TOWSON UNIVERSITY

OFFICE OF GRADUATE STUDIES

EVALUATING THE EFFECTS OF DEREVERBERATION TECHNIQUES ON SPEECH UNDERSTANDING IN DIFFERENT REVERBERANT ENVIRONMENTS

by

Michael DiSanti

A Dissertation

Presented to the faculty of

Towson University

in partial fulfillment

of the requirements for the degree

Doctor of Audiology

Department of Audiology, Speech-Language Pathology, and Deaf Studies

Towson University
Towson, Maryland 21252

May, 2018

TOWSON UNIVERSITY
OFFICE OF GRADUATE STUDIES

THESIS APPROVAL PAGE

This is to certify that the thesis prepared by Michael DiSanti, B.S. entitled "Evaluating the Effects of Dereverