively). The interaction between Filter and Synchrony was significant ($F_{4, 461} = 8.53$; $p < .0001$), as well as the interaction between Filter and Stimulus ($F_{16, 461} = 3.24$; $p < .0001$). The follow-up analyses with the Bonferroni correction to adjust the p-values were applied to answer specific questions, since significant main effects were found.

The results of the follow-up analyses indicated that the results of the normal-hearing group were similar to those of the hearing-impaired group. That is, the rms error under the mismatched conditions with Filters 4 and 5 was significantly different than that under the filter-off conditions ($t_{461} = 6.93$ with adjusted $p < .0001$, and $t_{461} = 6.71$ with adjusted $p < .0001$, respectively). The results indicate that the rms error under the mismatched conditions with Filters 4 and 5 was more than that under the matched conditions (Figure 10, left panel). No significant differences were found between filters within matched conditions (all adjusted *p* values were greater than .99). Table 8 displays the rms error difference comparisons across filters within mismatched conditions (values of the normal hearing group were displayed in parentheses). The rms errors with Filters 4 and 5 were significantly different than those with other filters (all adjusted *p* values were less than .05), whereas no difference was found between Filter 4 and 5 (adjusted *p* value > .99). The rest of the follow-up tests did not indicate any significant differences (all adjusted *p* values were greater than .05).

Because the main effect of Stimulus and the interaction of Filter and Stimulus were significant, the average rms error for each stimulus was plotted as a function of filter (Figure 11, Panel B). It appears that Stimulus “Guitar” resulted in the poorest localization accuracy in most conditions, especially in the conditions using Filters 4 and 5.

In summary, the results of listeners with normal hearing were consistent with those of the hearing aid users. That is, the localization accuracy was compromised with the bilaterally mismatched gain reduction schemes compared to the reference schemes. The deteriorated localization accuracy was restored with the bilaterally matched gain reduction schemes.

The proportion of variance accounted for (r^2)

The same linear regression as was used for the hearing-impaired group was applied to calculate the r^2 per subject for each condition for listeners with normal hearing. The range of

r^2 across all normal-hearing listeners with simulated hearing loss was from 0.47 to 0.91, which also reveals large between-subject variability. The normal-hearing group indicated similar results to the hearing-impaired group. That is, the localization consistency under the mismatched conditions is less than that under the matched conditions within subjects. The listeners who had more localization accuracy (lower rms errors) had higher localization consistency (higher r^2 values). For example, Subject#3 in the normal-hearing group had the poorest localization accuracy and the lowest localization consistency.

Mean signed errors

The data of the mean signed error for each condition are shown in the Figure 17. The response patterns were similar to those of hearing aid users. The direction of the systematic bias under all conditions was toward the left side, except for one mismatched condition for Filter 3. Most responses are biased away from the location of the interfering noise on the right side. Similar to listeners with hearing loss, the systematic bias of the matched conditions were also "pulled back" to the right ear (i.e., less bias toward the left), compared to the mismatched conditions.

In summary, the results of the localization experiment support the proposed two hypotheses. Relative to Hypothesis #1, hearing aid users had inferior performance in localization in spatially-separated noise for a bilaterally mismatched gain reduction scheme, as compared to the reference scheme. Relative to Hypothesis #2, the hearing aid users had superior performance in localization for a bilaterally matched gain reduction scheme, as compared to a bilaterally mismatched gain reduction scheme. The normal-hearing group performed similarly to the hearing-impaired group. Both groups had less localization accuracy and localization consistency in the mismatched conditions than in the matched conditions. The systematic bias tended more toward the left side in the mismatched conditions as opposed to the matched conditions. Self-reported localization abilities were not found to have a strong relationship with measured localization performance. Additionally, only a weak correlation was found between the localization accuracy and the speech recognition performance for listeners with hearing impairment.

Sound Quality Rating

The subjects rated the sound quality of the CST sentences for each condition based on three parameters from the ITU P.835 (2003): 1) signal distortion (SIG); 2) background intrusiveness (BAK); and 3) overall mean opinion score (OVRL). Figure 18 depicts the mean scores and one standard deviation for each subscale across all conditions. The ITU sound quality score change from the filter-off to filter-on conditions in each subscale was calculated, and a repeated measures ANOVA was conducted to determine the main effects of Synchrony and Filter, and any significant differences between conditions. The follow-up analyses with the Tukey-Kramer correction to adjust the p-values were applied as appropriate.

Effect of Synchrony

The main effect of Synchrony was not significant for either of the ITU subscales ($F_{1, 211} = 1.92$ with $p = 0.17$ for the SIG; $F_{1, 211} = 0.03$ with $p = .59$ for the BAK; $F_{1, 211} = 2.45$ with $p = .12$ for the OVRL, respectively). That is, the sound quality ratings of the mismatched and matched conditions were similar for all three ITU subscales. There was no significant difference between the mismatched/matched and filter-off conditions on either the ITU signal distortion subscale or the overall mean opinion score. However, the ratings for both mismatched and matched gain reduction conditions on the ITU background intrusiveness were significantly higher (less intrusive) than that for the filter-off conditions (both $p < .05$). Figure 19 shows the mean score differences between the filter-on and filter-off conditions for each subscale and as a function of Synchrony.

Effect of Filter

The main effect of Filter was significant for both the ITU signal distortion and background intrusiveness ($F_{4, 211} = 7.32$ with $p < .0001$ for the SIG; $F_{4, 211} = 8.86$ with $p < .0001$ for the BAK, respectively), but not for the overall mean opinion score ($F_{4, 211} = 2.15$; $p = .08$). No interaction was found between the Synchrony and Filter.

Table 11 shows the mean score difference comparisons across filters on the ITU signal distortion and background intrusiveness subscales. The ratings for Filter 4 were all higher (less signal distortion and less noise intrusiveness) than those for other filters, except Filter 5. Some of the ratings for Filter 5, in both signal distortion and background intrusiveness scales, were higher than Filters 1 and 3. No difference was found between Filter 4 and 5.

In summary, the results suggest that the use of gain reduction can reduce the perception of background intrusiveness, regardless of whether it is mismatched or matched, compared to the reference schemes (filter-off). In terms of signal distortion or overall rating, no differences were found between filter-on (gain reduction either in one ear or in both ears) and filter-off (no gain reduction in either ear) conditions. Furthermore, the results suggest that the use of Filters 4 and 5 can provide less signal distortion and less noise intrusiveness than that the other filters.

Listening-effort Rating

In this experiment, listeners were asked to use any number from 0 (i.e., no exertion at all) to 10 (i.e., the perception of exertion is extremely strong) to rate the listening effort that they perceived during the CST test. The Borg-CR10 score change from the filter-off to filter-on condition was calculated and a repeated measures ANOVA was conducted to determine the main effects of Synchrony and Filter, and any significant differences between conditions.

The follow-up analyses with the Tukey-Kramer correction to adjust the p-values were applied, as appropriate.

Results showed a significant main effect for Filter ($F_{4, 211} = 2.74$; $p = .04$) but not for Synchrony ($F_{1, 211} = 0.11$; $p = .74$). No interaction between Filter and Synchrony was found. There was no difference between filter-on (gain reduction either in one ear or in both ears) and filter-off (no gain reduction in either ear) conditions.

Using the Tukey-Kramer adjustment for significance, follow-up tests showed no significant differences for any of the comparisons across filters. In order to show the average listening-effort ratings for each filter, the data of mismatched and matched conditions were collapsed to depict the mean Borg-CR10 score as a function of filter, since the main effect of Synchrony was not significant (Figure 20). The mean Borg-CR10 scores were around 5 (strong) or higher in the multi-talker babble, whereas they were less than 5 in the speech-shaped noise. This finding suggests that subjects generally reported more perceived listening effort in the babble than in the speech-shaped noise. Although no significant differences were found, the use of Filter 3 resulted in the highest average rating of listening effort among five filters, while the ratings for Filters 4 and 5 were relatively low compared to others. Finally, there was a trend that the listening-effort ratings were slightly lower in the filter-off conditions than those in the filter-on conditions.

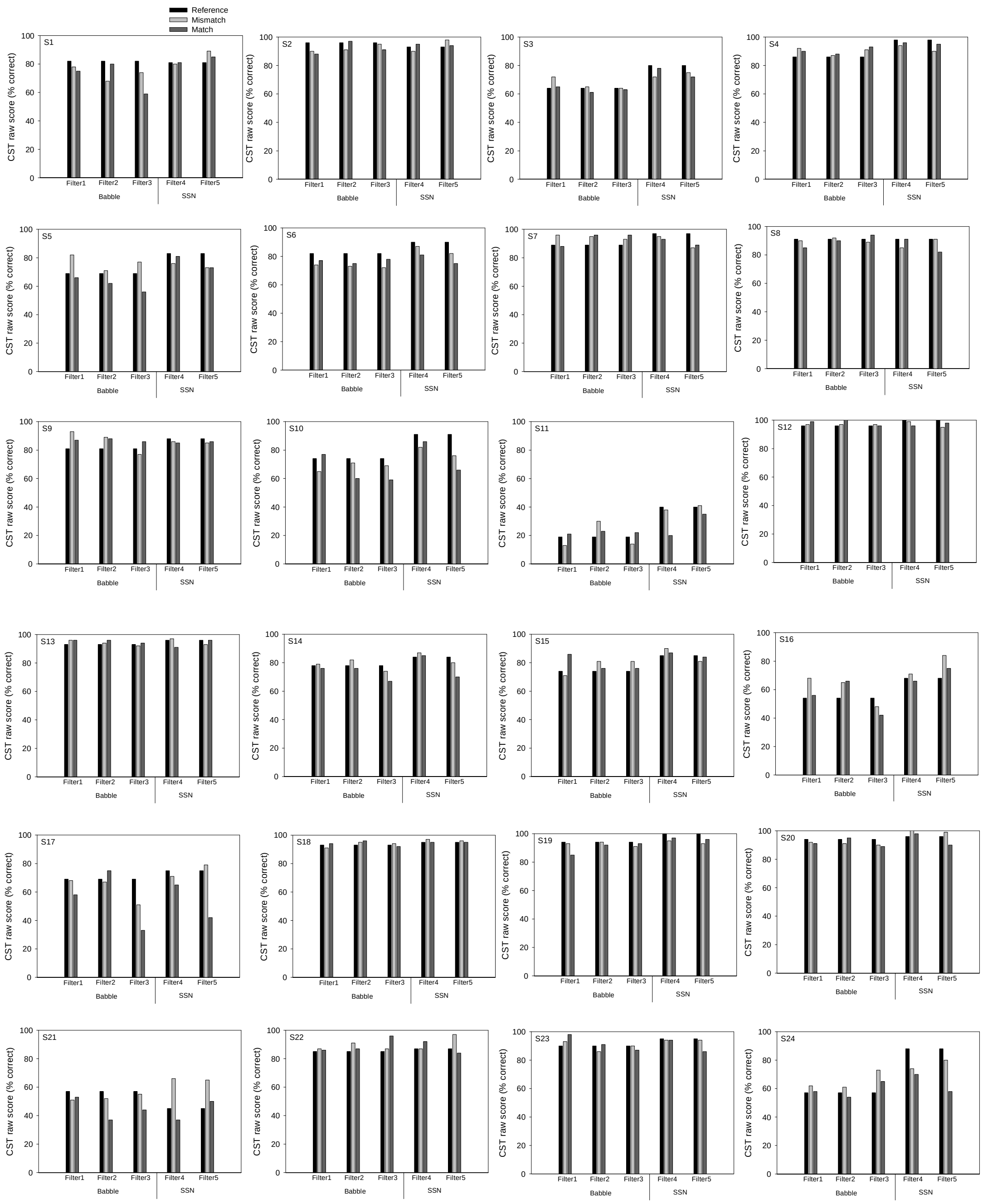

Figure 5. The CST raw scores (% correct) for each subject in the hearing-impaired group are plotted as a function of filter. The raw scores for two reference schemes (filter-off conditions) are displayed as well. “SSN” refers to the speech-shaped noise.

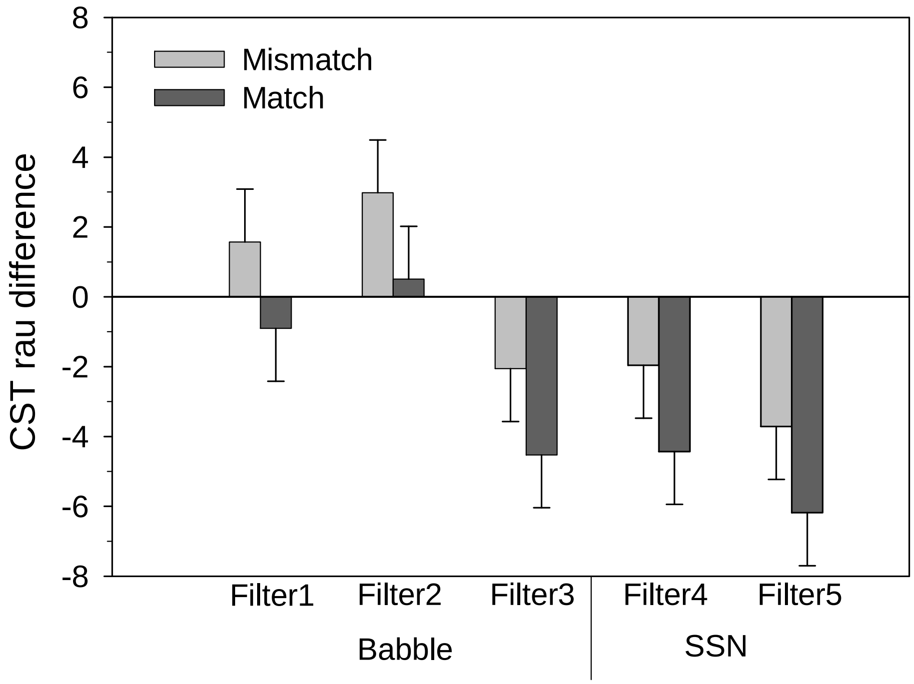

Figure 6. The mean CST rau differences between the mismatched and filter-off conditions, as well as the difference between the matched and filter-off conditions, are plotted as a function of filter. Values below zero indicate the CST performance was worse than that of the filter-off condition. "SSN" refers to the speech-shaped noise. The error bar displays the standard error.

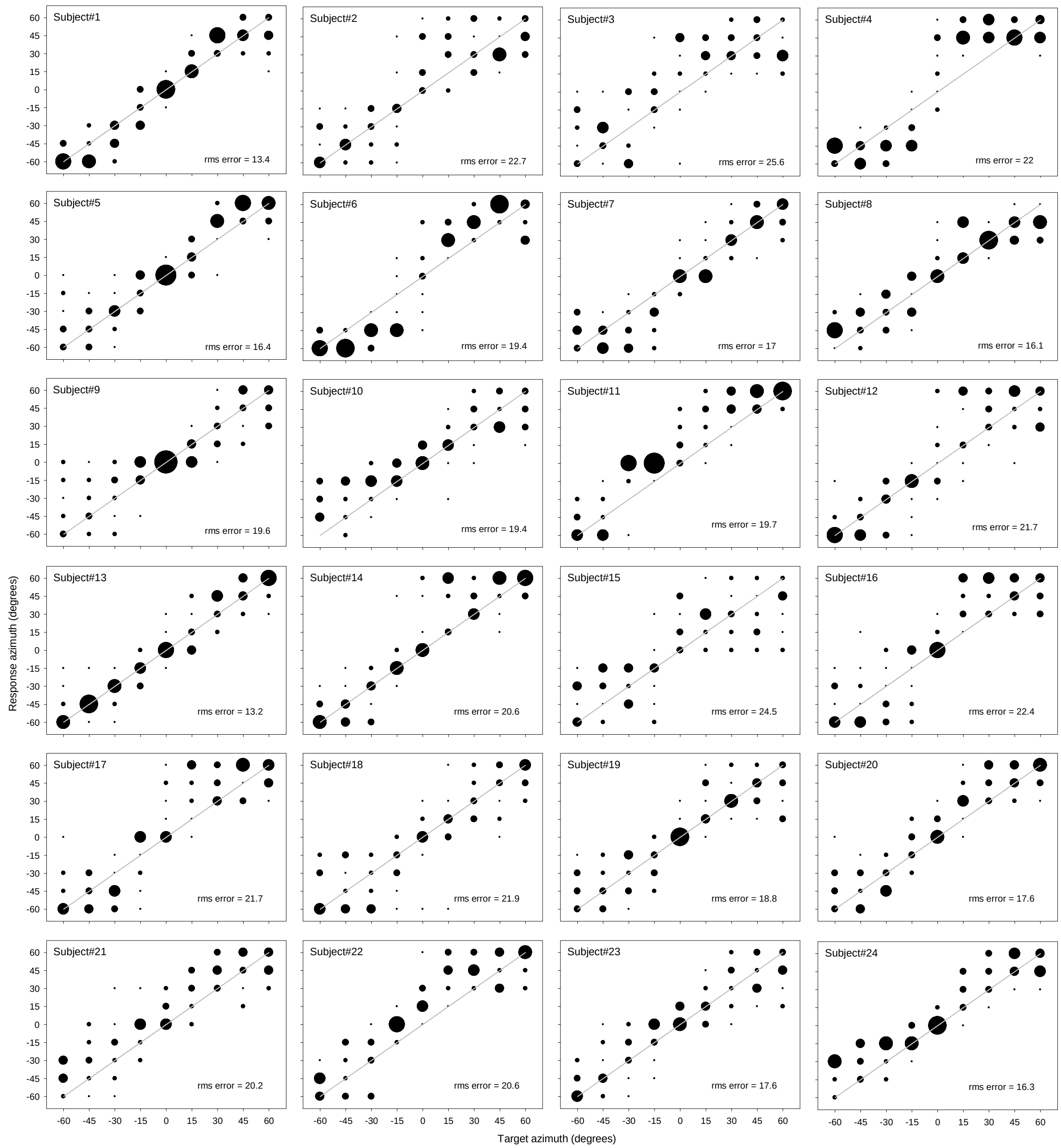

Figure 7. The individual baseline response patterns for the hearing-impaired group when tested in quiet aided. The response azimuth is plotted as a function of the target azimuth. The area of each filled circle is proportional to the number of responses made at that azimuth given by the subjects. The diagonal line shows an ideal localization performance. Negative and positive azimuths represent the left and right plane, respectively. The baseline rms error for each subject is displayed in each panel as well.

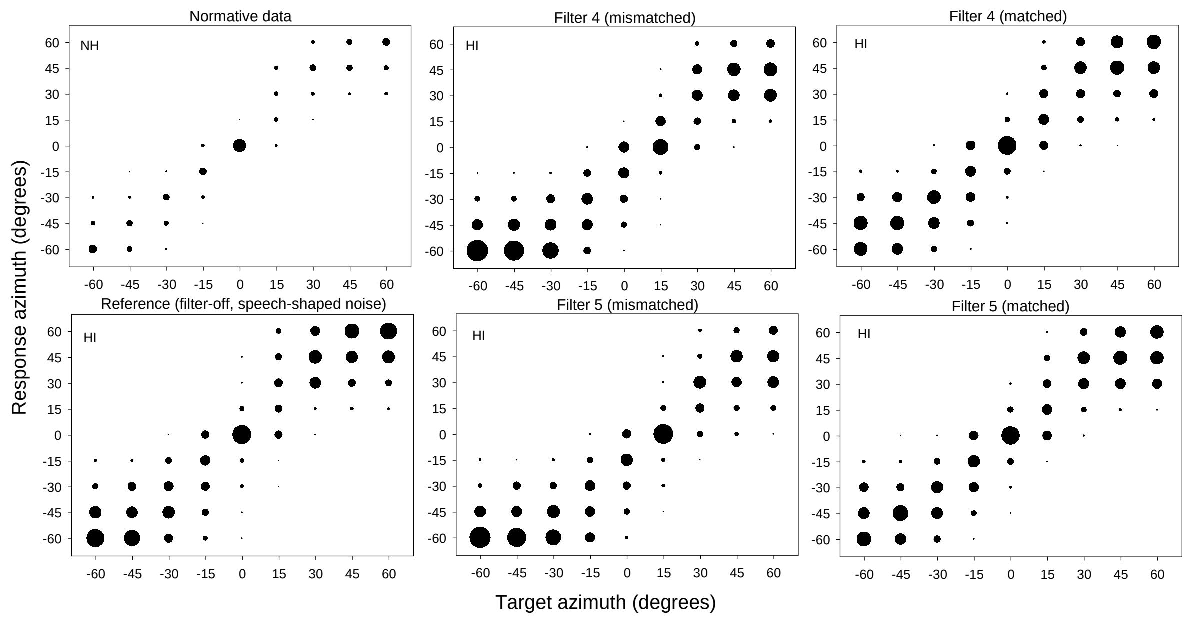

Figure 8. The response patterns for 10 normal-hearing (NH) subjects when tested in quiet (normative data), and 24 hearing-impaired (HI) listeners when tested in each of the five test conditions in the speech-shape noise. The response azimuth is plotted as a function of the target azimuth. The area of each filled circle is proportional to the number of responses made at that azimuth given by the subjects. Negative and positive azimuths represent the left and right plane, respectively.

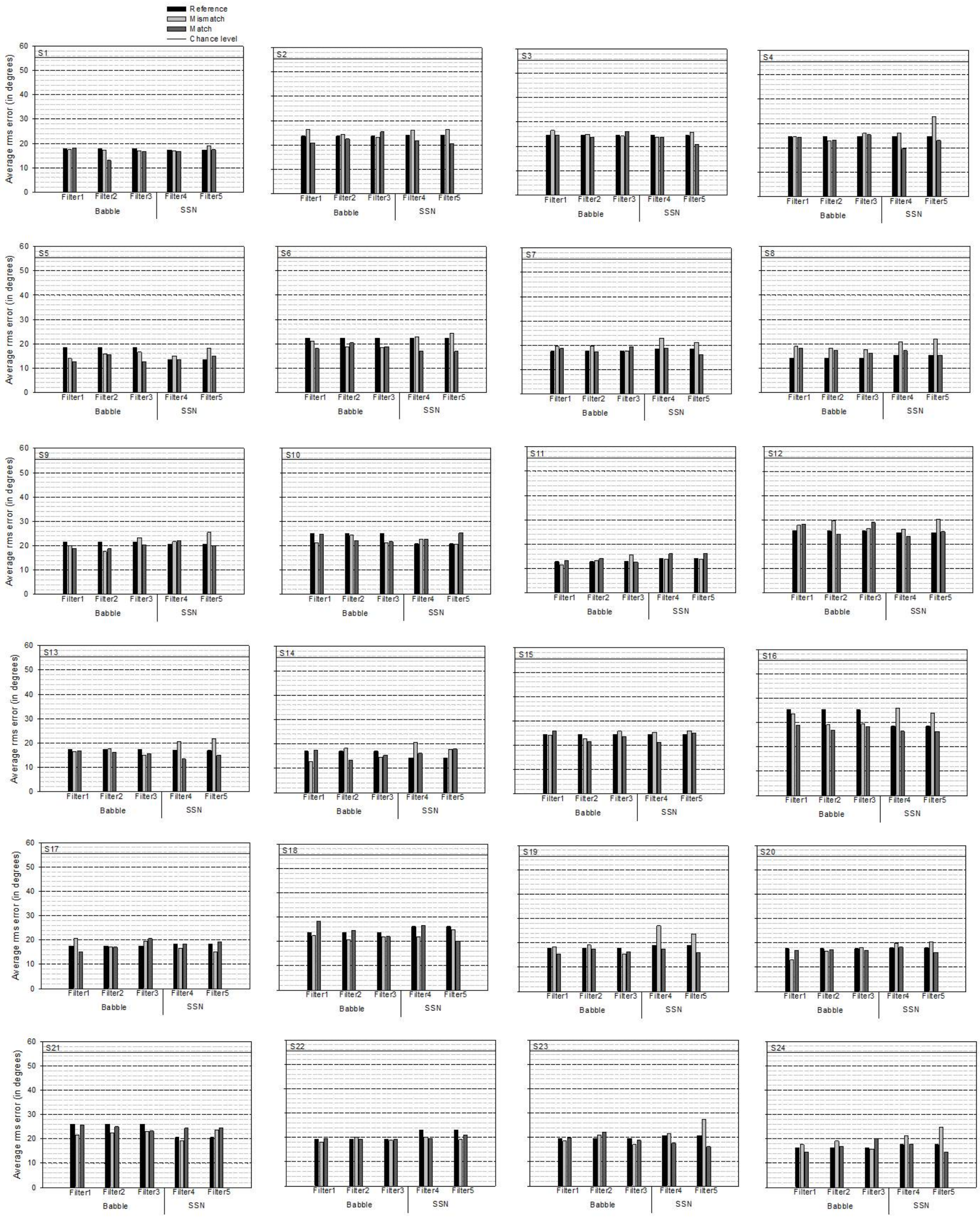

Figure 9. The rms error of each condition for each subject in the hearing-impaired group is plotted as a function of filter. "SSN" refers to the speech-shaped noise. The chance level is 55.43° for the localization experiment in this study.

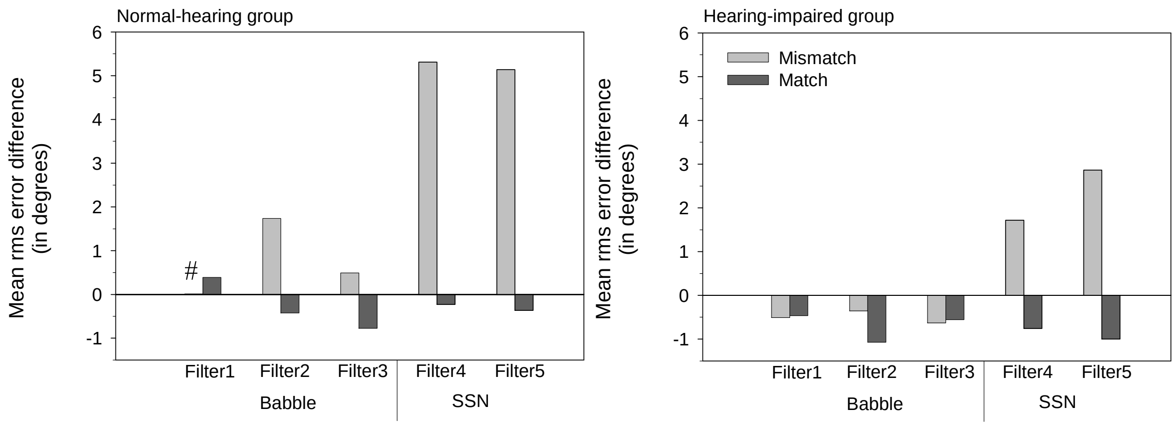

Figure 10. The mean rms error differences between the mismatched and filter-off conditions as well as the difference between the matched and filter-off conditions are plotted as a function of filter. The filter-off conditions refer to those using the reference schemes (no gain reduction in either ear). Values below zero indicate the rms error was less than that of the filter-off condition. "SSN" refers to the speech-shaped noise. The "#" indicates that the values were close to zero.

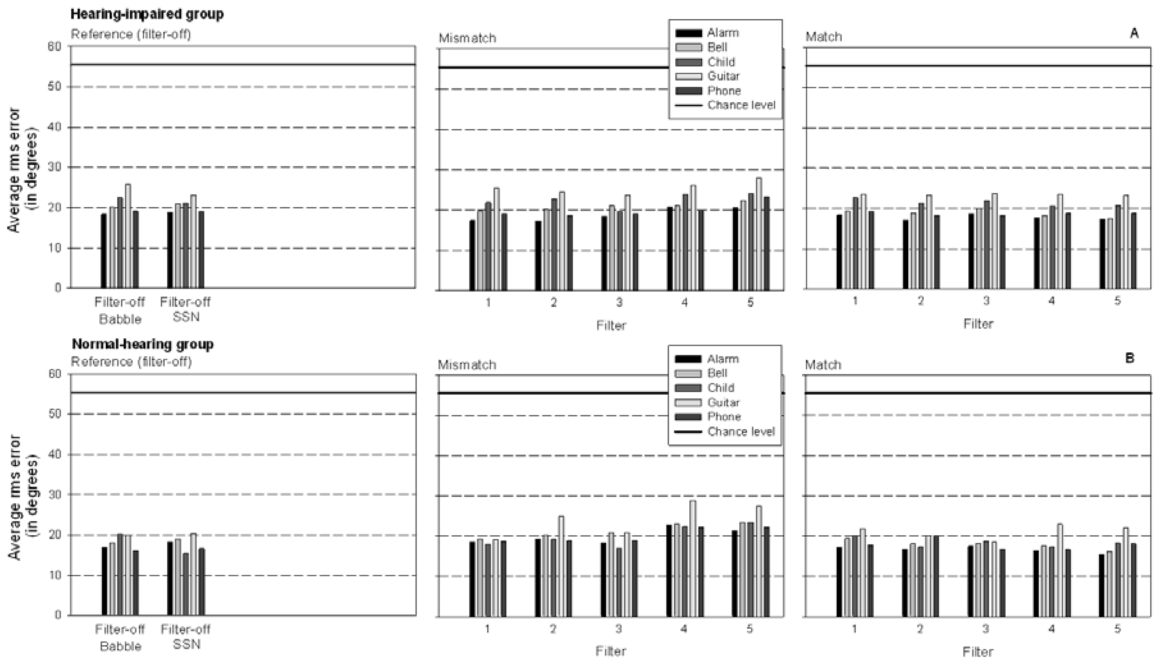

Figure 11. The average rms error for each stimulus is plotted as a function of filter in both mismatched and matched conditions as well as the average rms error for each stimulus in the filter-off conditions. Panel A displays the results for the hearing-impaired group; panel B displays the results for the normal-hearing group. "SSN" refers to the speech-shaped noise.

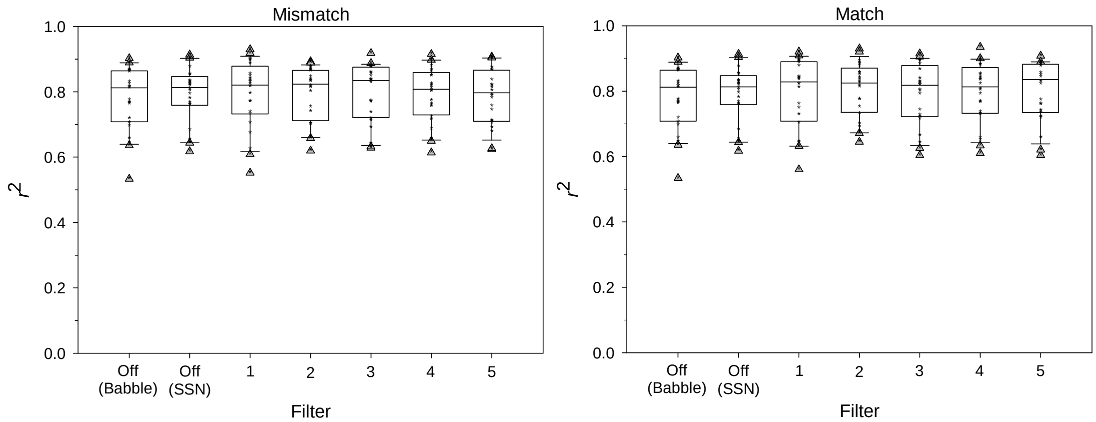

Figure 12. The values of r^2 are plotted as a function of filter for each listener with hearing loss and for two levels of Synchrony: mismatch and match. The values of r^2 for two filter-off conditions are included as well. The box represents the middle 50% of the data. The lower and upper outer lines that encase the box represent the 25th and 75th percentile of the data. The asterisk represents each individual subject in the hearing-impaired group. Solid horizontal lines indicate the median. The filled triangle represents outliers for the box plot.

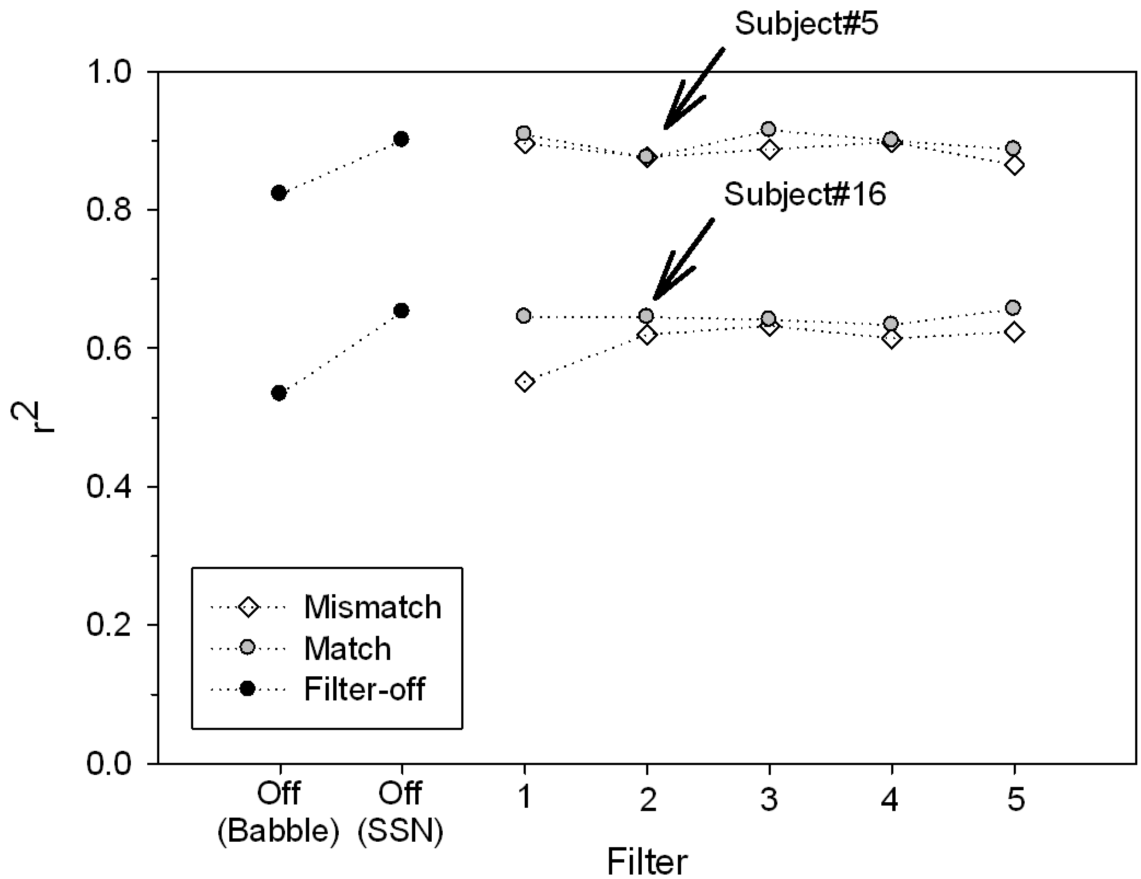

Figure 13. The values of r^2 are plotted as a function of filter for Subjects #5 and #16. Subject #5 shows better localization consistency than Subject #16. The localization consistency of the mismatched condition was slightly worse than that of matched condition for both subjects.

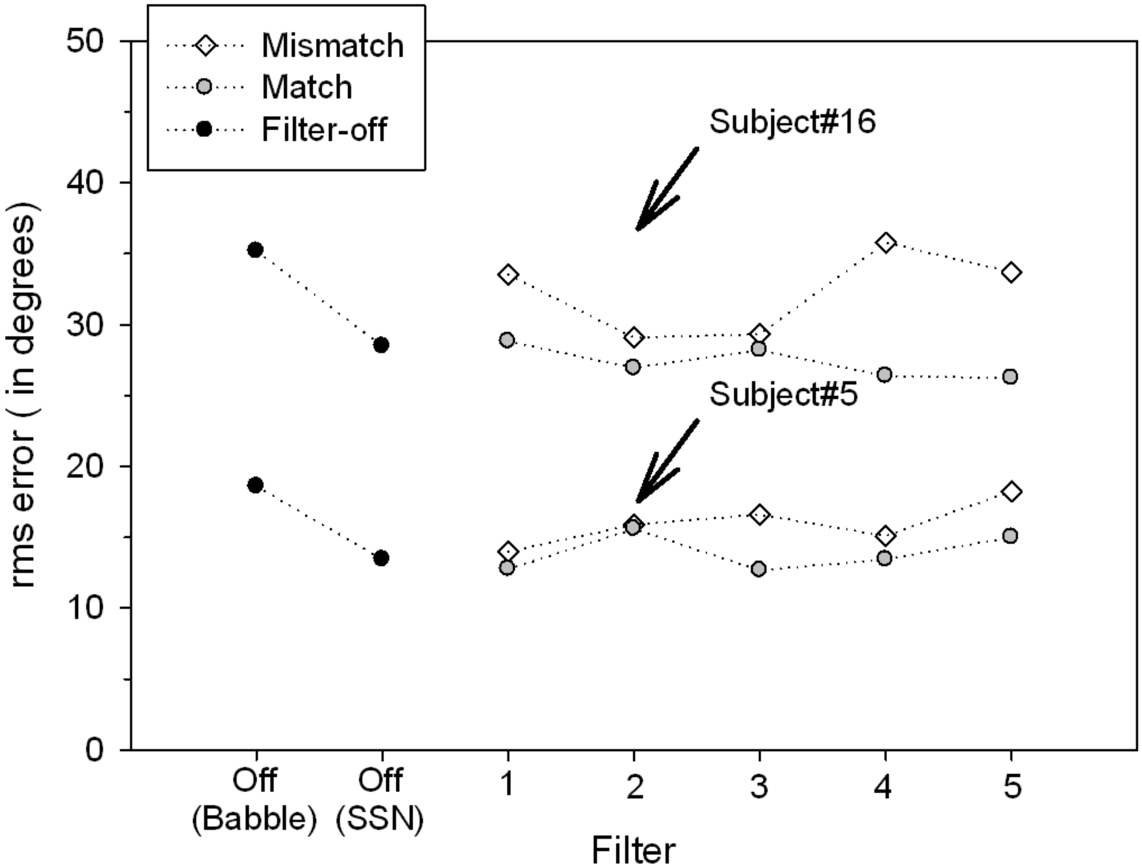

Figure 14. The values of rms errors are plotted as a function of filter for Subjects #5 and #16. Subject#5 shows better localization accuracy than Subject #16. The localization accuracy of the matched condition was better than that of the mismatched condition for both subjects.

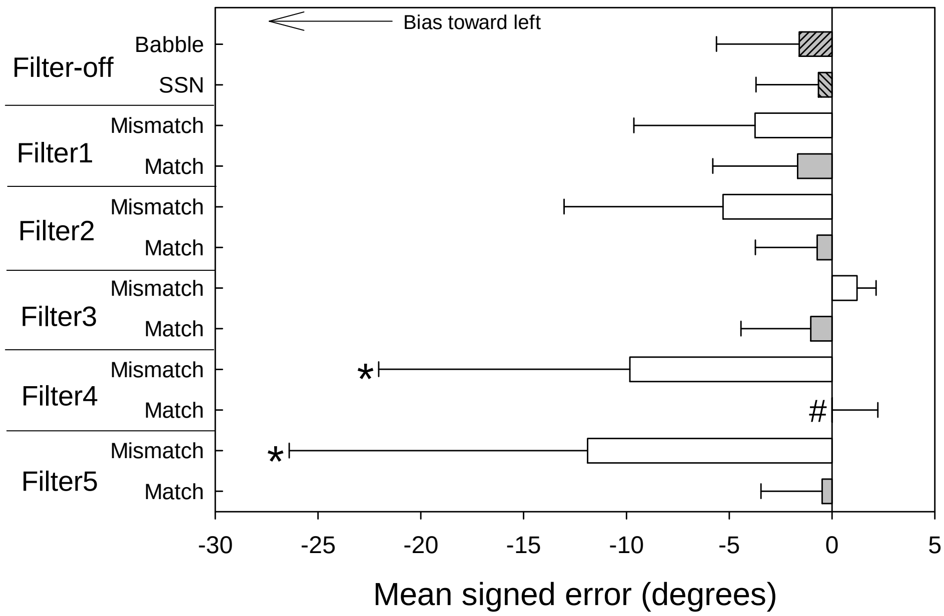

Figure 15. The mean signed error measured for the twelve conditions are shown for the hearing-impaired group. The error bar (one side) displays the 95% confidence interval. An asterisk shows that the mean signed error was significantly different from zero. The “#” indicates that the value of the mean signed error for this condition is close to zero. “SSN” refers to the speech-shaped noise.

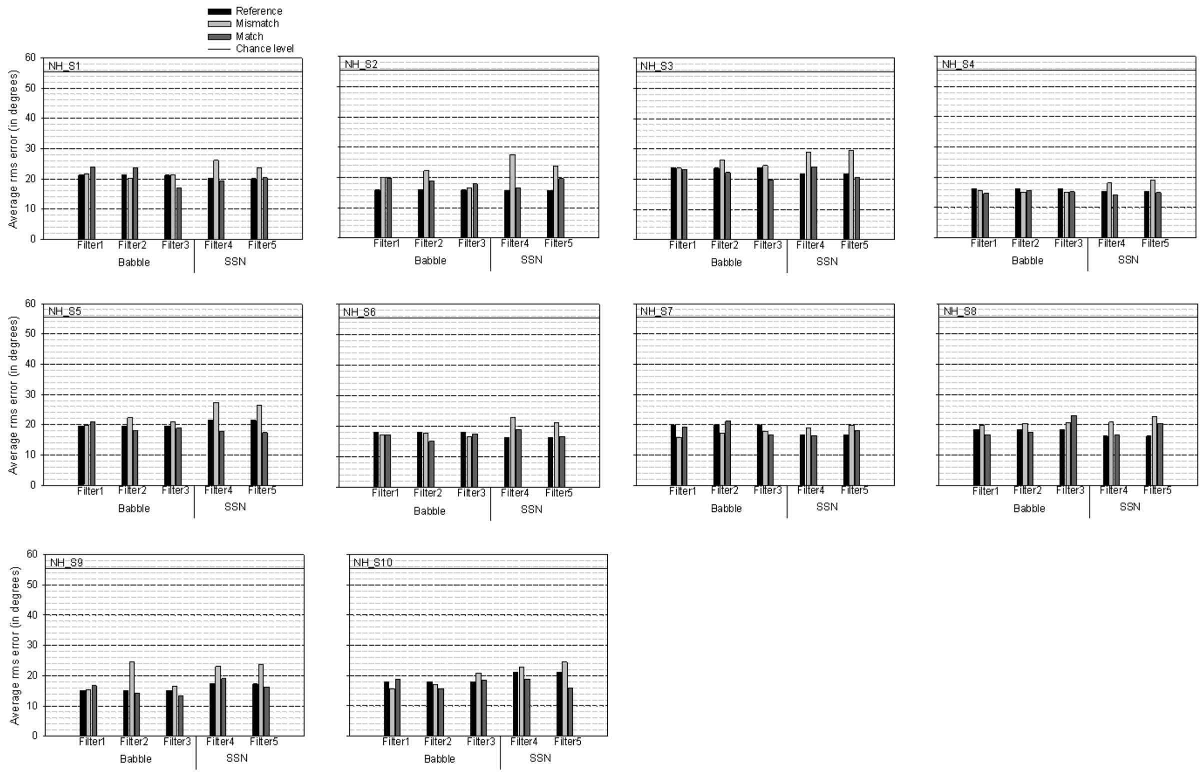

Figure 16. The rms error of each condition for each subject in the normal-hearing group is plotted as a function of filter. “SSN” refers to the speech-shaped noise. The chance level is 55.43° for the localization experiment in this study.

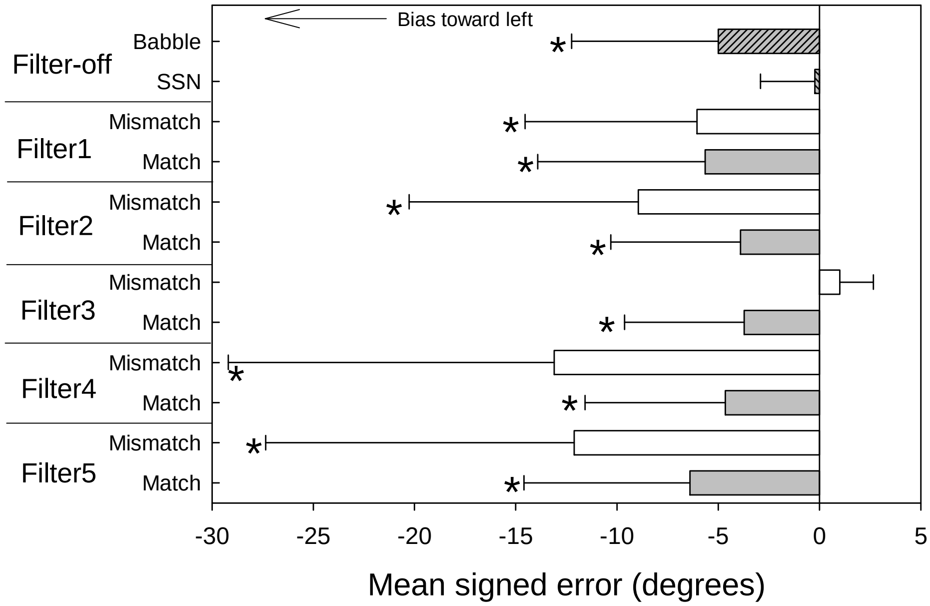

Figure 17. The mean signed error measured for twelve conditions for the normal-hearing group. The error bar (one side) displays the 95% confidence interval. An asterisk shows that the mean signed error was significantly different from zero.

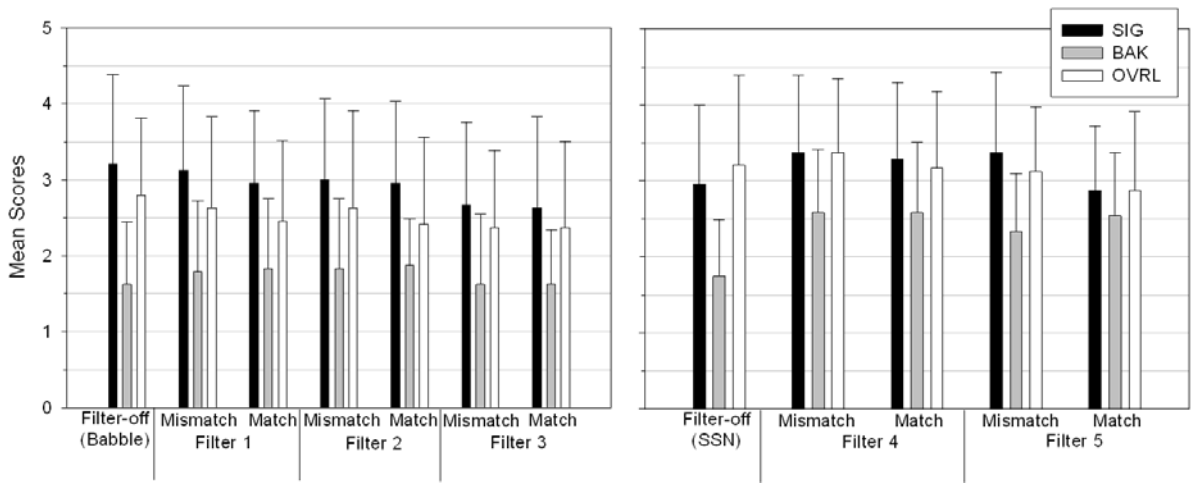

Figure 18. The mean ITU P.835 (sound quality) scores for the 12 conditions are shown for each subscale: SIG (signal distortion), BAK (background intrusiveness) and OVRL (overall mean opinion score). The error bar displays one standard deviation. The filter-off conditions refer to those using the reference schemes (no gain reduction in either ear).

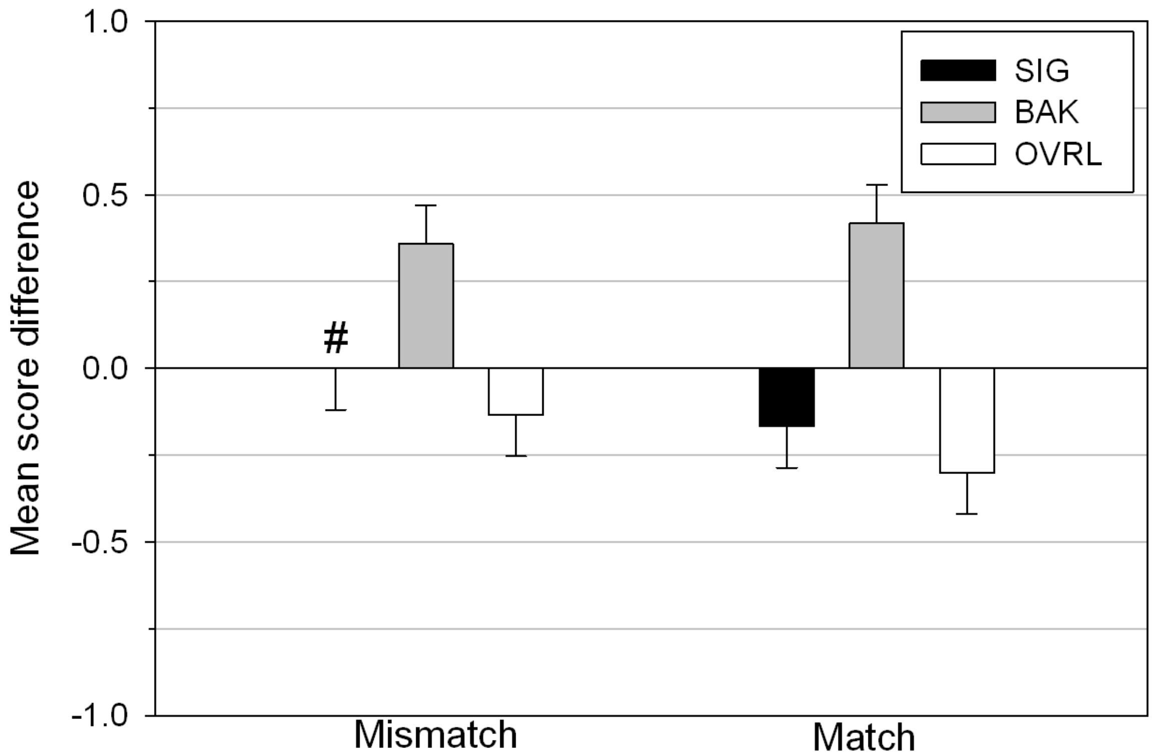

Figure 19. The mean ITU P.835 (sound quality) score differences for the filter-on and filter-off conditions as a function of Synchrony are shown for each subscale: SIG (signal distortion), BAK (background intrusiveness) and OVRL (overall mean opinion score). The error bar displays one standard error. The filter-on conditions refer to those using either bilaterally mismatched (gain reduction in one ear only) or matched (same gain reduction in both ears) gain reduction schemes. The filter-off conditions refer to those using the reference schemes (no gain reduction in either ear). Positive scores indicate the filter-on condition was rated to have better sound quality than a filter-off score. The “#” indicates the difference was zero.

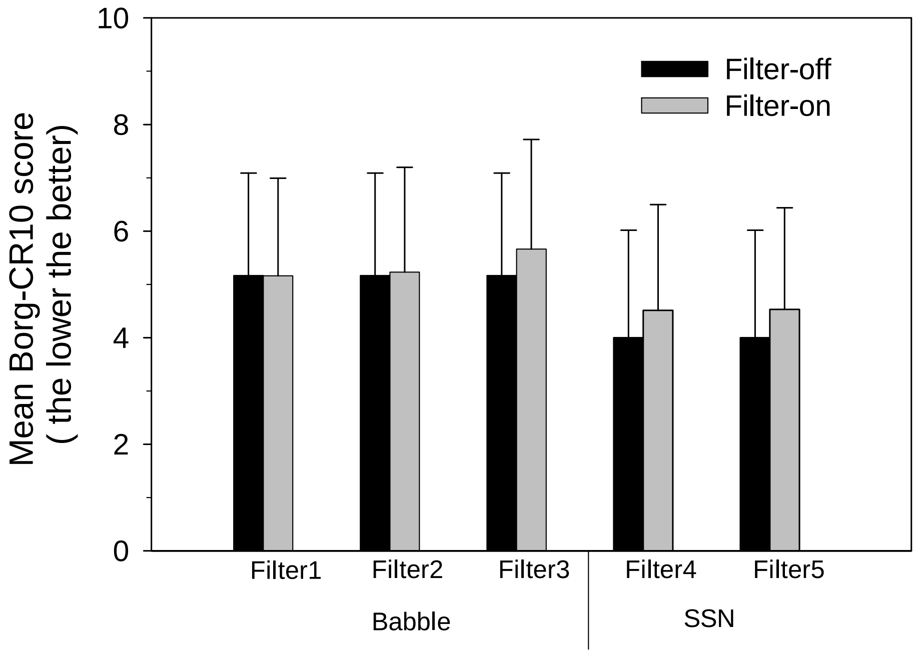

Figure 20. The mean Borg-CR10 scores (collapsed between the mismatched and matched conditions) of listening-effort ratings for all filter conditions are shown. The error bar displays one standard deviation. The scoring is from 0 (no effort at all) to 10 (extremely high effort); the lower the score, the less the perception of exertion is. "SSN" refers to the speech-shaped noise. The filter-on conditions refer to those using either bilaterally mismatched (gain reduction in one ear only) or matched (same gain reduction in both ears) gain reduction schemes. The filter-off conditions refer to those using the reference schemes (no gain reduction in either ear).

Table 8. The rms error difference comparisons across filters within mismatched conditions for both hearing-impaired and normal-hearing groups

	Filter2	Filter3	Filter4	Filter5
Filter1	-0.16 (-1.91)	0.12 (-0.53)	-2.23** (-5.88***)	-3.37*** (-5.69***)
Filter2		-0.28 (1.38)	-2.07* (-3.97**)	-3.22*** (-3.78**)
Filter3			-2.35** (-5.35***)	-3.49*** (-5.16***)
Filter4				-1.15 (0.19)

Note: The values in the parentheses were from the normal-hearing group. Because no significant differences across filters were found among the matched or filter-off conditions, the results are not displayed here.

*Adjusted $p < .05$; ** $p < .01$; ***$p < .001$(adjustment method: Bonferroni)

Table 9. The mean score and standard deviation for each item in the Speech hearing, Spatial hearing and Qualities of hearing domains (SSQ)

Item number		Mean	Standard deviation
	Speech-hearing items		
6	Talk with five people in noise without visual input	3.3	2.2
14	Follow one person speaking and telephone at same time	3.8	2.7
10	Talk with one person and follow TV	3.9	2.8
4	Talk with five people in noise with visual input	4.7	2.3
12	Follow conversation without missing start of new speaker	5.0	2.7
11	Follow one conversation when many people talking	5.3	2.3
7	Conversation in echoic environment	5.3	2.5
8	Ignore interfering voice of same pitch	5.7	2.4
9	Ignore interfering voice of different pitch	5.9	2.5
1	Talk with one person with TV on	6.1	2.3
5	Talk with one person in background noise	6.1	2.3
13	Have conversation on telephone	6.4	2.8
3	Talk with five people in quiet with visual input	7.3	1.8
2	Talk with one person in quiet room	8.9	0.8
	Spatial hearing items		
8	Judge distance from footsteps or voice	5.4	2.8
9	Judge distance of vehicle	5.8	2.7
15	Sounds closer than expected	5.8	2.4
5	Locate above or below on stairwell	6.0	3.3
11	Identify lateral movement from voice or footsteps	6.1	3.1
17	Sounds in expected location	6.3	2.4
10	Identify lateral movement of vehicle	6.3	2.9
1	Locate lawnmower	6.4	2.8
2	Locate speaker round a table	6.4	2.4
6	Locate dog barking	6.5	3.4
12	Identify approach or recede from voice or footsteps	6.6	2.8
7	Locate vehicle from footpath	6.6	2.6
16	Sounds further than expected	6.7	2.1
4	Locate a door slam in unfamiliar house	6.8	3.0
13	Identify approach or recede of vehicle	6.9	2.5
14	Internalization of sounds	7.8	2.5
3	Lateralize a talker to left or right	8.3	2.0
	Qualities of hearing items		
14	Need to concentrate when listening	4.3	3.0
18	Effort of conversation	4.7	2.8
15	Sounds unnaturally quiet when hear from one aid	5.5	2.9
16	Understand when driver of a car	5.9	2.2

Table 9. Continued

19	Ability to ignore competing sounds	5.9	2.5
17	Understand when car passenger	6.2	1.7
2	Sounds appearing jumbled	6.6	3.3
1	Separation of two sounds	6.7	2.9
11	Naturalness of everyday sounds	6.9	2.5
10	Naturalness of other voices	7.0	2.0
7	Identify instruments in music	7.3	2.4
8	Naturalness of music	7.4	1.8
3	Music and voice as separate objects	7.5	2.0
9	Clarity of everyday sounds	7.6	1.3
12	Naturalness of own voice	7.7	2.6
6	Distinguish different sounds	7.7	2.2
4	Identify different people by voice	7.8	1.9
13	Judging mood from voice	8.0	1.4
5	Distinguish familiar music	8.4	1.4

Note: The mean scores are ranked from the lowest to highest in each domain.

Table 10. The Pearson partial correlation between each item in the spatial hearing domain and better ear four frequency thresholds (4FA)

	Spatial hearing items	Pearson partial correlation
Item5	Locate above or below on stairwell	-0.69**
Item7	Locate vehicle from footpath	-0.68**
Item13	Identify approach or recede of vehicle	-0.64**
Item11	Identify lateral movement from voice or footsteps	-0.61**
Item14	Internalization of sounds	-0.61*
Item9	Judge distance of vehicle	-0.59*
Item12	Identify approach or recede from voice or footsteps	-0.57*
Item4	Locate a door slam in unfamiliar house	-0.56*
Item3	Lateralize a talker to left or right	-0.55*
Item10	Identify lateral movement of vehicle	-0.53*
Item15	Sounds closer than expected	-0.51
Item1	Locate lawnmower	-0.50
Item8	Judge distance from footsteps or voice	-0.47
Item2	Locate speaker round a table	-0.42
Item6	Locate dog barking	-0.38
Item17	Sounds in expected location	-0.36
Item16	Sounds further than expected	-0.36

Note: The absolute correlation values are ranked from the highest to the lowest.

*p <.01; **p <.001

Table 11. The mean ITU P.835 (sound quality) score difference comparisons across filters on the ITU signal distortion (SIG) and background intrusiveness (BAK) subscales for the hearing-impaired group

		Filter2		Filter3		Filter4		Filter5	
		SIG	BAK	SIG	BAK	SIG	BAK	SIG	BAK
Filter1	SIG	0.06		0.40		-0.54*		-0.33	
	BAK		-0.04		0.19		-0.65**		-0.50*
Filter2	SIG			0.33		-0.60*		-0.40	
	BAK				0.23		-0.60*		-0.46
Filter3	SIG					-0.94***		-0.73**	
	BAK						-0.83***		-0.69**
Filter4	SIG							0.21	
	BAK								0.15

Note: The score difference comparison on the overall mean opinion subscales (OVRL) is not shown because no significance was found.

*Adjusted $p < .05$; ** $p < .01$; ***$p < .001$(adjustment method: Tukey-Kramer)

CHAPTER 5

DISCUSSION

The results of this investigation confirmed the negative impact of bilaterally *mismatched* gain reduction on localization performance but not on speech perception performance. The results also indicate a negative impact of bilaterally *matched* gain reductions on speech perception performance, although not for localization. The matched results for speech perception were unexpected. The following section will provide a discussion of the possible reasons for these findings as well as for other outcome measures.

Speech Perception Performance

In this study, the impact of five different gain reduction patterns on speech perception in spatially-separated noise was assessed in two modes: matched and mismatched, across ears. The results were then compared to those of reference schemes (bilaterally linear amplification schemes without gain reduction). The results suggested: 1) the use of bilaterally mismatched gain reduction schemes did not result in deteriorated speech performance compared to the use of the reference schemes; 2) matching the gain reduction scheme between ears resulted in deteriorated speech performance compared to both the reference and the mismatched gain reduction schemes. Because binaural cues, especially ILD cues, could be disrupted using the mismatched gain reduction schemes, it was postulated that hearing-aid users might benefit less from spatial separation and experience poorer performance with the mismatched gain reductions than with the matched gain reductions. However, the results did not support our hypotheses.

The Impact of Binaural Cues

In this study, natural timing and level cues were purposely maintained for all sounds before they were processed through the mismatched and matched gain reduction schemes. Consequently, we determined that the mismatched gain reduction schemes disrupted ILD cues since the output level at one ear was reduced. Although we did not measure ITDs and ILDs, there is evidence to support the notion that the magnitude of ILDs with bilaterally independent DNR algorithms increases when the masker is presented from a single side (Keidser et al., 2006). In the present study, the time delays were controlled, although ITD cues could still be influenced due to the mismatched gain reduction in the low-frequency area. We expected that ILD cues would be restored to normal when the gain reduction schemes were matched between ears. Thus, we expected speech perception to be compromised using the mismatched gain reduction schemes, yet be restored when using the matched gain reduction schemes. However, the results were not consistent with our predictions.

ITDs are the primary cues used in binaural analysis (binaural summation and binaural squelch) for low-frequencies. Because ITDs were not likely to be impacted regardless of whether gain reduction schemes were mismatched or matched between ears in this study, the subjects were still able to make use of ITD cues compared to the conditions using the reference schemes.

In contrast, ILDs are the main basis for the better-ear effect, which arises from the acoustical shadowing of the head. When the speech is presented from the front and the background noise is at one side, the speech level should be the same between ears (i.e., the ILD is zero for speech) while the noise level should be higher in the near ear compared to the far ear due to the head shadow. This results in an improvement of the SNR at the far ear. It has been postulated that the brain can determine which ear has the better SNRs and attend to that ear to receive the advantages due to the better-ear effect (Edmonds & Culling, 2006).

It has been found that ITDs and ILDs contribute differently to spatial benefit (e.g., Bronkhorst & Plomp, 1988; Edmonds & Culling, 2005). These investigators used binaural intelligibility level difference (BILD) to quantify this benefit. In general, the results from these studies suggest that the BILD from the combined ITDs and ILDs is smaller than the summed BILD from each individual cue. When the target and masker are separated by up to 60°, the relative contributions of ITDs and ILDs are roughly equivalent. When the separation between the target and masker is more than 60°, such as when the target is from the front and the masker is from a 90° azimuth, the contribution of ILDs is greater than that of ITDs. In other words, the contribution of ILDs to the BILD is dominant when the interfering noise is from the side and the speech is from the front. For the purpose of the study, we can infer that the ILDs are the primary cues giving rise to the binaural advantages.

As per our design, for the mismatched gain reduction schemes studied herein, the overall level of the near ear was reduced regardless of which gain reduction pattern was used. The ILDs consequently increased for both speech and noise whereas the SNRs in each ear did not change compared to the reference scheme (Note: The perceived locations of the speech and noise may be shifted toward the left to some degree). As a consequence, the ear with better SNR was not influenced by the mismatched gain reduction condition. This could explain why there was no significant difference in speech perception performance between the mismatched and filter-off conditions. Still, some individual subjects showed better speech perception performance in the mismatched conditions compared to the filter-off conditions. It is likely that reducing the gain at the near ear (right side in this study) with worse SNR actually enhanced the better-ear effect and allowed the listeners to attend more to the ear having better SNR. It is also possible that the speech perception performance in the matched conditions was worse than that in the mismatched conditions due to the elimination of this better-ear effect when the gain reduction was matched between ears. Similarly, Bronkhorst and Plomp (1988, 1989) found that a reduction of 20dB for the ear with the poorer SNR did not cause a decrement in speech performance with spatially-separated noise, whereas a reduction for the ear with better SNR resulted in significantly poorer performance. Furthermore, Bronkhorst and Plomp found that the spatial release due to ITD cues was not impacted by the attenuation of 20dB in either ear. Another possible explanation for the deteriorated speech performance in the matched conditions is due to audibility reduction; the bilaterally matched gain reduction schemes compromised the audibility of the better ear because gain reduction was up to approximately 10 dB in the low-frequencies and 7dB in the high-frequencies across five gain reduction patterns. Therefore, listeners with hearing loss could not benefit from the better-ear effect due to the reduced audibility resulting from the applied gain reductions.

It should be noted that, in a cocktail party environment, listeners have to use all available acoustic cues to selectively attend to the target talker. Listeners can take advantage of multiple cues including level differences between talkers, the fundamental frequency of the talker's voice, and spatial cues. Higher-level factors, such as an *a priori* knowledge of the target location, can also contribute to better performance (Kidd et al., 2005). In addition, Freyman, Balakrishnan, and Helfer (2001) found that if listeners can perceive the location of the target and masker using the precedence effect (in which the first direct sound and the following-arrived indirect sounds are perceived together as from the first sound source), the absence of interaural cues would not interfere with the benefit from spatial separation. Because the subjects were aware of the location of the target and masker for the speech perception task in this current study, this factor was minimized.

The Impact of Gain Reduction Patterns

Recall that the gain reduction patterns were determined using two distinct noises: multi-talker babble and speech-shaped noise. In the preliminary stages of the study, one gain reduction pattern did not result in any changes between the DNR-on and DNR-off conditions in the multi-talker babble when the SNR was -5 dB; consequently, this gain reduction pattern was not used as a filter in the present study. The remaining gain reduction patterns used in the study had varied magnitudes of gain reduction across frequencies. A greater amount of gain reduction was observed in the speech-shaped noise than in the multi-talker babble. One specific gain reduction pattern (Filter 3) had a gain boost at the mid-to-high-frequencies as well as a low-frequency gain reduction.

As expected, the speech perception performance for the filter-off condition using multi-talker babble was worse than using speech-shaped noise. When a masker is speech-shaped noise, the masking that occurs is mainly a result of the energy overlap between the speech signal and the masker, and is often referred to as “energetic” masking. When a masker is multi-talker babble, the masking is not only from energetic masking but also from “informational” masking. Informational masking occurs when the speech signal and the masker (competing speech) are both audible but the listener is unable to isolate the speech signal from the masker (Kidd, Mason, & Gallun, 2005). It is believed that informational masking involves higher level processes along the auditory system. Therefore, the existence of both energetic and informational masking may have made the speech perception task more difficult using multi-talker babble than speech-shaped noise. However, it is interesting that the CST performance *change* from the filter-off to filter-on conditions using multi-talker babble was generally greater than using speech-shaped noise regardless of whether the gain reduction was matched or mismatched (i.e., Synchrony). This does not mean that the performance of the filter-on conditions in the babble was better than that in the speech-shaped noise. Rather, it suggests only that listening using any gain reduction patterns might further degrade performance for speech-shaped noise than multi-talker babble. Again, it should be noted that *more* gain reduction was used with the speech-shaped noise than the multi-talker babble, as per the current algorithms. Also, speech perception scores when using Filters 3, 4, and 5 in the matched filter-on conditions were poorer than in the filter-off conditions. Because these three gain reduction patterns (Filters 3, 4, and 5) all have significant high-frequency gain alterations compared to Filters 1 and 2, the matched conditions may have interfered with the better-ear effect as discussed previously.

Beyond laboratory setting, real-world environments usually contain multiple sound sources, incorporating both talkers and non-speech steady-state noise alike. Real-world stimuli that trigger DNR algorithms could be a combination from both steady-state noise and multi-talker babble. Therefore, it can be assumed that the actual gain reduction effect in a real-world environment could be some variation of the gain reduction patterns investigated here. In addition, the DNR patterns used in this study were determined in a free field with both the signal and noise coming from the front in the horizontal plane, while the actual speech and noise stimuli used in the speech perception task were located at 0° and 90°, respectively. The gain reduction patterns may not have reflected how the algorithm might actually have altered gain. Therefore, the impact of gain reduction might be underestimated in the current study.

Individual Data

It is noteworthy that the CST testing material has many contextual cues, and seemed not to significantly challenge most subjects in the present study, as noted in the high scores obtained. In addition, the asymmetrical configuration of the interfering noise made it even easier for subjects to benefit from spatial separation. The spatial release from masking could be up to 7dB or more for listeners with hearing loss (e.g., Bronkhorst & Plomp, 1989). It is not surprising that the majority of the subjects obtained high scores on this test. Figure 5 depicts the raw scores (% correct) for each subject in each condition. Three of the 24 subjects had raw scores of 100% in some conditions. Out of these three subjects, Subject #12 performed extremely well in all conditions (≥ 95%). In order to investigate individual data and exclude the ceiling effect, we scrutinized the data in the following ways:

First, data from subjects who had raw scores (% correct) greater than 95% were removed from the data set. Fourteen subjects remained for the analysis. The mean CST score differences (rau) between the mismatched and reference schemes, and the difference between the matched and reference schemes were calculated for the remaining 14 subjects. The same procedures were carried out for subjects who had raw scores greater than 90% and 80%, respectively. Figure 21 shows CST score differences (rau) calculated for the mismatched and reference schemes and for the matched and reference schemes. Values below zero indicate the CST performance was worse than that of the filter-off conditions. Panels A, B, C, and D display the group data for all 24 hearing-impaired subjects who participated in the study, 14 hearing-impaired subjects whose raw scores (% correct) were less than 95%, 12 hearing-impaired subjects whose raw scores (% correct) were less than 90%, and 4 hearing-impaired subjects whose raw scores (% correct) were less than 80%, respectively. The figure reveals an obvious trend in which the speech perception performance of the matched conditions was worse than that of the mismatched conditions for these four subgroups. Although ceiling effects existed for some subjects in the speech recognition test, the trend provides evidence to support that matching gain reduction between ears could result in deteriorated speech performance.

Second, the same data set (CST score differences from the filter-off to the mismatched filter-on conditions as well as the score differences from the filter-off to the matched filter-on conditions) was used to plot individual performance for all 24 subjects (Figure 22).The largest difference between the filter-off and the mismatched conditions was 25 rau; the largest difference between the filter-off and the matched conditions was 34 rau; the largest difference between the mismatched and the matched conditions was 36 rau. Panels A to C show the results for the multi-talker babble, and Panels D and E show the results for the speech-shaped noise. The bar above zero indicates that the score of the mismatched/matched gain reduction schemes was better than that of the reference schemes. An obvious trend is seen in which the majority of the 24 subjects performed worse in the matched conditions than in the mismatched conditions. In contrast, no clear trends emerged when comparing speech perception performance between the filter-off and mismatched conditions. Apparently, some subjects performed better in the mismatched conditions than in the filter-off conditions while others did not. Figure 23 shows the derived CST rau differences between the matched and the mismatched conditions in a box plot for each filter for all 24 subjects. The box represents the middle 50% of the data while the lower and upper outer lines that encase the box represent the 25th and 75th percentile of the data. The value below zero indicates that the speech perception performance for the matched conditions was worse than for the mismatched conditions. Inside the box, the dashed horizontal lines indicate the mean, and the solid horizontal lines indicate the median. The asterisk represents each individual subject. The filled triangle represents outliers for the box plot. It is interesting that the box is shifted to the high end of the outer line for Filters 3, 4, and 5. This suggests that the data are negatively skewed, which further indicates that the majority of the subjects had worse speech performance in the matched conditions compared to the mismatched conditions. However, it is noteworthy that most score differences are within a +15 ~ -15 rau range. Cox et al. (1988) suggested a 95% critical difference (15.5 rau) for the CST test when comparing performance among hearing-impaired listeners. Because Cox et al. (1988) presented stimuli monaurally while we presented stimuli bilaterally, we would expect speech performance in the current study to be better than that in the Cox et al. study. Therefore, it is likely that the critical difference in this study is actually greater than 15.5 rau. Because most score differences between the mismatched and matched conditions were less than 15 rau, the negative impact of bilaterally matched gain reduction seemed trivial for most subjects in the present study. However, we should not ignore those subjects who did poorly in the matched conditions and are represented by the outliers at the low end of the box plot.

Localization Performance

Results from the localization experiment support both hypotheses in this study. That is, relative to the reference scheme (no gain reduction in either ear), the use of mismatched gain reduction schemes (gain reduction in the right ear only) led to poorer localization accuracy and lower localization consistency. Relative to the mismatched gain reduction schemes, the use of matched gain reduction schemes (same gain reduction in both ears) resulted in an improved localization performance. Only a relative weak correlation was found between self-reported and measured localization performance.

Response Patterns and Mean Signed Errors

For listeners with hearing impairment, both response patterns and mean signed errors of the mismatched schemes indicated a bias toward the left ear, with matched conditions “pulled back” to the right ear (i.e., less bias toward the left). Listeners with normal hearing showed results similar to those with hearing impairment. Keidser et al. (2006) suggested that the bias is usually directed to the ear fitted with more gain which seems both logical and probable. In the present study, more gain was given to the left ear for the mismatched conditions compared to the matched conditions. Therefore, more bias was toward the left in the mismatched conditions than in the matched conditions. Surprisingly, the bias of the mismatched condition for Filter 3 showed a different pattern of bias than the other mismatched conditions. From Figure 15, we can see that the bias of the *mismatched* condition with Filter 3 is toward the right side whereas the bias of the *matched* condition is toward the left side. It is unclear why the result of Filter 3 differs from the others. This might be due to the unique high-frequency gain boost in Filter 3 that is not present in the other filters, which made the bias toward the right side in this mismatched condition.

The Impact of Binaural Cues

As discussed previously, ILD cues were altered using the mismatched gain reduction schemes, especially for Filters 4 and 5, whereas ITD cues were less likely to be altered in this study. It is not surprising that localization accuracy deteriorated when subjects were given the bilaterally mismatched gain reduction schemes. Although past studies generally support the notion that low-frequency ITDs are the dominant cues to establish the source locations in the horizontal plane (e.g., Wightman & Kistler, 1992), some investigators have reported that high-frequency ILDs are actually more useful in terms of left-right localization in the horizontal plane (e.g., Carlile et al., 1999). Therefore, it seems likely that disrupted ILD cues due to independently bilateral signal processing could negatively impact localization performance. Moreover, the improved localization accuracy using the matched gain reduction schemes supports the idea that distorted ILD cues can be restored to or close to the naturally occurring ILD cues.

Because the amount of gain reduction in each filter condition differed from low to high frequencies, the filters' effect on the ILD cues varied. The use of Filters 1, 2, and 3 did not result in significant differences in localization performance among filter-off, mismatched, and matched conditions. One possible reason is that the amount of gain reduction in the high frequency area for these three filters was trivial (+/- 1~2 dB). On the other hand, the use of Filters 4 and 5 resulted in worse localization performance under the mismatched conditions compared to either the filter-off (2~3° average rms error difference) or matched conditions (3~4° average rms error difference). It is likely that reducing gain by an average of 6dB in the high frequency region for Filters 4 and 5 is substantial and sufficient to alter localization performance. Besides the distorted ILD cues, decreased localization accuracy might also be due to the fact that some of the testing stimuli were below detection thresholds when listening in noise. In other words, audibility of the signal may have been compromised at the ear ipsilateral to the noise (Lorenzi et al., 1999b). As a consequence, listeners might not be able to access naturally occurring binaural cues because they cannot detect the signal in an adverse noisy environment.

One could argue that reduced audibility also occurred in the matched conditions, which means that subjects might not be able to take optimal advantage of binaural cues in these situations either. However, because the results from the current study have shown that average localization performance using the matched gain reduction schemes was better than that using the mismatched gain reduction schemes, this argument is not likely to be founded. Since most hearing-impaired subjects in this study had mild hearing loss in the low frequencies, it is possible that these subjects could still make use of available ITD cues. Therefore, it is not likely that any reduced high-frequency audibility interfered with the localization performance for the matched conditions.

Related to the audibility issue, Good and Gilkey (1996) proposed that detectability of signals in noise may vary according to location. Subjects might narrow the range of the possible sources during the experiment for that matter. For example, when the masker emanates from the front, the signal presented from other locations is more likely to be detected than when presented from the front or behind. Listeners may use this varied "detectability" as a "non-spatial" cue to localize signals. Because the location of the masker was fixed at a 90° azimuth, our subjects likely had more difficulty detecting sounds coming from the right side than those from the left side, especially softer sounds. As a result, some subjects might mistakenly localize any soft sounds to be from the right side.

It is interesting to note that the impact of the mismatched schemes was more severe on listeners with normal hearing than on listeners with hearing impairment. That is, for the normal-hearing group, the average localization rms error was raised by 5° from the filter-off conditions to the mismatched conditions, whereas for the hearing-impaired group, it was raised by 3°. We also found differences among the groups for the main effect of the stimulus; that is, it was not significant for the hearing-impaired group but it was significant for the normal-hearing group. Everyday sounds were purposefully chosen as stimuli in the localization experiment because of their ecological validity and because they were representative of a wide range of acoustic parameters. Despite this, it is unclear why the main effect of Stimulus was significant only for those subjects with normal hearing. This could be due to an interaction between everyday sounds and the background noise type for the normal-hearing group, but not for the hearing-impaired group. Figure 24 displays the average rms error for each everyday sound in two filter-off conditions for both groups. For the normal-hearing group, when Stimulus “Child” was presented with babble, it was found to result in the poorest localization accuracy whereas when it was presented in speech-shaped noise, it resulted in the best localization accuracy (the rms error difference was 5°). It is likely that the fluctuating multi-talker babble which has human voices interfered with the perception of “child laughing” for the normal-hearing subjects, which resulted in the greatest rms error among five everyday sounds. Interestingly, subjects in both groups had reported that the stimulus “Guitar” was the most difficult sound to localize in noise.

A number of studies have suggested that the human brain can acclimate or adapt to altered or distorted localization cues quickly (e.g., Bauer, Matuzsa, Blackmer, & Blucksberg, 1966; Drennan et al., 2005; Noble & Byrne, 1990, 1991; Shinn-Cunningham, Durlach, & Held, 1998a, 1998b). For example, Bauer et al. (1966) found that after a period of time people with normal hearing can localize sounds with adaption when one ear is plugged. Noble and Byrne (1990, 1991) noted that hearing aid users wearing their own BTE hearing aids displayed better localization performance than when wearing newly-fit ITE hearing aids. More recently, Drennan et al. (2005) reported that localization performance was similar between phase-preserving amplification (in which the naturally occurring ITD cues are preserved) and non-phase-preserving amplification after a period of use for hearing aid users. These data support the notion that, although binaural cues are distorted using the mismatched gain reduction schemes, any negative impact on localization performance might be reduced after an extensive training or long-use experiences.

Self-reported Localization Performance

Self-reported localization performance was assessed using the SSQ inventory. Although the response patterns in this study were similar to that reported in Gatehouse and

Noble (2004), mean item scores were higher (i.e., better self-reported localization ability). This is likely because subjects from the Gatehouse and Noble study were all unaided whereas most subjects in the present study were regular hearing-aid users.

The present study found the relationship between self-reported spatial hearing and measured localization ability to be weak. It is possible that the virtual environment and/or the localization task employed herein were not representative of real-world efforts. It should also be noted that hearing aid simulation was used in the present study, rather than actual hearing aid devices. If we had used actual devices, we might have been able to more accurately assess real world outcomes albeit in a laboratory environment. However the use of actual hearing aids would have required control/consideration for a number of other parameters, these including time constants, gain differences, channel interaction, and so on. Using a simulation allowed for direct control over the variable in question: gain reduction patterns. Future research can address the effectiveness of these algorithms in the real world with actual hearing aid devices.

Discrepancy between the Results of Speech Perception and Localization Performance

It is well-known that binaural cues (ITDs and ILDs) are important for both localization and spatial release from masking due to the separation of speech target and masker (e.g., Wightman & Kistler, 1992; Zurek, 1993). Given the above discussion on ILD cues and distortion observed with the bilaterally mismatched gain reduction schemes, one might expect that both localization and speech perception performance in spatially-separated noise would have deteriorated. However, based on the results of this study, this assumption did not hold true. It is unclear why the mismatched gain reduction schemes impacted localization but not speech perception. It is possible that listeners with hearing impairment do not use the binaural cues in the localization task in the same manner as in the speech recognition task. Although Edmonds and Culling (2006) proposed that the same binaural cues are used in sound localization and speech perception in spatially-separated noise, this does not necessarily infer a "common mechanism" behind these two processes. Instead, the auditory system may use ITD and ILD cues differently to locate sound sources and receive benefit from spatial separation. It is possible that localization accuracy in the horizontal plane depends upon the intactness of binaural cues and how well listeners can make use of them, whereas spatial "unmasking" does not require intact binaural cues to extract useful information.

For communication in real world environments, the target talker and the interference are not always predictable in space and time. Listeners often have to locate and alter their attention among multiple sound sources. In contrast, the scenario generated in this study was extremely asymmetrical and predictable. That is, the target was always from the front and the masker was always from the right side. Listeners did not need to switch attention to accomplish the speech perception task. It may underestimate the effect of the mismatched and matched gain reduction schemes (i.e., Synchrony).

Sound Quality Rating

Although no significant differences for the overall sound quality rating were found among the different gain reduction patterns, several patterns did show a trend towards better sound quality, i.e., they resulted in less signal distortion (i.e., higher SIG scores) and less noise intrusiveness (i.e., higher BAK scores). Furthermore, with the background of speech-shaped noise, the results tended to trend towards better quality judgments as well (Figure 18). As was discussed previously, this could be related to the background noise type (energetic rather than informational) and the different amount of gain reduction found with the two background noise types studied here.

It is interesting that the status of Synchrony, regardless of whether it is matched or mismatched, did not result in different sound quality ratings. Additionally, no difference was found for signal distortion and overall effect between the filter-on and filter-off conditions. However, noise intrusiveness with all the different filters (i.e., filter-on) was perceived to be reduced compared to the reference schemes (i.e., filter-off). These results suggest that the reduced perceived noise intrusiveness due to the use of gain reduction schemes was not done at the expense of introducing more signal distortion.

Another interesting finding was that the sound quality ratings for Filters 4 and 5 were higher than those for Filters 1, 2, and 3 in terms of signal distortion and noise intrusiveness. This could be due to the background noise used in Filters 4 and 5 being speech-shaped noise. Although speech-shaped noise has a similar frequency spectrum to multi-talker babble, the latter contains contextual information which could more severely degrade sound quality rating. In addition, the amount of gain reductions for Filters 4 and 5 were greater than for Filters 1, 2, and 3. This difference may be due solely to the use of different noise types. Additionally, it was noted that the listening conditions with Filter 3 provided slightly higher signal distortion, more background intrusiveness, and poorer overall sound quality compared to other filter-on conditions. It is possible that the high-frequency gain boost of Filter 3 plus the low-frequency gain reduction caused the reduced sound quality ratings, an explanation that is consistent with previous findings in Gabrielsson, Schenkman, and Hageman (1988). These investigators found that, although listeners may feel the sound is “brighter”, a combination of reduced low-frequency response and increased high-frequency response can lead to reduced sound quality.

It should be noted that the methodologies across studies for measuring sound quality vary substantially. For example, some investigators use paired-comparisons of preference whereas others use categorical ratings or mean opinion scores. The noise type and effect on sound quality rating are varied as well. Therefore, it is difficult to generalize results across studies to real-world environments.

Listening-effort Rating

It is generally accepted that listening in noise requires more effort than listening in quiet (e.g., Pichora-Fuller, Schneider, & Daneman, 1995). Because persons with hearing loss experience greater difficulty in understanding speech in noisy environments than do those with normal hearing, one can speculate that hearing-impaired listeners use greater effort to recognize speech than do normal-hearing listeners. It is not surprising that the subjects of the current study reported the lowest rating on the item “need to concentrate when listening” in the qualities of hearing domain of the SSQ inventory.

The results from the Borg-CR10 scale suggest that the different gain reduction patterns used in this investigation do not result in significantly different perceived listening effort. However, the trend of the results suggests that subjects generally reported more effort in multi-talker babble than in speech-shaped noise. Because multi-talker babble may carry meaningful content (informational masking) and interfere with speech perception more than steady-state speech-shaped noise (energetic masking), subjects may rely more heavily on concentration and other cognitive skill to understand speech in babble than in speech-shaped noise (e.g., Kidd et al., 2005).

It is interesting to note that subjects reported more listening effort was required using Filter 3 than the other filters. Recall that Filter 3 has a low-frequency gain reduction and a high-frequency gain boost pattern. It is unclear how this additional gain in the high frequency region increased the perceived effort for the speech perception task.

Finally, it is noteworthy that the use of gain reduction schemes (matched or mismatched) actually reduced the perception of noise intrusiveness compared to the reference schemes for the sound quality rating, whereas the use of gain reduction schemes did not reduce perceived listening effort compared to the reference schemes. This discrepancy suggests that, although subjects reported less noise intrusiveness using gain reduction schemes, they still had to maintain a level of concentration to recognize speech in noise.

Limitations of the Study

As mentioned previously, it is not easy to generalize laboratory results to the real-life performance for people with hearing loss because of unavoidable limitations. However, we think that the results of the current study still have provided evidence regarding the potential impairment to localization and speech perception due to the use of bilaterally mismatched/matched gain reduction schemes.

One limitation is that we used hearing aid simulation, not actual hearing aids, in a virtual environment to investigate the impact of bilateral gain reductions on localization and speech perception in noise. Consequently, many factors, such as the time constants of DNR algorithms, processing delay within hearing aids, non-linear circuits, and/or environmental factors (e.g., reverberation), were excluded from consideration in the current study.

Another limitation is that performance in only the frontal horizontal plane was assessed. Bilateral mismatched gain reduction in which a large amount of high-frequency gain reduction might severely distort high-frequency spectral cues. This could negatively interfere with front-back discrimination and localization in the vertical plane.

Third, the limitation involves the use of only five gain-reduction patterns. Ideally, the level or gain change across frequencies between ears should be systematically studied to better understand the causal impacts observed.

CONCLUSION

Localization performance in the frontal horizontal plane relies heavily on intact binaural cues. Bilaterally mismatched gain reduction schemes can reduce localization performance due to the distorted ILD cues. However, when the mismatched gain reduction schemes were again matched between ears, the reduced localization performance was shown to improve. Self-reported localization abilities were not found to have a strong relationship with measured localization performance in this study. Although speech perception performance in spatially-separated noise may use the same binaural cues to benefit from spatial separation, the impact of distorted ILD cues was different from that on localization performance. That is, the bilaterally mismatched gain reduction schemes which attenuated the input at the ear with worse SNR actually did not interfere with the speech performance. However, when the gain reduction was matched between ears, speech performance was worsened. In addition, only a weak correlation was found between localization accuracy and speech recognition performance. Finally, the different gain reduction patterns used in the current study did not result in significantly different overall sound quality ratings and listening-effort ratings, although the use of gain reductions reduced the perceived noise intrusiveness regardless of whether it is mismatched or matched compared to the reference schemes.

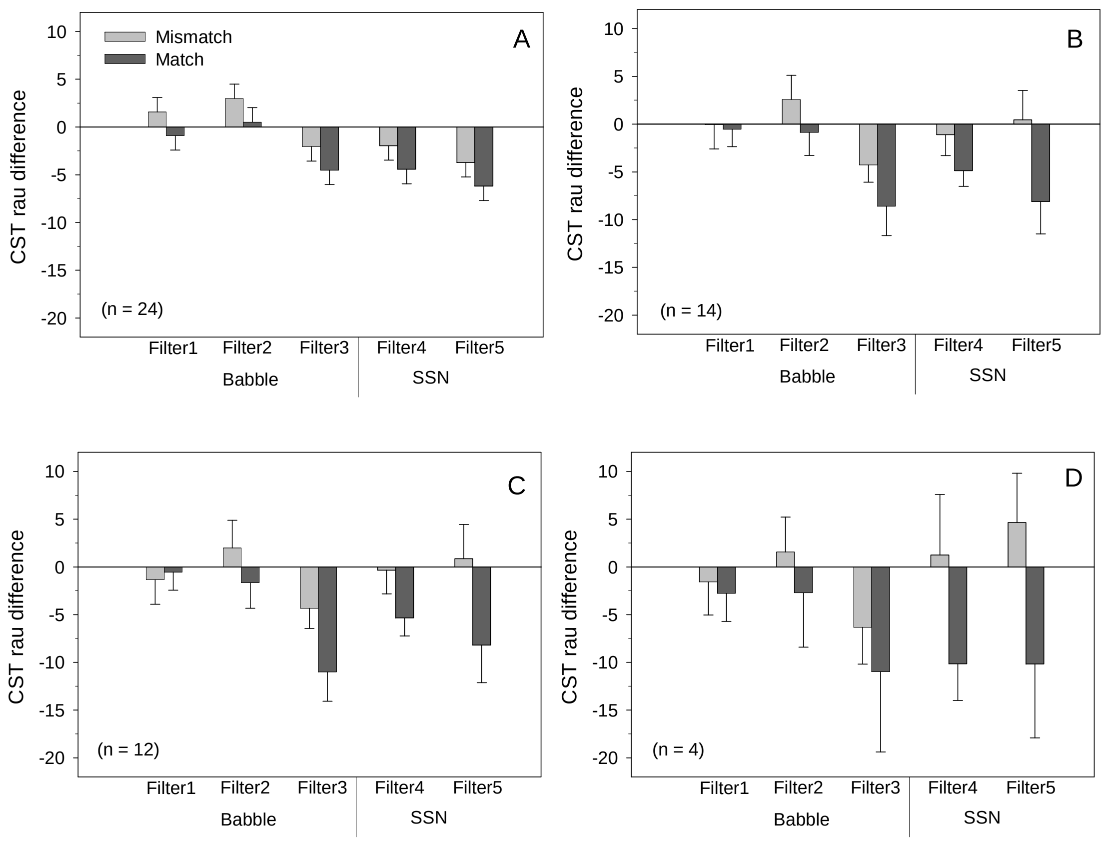

Figure 21. The CST score differences (rau) between the mismatched and reference schemes as well as the difference between the matched and reference schemes are plotted as a function of filter. Values below zero indicate the CST performance was worse than that of the filter-off condition. “SSN” refers to the speech-shaped noise. The error bar displays one standard error. Panel A displays the group data for all 24 hearing-impaired subjects; Panel B displays the group data for 14 hearing-impaired subjects whose raw scores (% correct) are less than 95%. Panel C displays the group data for 12 hearing-impaired subjects whose raw scores (% correct) are less than 90%; Panel D displays the group data for 4 hearing-impaired subjects whose raw scores (% correct) are less than 80%.

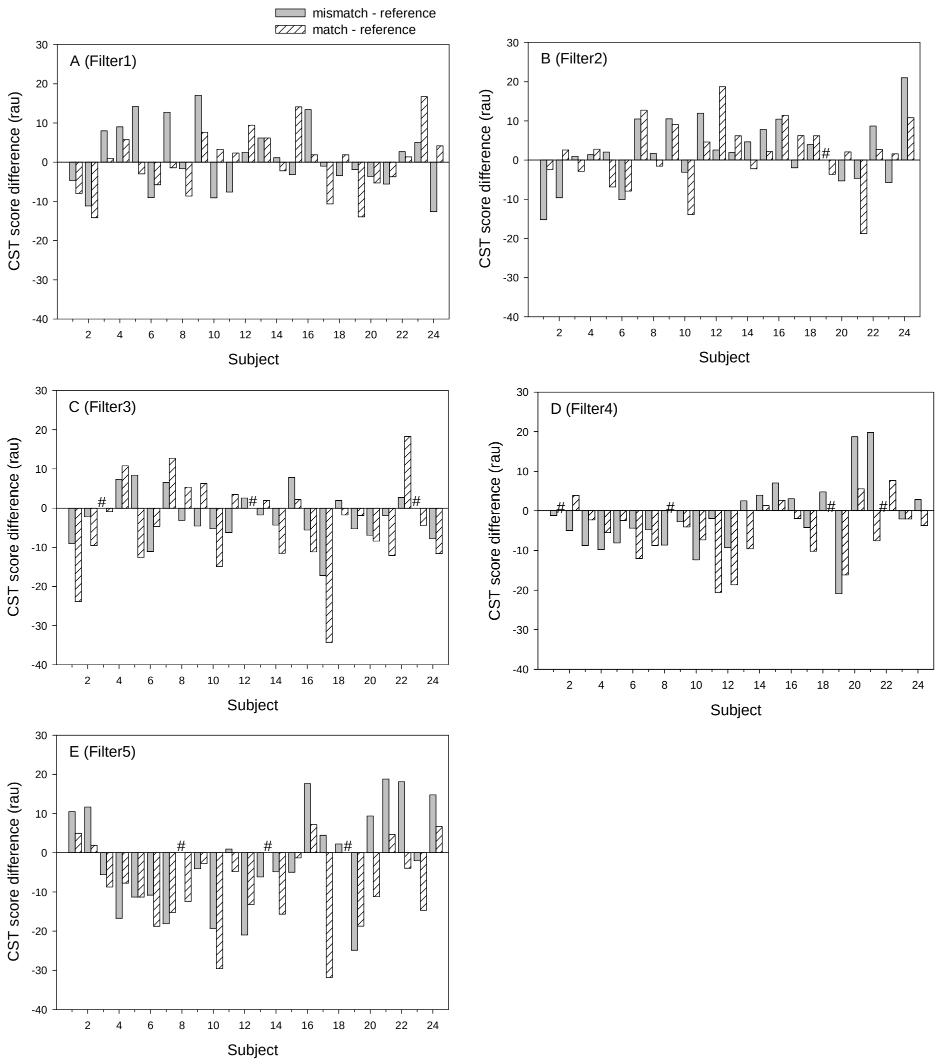

Figure 22. The CST score differences (rau) between the mismatched and reference schemes as well as the differences between the matched and reference schemes are shown for all 24 hearing-impaired subjects. The bar above zero indicates that the score was better than that of reference schemes. Panels A to C show the results for each gain reduction pattern (filter) in the multi-talker babble. Panels D and E show the results for each gain reduction pattern (filter) in the speech-shaped noise. The “#” indicates the score difference was zero.

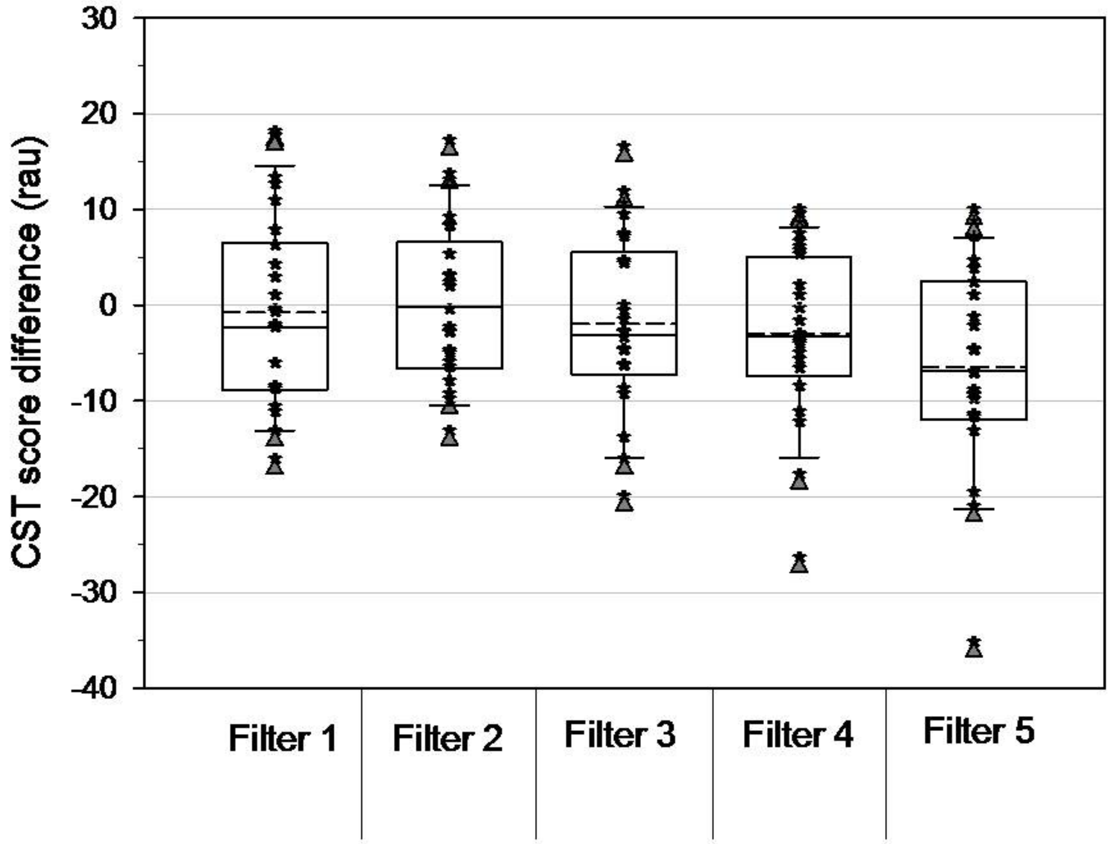

Figure 23. The CST score differences (rau) were derived between the matched and the mismatched conditions and are shown in a box plot for each filter for all 24 hearing-impaired subjects. The values below zero indicate that the speech performance of the matched conditions were worse than that of the mismatched conditions. Inside the box, dashed horizontal lines indicate the mean and solid horizontal lines indicate the median. The box represents the middle 50% of the data. The lower and upper outer lines that encase the box represent the 25th and 75th percentile of the data. The asterisk represents each individual subject. The filled triangle represents outliers for the box plot.

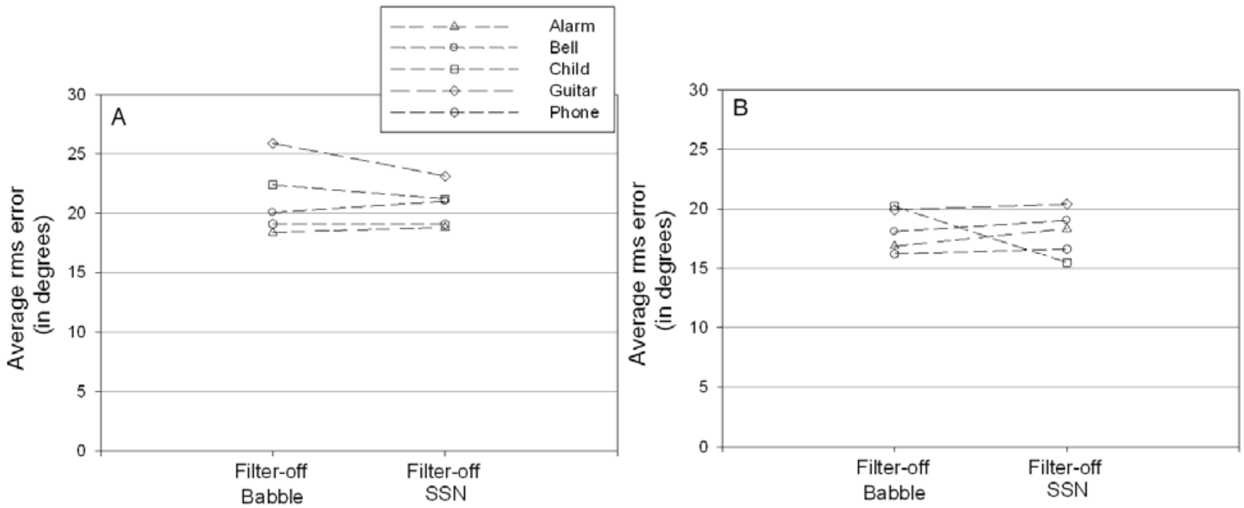

Figure 24. The average rms error for each stimulus in two filter-off conditions is displayed. Panel A is shown for the hearing-impaired group and Panel B is shown for the normal-hearing group.

APPENDIX A: MEASURED FREQUENCY RESPONSE OF ER-2 INSERT EARPHONES

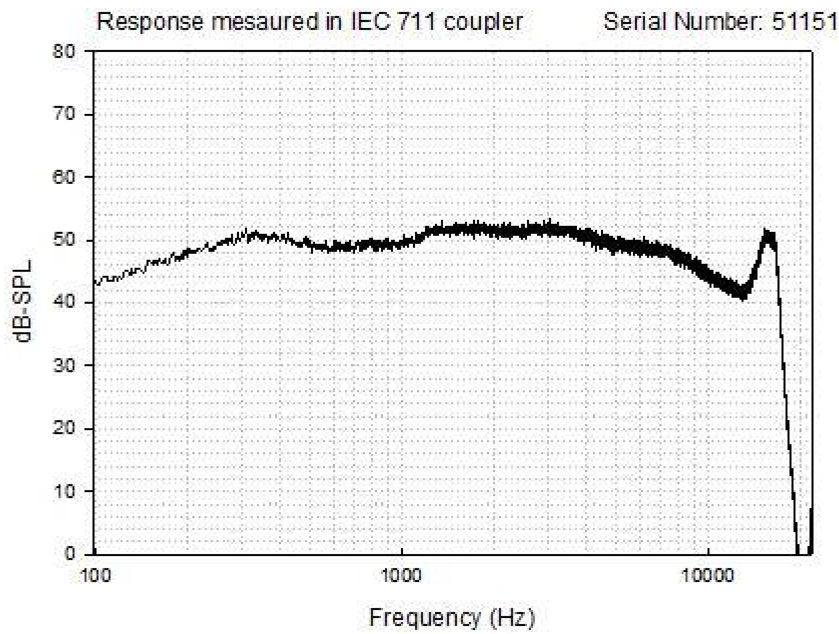

Figure A1. The frequency response of the left insert earphone.

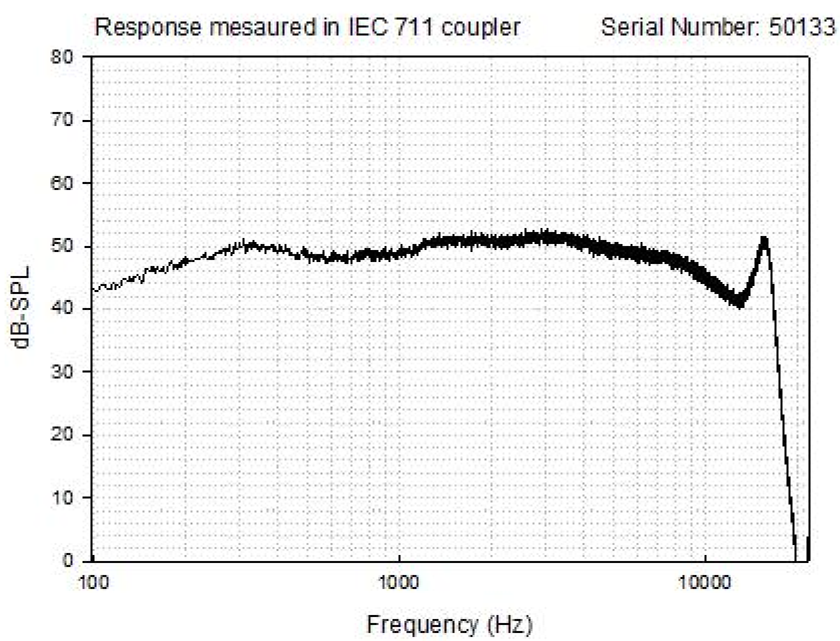

Figure A2. The frequency response of the right insert earphone.

APPENDIX B: FITTING STEP NOTES

If the thresholds at 3kHz are worse than 70dB HL, the formula will limit them to 70dB HL. Both the CST and everyday sounds input levels are at 65dB SPL. The PC sound card setting needs to be restored to the default and then adjusted the output 1 and 2 bar to -6. The rest of the windows volume setting is at the maximum including the Windows Media Player. The Gain is the same for both CST and localization experiment. But the dial level and the position of the External A/B (VU meter) knob should be different. Make sure to record their numbers.

Left Ear Fitting Steps

Step #1

- Let the subject sit in front of the AudioScan and face the REM speaker. The distance will be about 90cm (45-90).

- Measure the REUG

- Use the AudioScan Insertion Gain function

- Type in the audiothresholds into the AudioScan.

- Choose the REUR "measured".

- Put the tube in the open ear canal and the reference microphone should face outwards.

- Choose Pink noise 65dBSPL to get the REUG. Read the values from the table.

Step #2

- Open the MATLAB program

- Define the ear: ear = 'L';

- Define the audiothresholds

- Define REUG

- Run from the MatLab: gainFilter1(ear, LeftTresholds, REUG)

    - Save graphs to C:\myWork\hua\fitting\fittingData

- Put the Insertion Gain and REAG_Target values to C:\myWork\hua\fitting\fittingData\Subject#1\ Sub#1_REAR_ calculation.xls

Step #3

- In the folder: C:\myWork\hua\fitting\fittingData, three files will show up

CST76_Left_FirstFit.wav

CSTCalNoise1minLeft_FirstFit.wav

Left_FirstFit.m

- Open the following two files in the Adobe Audition

CST76_Left_FirstFit.wav

CSTCalNoise1minLeft_FirstFit.wav

Step #4

- Set the audiometer dial at 75dB HL

- Turn off the Amplifier at this moment. Play the CSTCalNoise1minLeft_FirstFit.wav and make sure the VU meter at -10

- Turn on the Amplifier. Play the CST76_Left_FirstFit.wav and measure the REAR from the AudioScan Speechmap Speech Live function (REAR_meas).

- Print the table from the AudioScan to the PC under C:\verifit\CST\SubjectID.

- Pre-build a folder named after the subject. Save the final REAR_meas table graph under this table.

- Type the measured REAR to the template (C:\myWork\hua:FinalTemplate of REAR calculation.xls) and adjust the dial to make sure the values of REAR_meas are close to the REAR_target as possible.

- In this case, the dial usually will be increased. But 75dB HL would be a good point to start.

- Calculate the average Filter Adjustment and round it to get the dial change. e.g, “4” means to increase the dial by 4 dB. “-1” means to decrease the dial by 1 dB.

- Because the whole system should be linear, we can expect the rms error after the dial change. e.g., if the Measured REAR = 61, 68, 77, 80, 80, 79, 75 at each frequency, the rms error is 5.50. By adjusting the dial -2dB, we can expect the measured REAR would be = 59, 66, 75, 78, 78, 77, 73. Now the rms error is 1.98. By doing this, the number of repetition can be limited.

Step #5

- Put the new measured REAR (REAR_meas or expected REAR_meas) into gainFilter2.
- Run gainFilter2(ear,REAR_meas)
- The new fitting data will be saved in C:\myWork\hua\fitting\fittingData

  CSTCalNoise1min_Left_Fit.wav

  CST40f3_Left_Fit.wav (to measure the biggest peak would not be over 0)

  CST76_Left_Fit.wav

  Left_Fit.m

- Turn off the Amplifier. Play the CST40f3_Left_Fit.wav, which has the biggest peak. Make sure the max VU meter level is 0. Repeat the largest peak from the Adobe and watch from the Audiometer.
- Use the CSTCalNoise1min_Left_FirstFit.wav to determine how much gain has been reduced by adjusting the VU meter. The previous VU meter reading is -10.
- Remember the approximate change of the VU meter and adjust the dial accordingly. e.g., If the VU meter was deceased 5dB in the last step, increase the dial 5dB now.
- Turn on the Amplifier.
- Measure the REAR and put the values into the template.
- Usually the rms error should be within 5dB now. That means 3 measurements of REAR_meas should be plenty. If the rms error is far, then the measurement is not

- Double check the level from CST40f3_Left_Fit.wav. Make sure the level would not be too high to exceed the level that the insert earphone cannot tolerate.

Step #6

- Write down the dial level on the Audiometer in the FinalTemplate of REAR calculation.xls (Remember to measure the 2cc Coupler level at this setting for both cal_noise and CST76 at the end of fitting)

- Mark the position of external A!!!

Step #7

- Run gainFilter3(ear) (scale by 0.99)
- The input level is 65dBSPL for 5Comb

  EDS_calpink_Left_Fit.wav

  5Comb_Left_Fit.wav

  BellSNR-5S045N090_Left_Fit.wav

  These three files will be saved in C:\myWork\hua\fitting\EDSfittingData
- Turn off the Amplifier
- Use the CSTCalNoise1min_Left_FirstFit.wav to determine how much gain has been reduced by adjusting the VU meter. The current VU meter reading* is__-18_.
- Play the BellSNR-5S045N090_Left_Fit.wav
- Use the same dial as the CST but start the External A knob at minimum. Thus it can make sure the max VU meter reading is at **0.** Repeat the largest peak from the Adobe and watch from the Audiometer.
- Use the CSTCalNoise1min_Left_FirstFit.wav to determine how much gain has been reduced by adjusting the VU meter. The current VU meter reading is_-19.5__ and

compare to the previous one (current VU meter reading* from above) _-18__. What is the change? __1.5 or 2dB_

- Remember the approximate change of the VU meter and adjust the dial accordingly. e.g., if the VU meter was deceased 5dB in the last step, then increase the dial 5dB now.
- Turn on the Amplifier.
- Play the 5Comb_Left_Fit.wav
- Put the REAR_EDS_meas in the Template of REAR calculation.xls
- Make sure the rms error is within the limits (5dB rms)
- Check the level of BellSNR-5S045N090_Left_Fit.wav

Step #8

- Back to the MATLAB program
- Type: REAR_EDS_meas =
- Run checkEDS(ear, REAR_EDS_meas)

Step #9

- Check the connection of the resistor BOX to make sure the resistance is still 22 ohms.
- The resistor BOX is behaving like a fuse to protect the ER-2 insert earphones.

Right Ear Fitting Steps

Repeat the same fitting steps for the left ear

Step #10 for both ears (final step)

- Generate a new .wav file and combine the left and right ear channel together
- Reduce 3dB for each channel because it is a bilateral fitting
- Make sure that both channels sound equally loud, and check the comfortableness using the IHAFF rating. Adjust the Audiometer dial if necessary.

- If any changes are made, re-measure the REAR and save the data. Make sure this is the final dial setting.
- rms error should be similar between ears.

APPENDIX C: BORG-CR 10 INSTRUCTION

- Use this rating scale to report how strong your perception is. It can be exertion, pain or something else.
- First look at the verbal expressions. Start with them and then the numbers. It’s very important that you report what you actually experience or feel, not what you think you should report. Be as spontaneous and honest as possible and try to avoid under- or overestimating. Look at the verbal descriptors and then choose a number.
- Perceived exertion: When rating perceived exertion give a number that corresponds to how hard and strenuous you perceive the work to be. The perception of exertion is mainly felt as strain and fatigue in your muscles and as breathlessness or any aches.

For perceived exertion note the following:

0 ”Nothing at all”, means that you don’t feel any exertion whatsoever, e.g. no muscle fatigue, no breathlessness or difficulties breathing.

1 “Very weak” means very light. As taking a shorter walk at your own pace.

3 “Moderate” is somewhat but not especially hard. It feels good and not difficult to go on.

5 “Strong”. The work is hard and tiring, but continuing isn't terribly difficult. The effort and exertion is about half as intense as “Maximal”.

7 “Very strong” is quite strenuous. You can go on, but you really have to push yourself and you are very tired.

10 “Extremely strong – Maximal” is an extremely strenuous level.

For most people this is the most strenuous exertion they have ever experienced.

• is ”Absolute maximum – Highest possible”, for example ”12” or even more.

0	Nothing at all	
0. 3		
0.5	Extremely weak	Just noticeable
0.7		
1	Very weak	
1.5		
2	Weak	Light
2.5		
3	Moderate	
4		
5	Strong	Heavy
6		
7	Very strong	
8		
9		
10	Extremely strong	"Maximal"
11		
÷		
•	Absolute maximum	Highest possible

The Borg CR10 scale

APPENDIX D: ITU-T P.835 METHOD

INSTRUCTION

Attending ONLY to the BACKGROUND, select the category which best describes the sample you just heard.

The BACKGROUND in this sample was

5 – Not noticeable

4 – Somewhat noticeable

3 – Noticeable but not intrusive

2 – Fairly conspicuous, somewhat intrusive

1 – Very conspicuous, very intrusive

Attending ONLY to the SPEECH SIGNAL, select the category which best describes the sample you just heard.

The SPEECH SIGNAL in this sample was

5 – Very natural, no degradation

4 – Fairly natural, little degradation

3 – Somewhat natural, somewhat degraded

2 – Fairly unnatural, fairly degraded

1- Very unnatural, very degraded

Select the category which best describes the sample you just heard for purposes of everyday speech communication.

The OVERALL SPEECH SAMPLE was

5 – Excellent

4 – Good

3 – Fair

2 – Poor

1 – Bad

APPENDIX E: SPECTRUM AND TIME WAVEFORM OF FIVE EVERYDAY SOUNDS

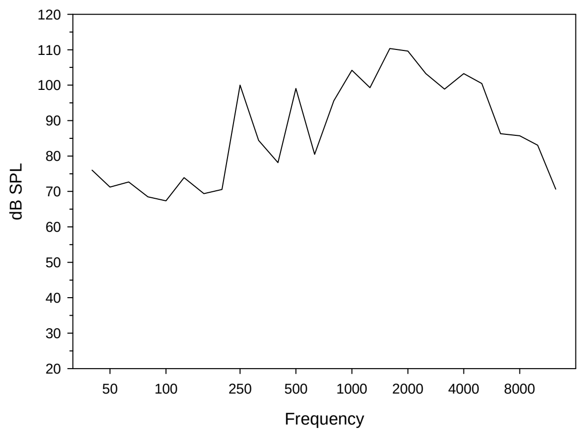

Figure E1. The spectrum of Stimulus Alarm.

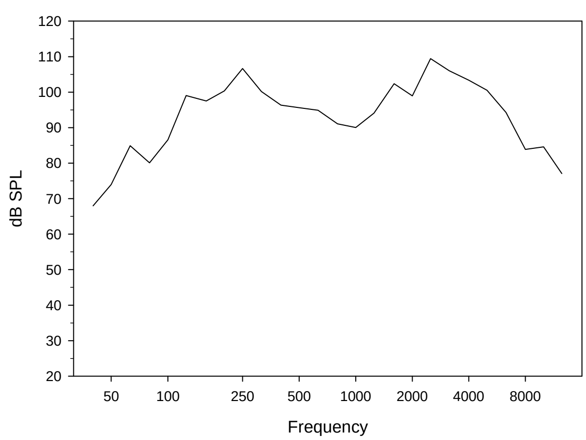

Figure E2. The spectrum of Stimulus Bell.

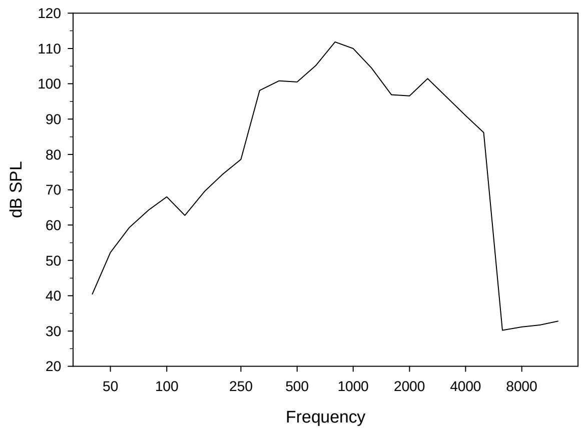

Figure E3. The spectrum of Stimulus Child.

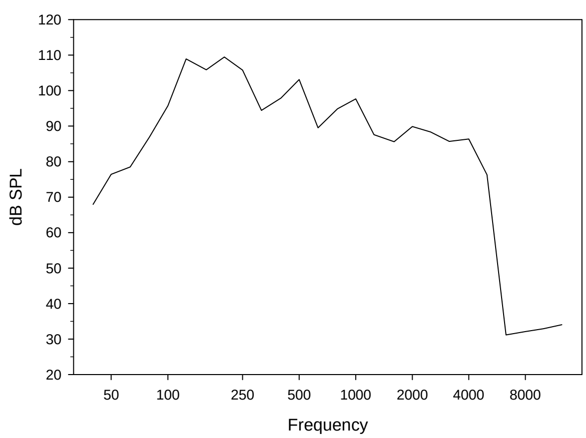

Figure E4. The spectrum of Stimulus Guitar.

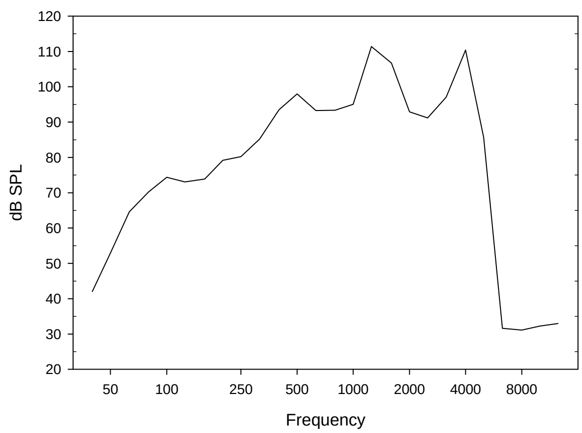

Figure E5. The spectrum of Stimulus Phone.

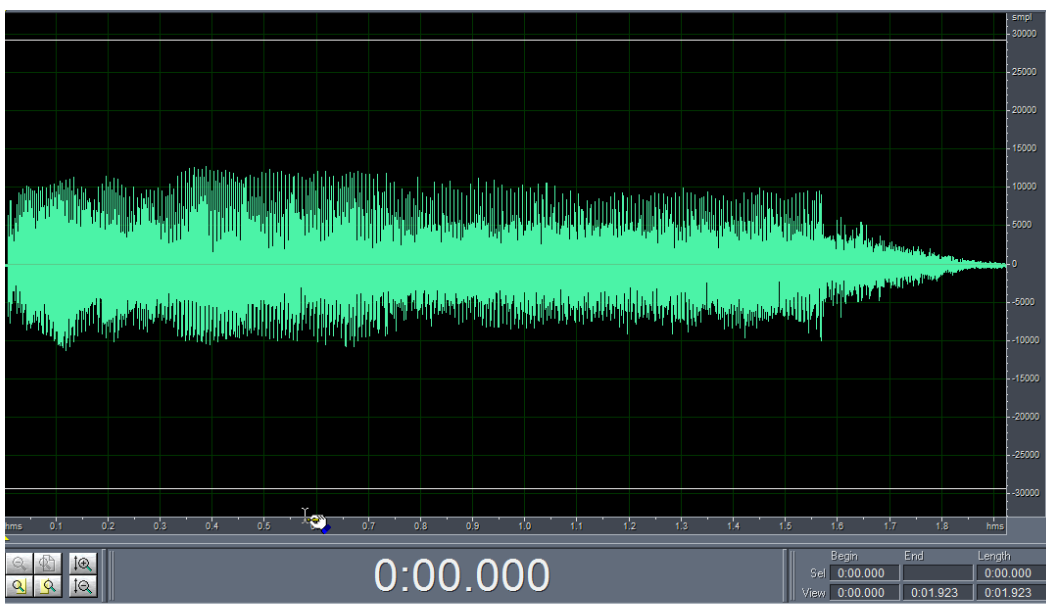

Figure E6. The time waveform of Stimulus Alarm.

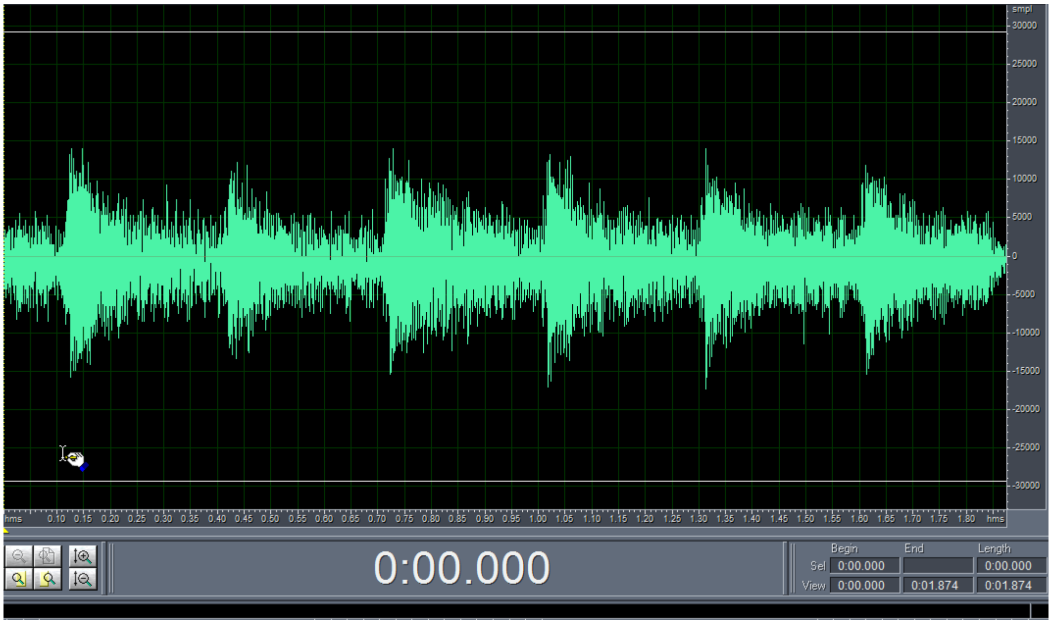

Figure E7. The time waveform of Stimulus Bell.

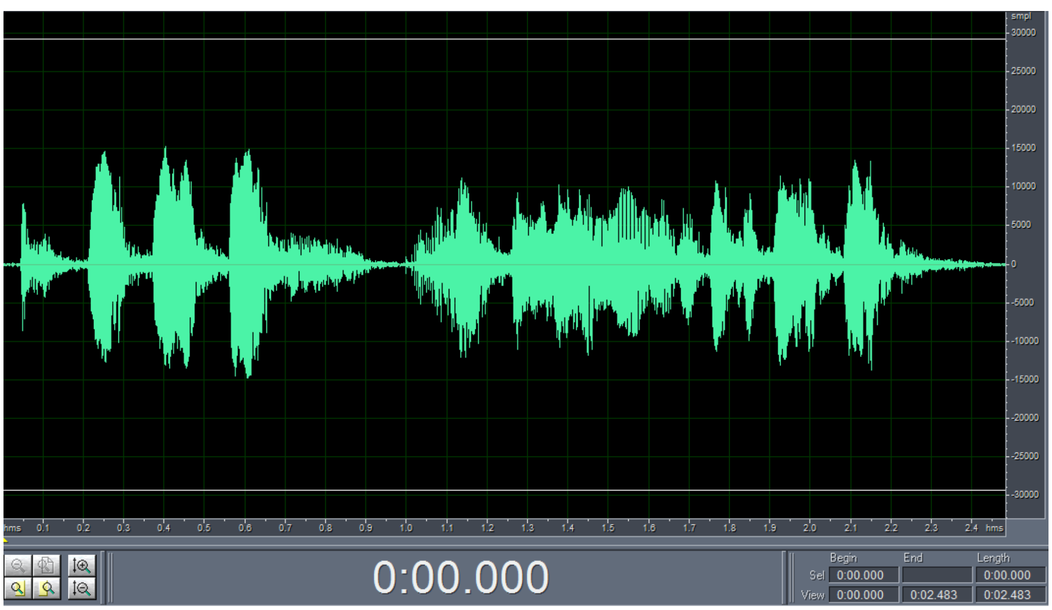

Figure E8. The time waveform of Stimulus Child.

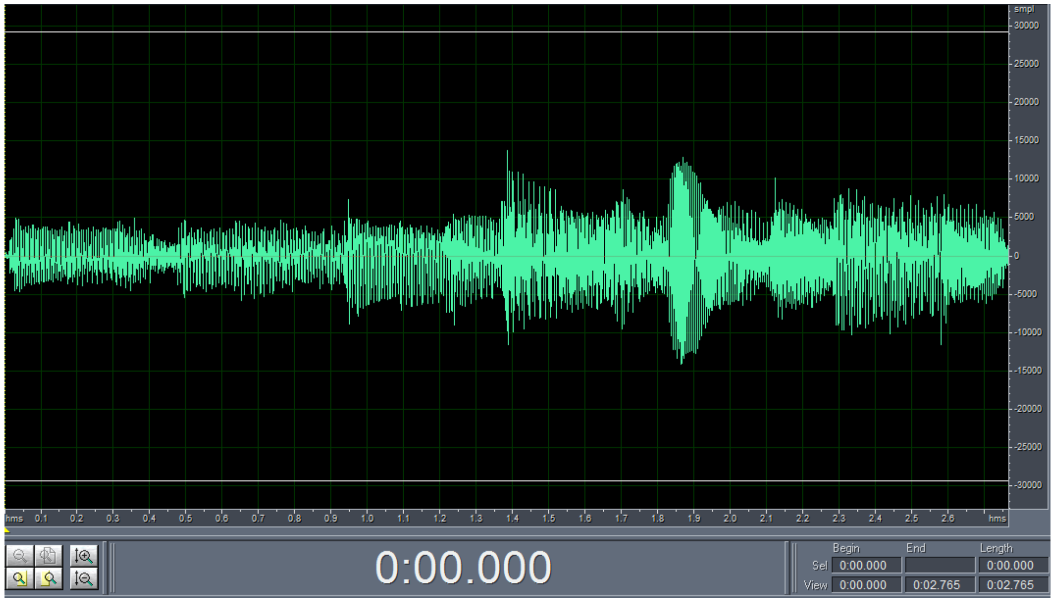

Figure E9. The time waveform of Stimulus Guitar.

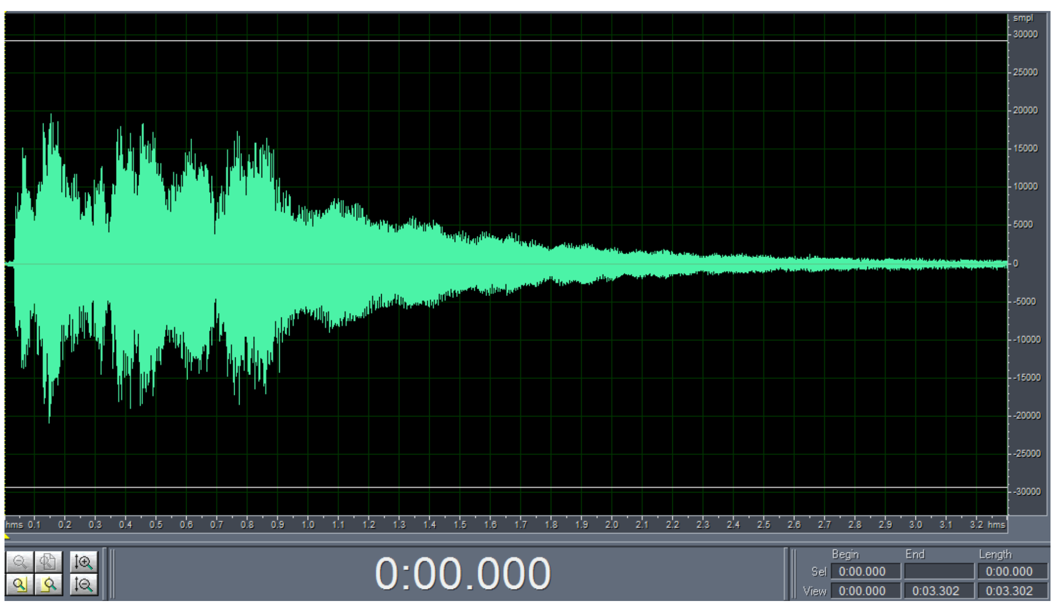

Figure E10. The time waveform of Stimulus Phone

APPENDIX F: SPEECH, SPATIAL, AND QUALITIES OF HEARING (SSQ) SAMPLE AND ITEM DESCRIPTIONS

Speech hearing items

1	Talk with one person with TV on
2	Talk with one person in quiet room
3	Talk with five people in quiet with visual input
4	Talk with five people in noise with visual input
5	Talk with one person in background noise
6	Talk with five people in noise without visual input
7	Conversation in echoic environment
8	Ignore interfering voice of same pitch
9	Ignore interfering voice of different pitch
10	Talk with one person and follow TV
11	Follow one conversation when many people talking
12	Follow conversation without missing start of new speaker
13	Have conversation on telephone
14	Follow one person speaking and telephone at same time

Spatial hearing items

1	Locate lawnmower
2	Locate speaker round a table
3	Lateralize a talker to left or right
4	Locate a door slam in unfamiliar house
5	Locate above or below on stairwell
6	Locate dog barking
7	Locate vehicle from footpath
8	Judge distance from footsteps or voice
9	Judge distance of vehicle
10	Identify lateral movement of vehicle
11	Identify lateral movement from voice or footsteps
12	Identify approach or recede from voice or footsteps
13	Identify approach or recede of vehicle
14	Internalization of sounds
15	Sounds closer than expected
16	Sounds further than expected

17 Sounds in expected location

Qualities items

1 Separation of two sounds
2 Sounds appearing jumbled
3 Music and voice as separate objects
4 Identify different people by voice
5 Distinguish familiar music
6 Distinguish different sounds
7 Identify instruments in music
8 Naturalness of music
9 Clarity of everyday sounds
10 Naturalness of other voices
11 Naturalness of everyday sounds
12 Naturalness of own voice
13 Judging mood from voice
14 Need to concentrate when listening
15 Sounds unnaturally quiet when hear from one aid
16 Understand when driver of a car
17 Understand when car passenger
18 Effort of conversation
19 Ability to ignore competing sounds

1. You are talking with one other person and there is a TV on in the same room. Without turning the TV down, can you follow what the person you're talking to says?	Not at all Perfectly 0 1 2 3 4 5 6 7 8 9 10 Min Max	tick if not applicable ☐ or wouldn't hear it	aid not used ☐

(Noble & Gatehouse, 2004)

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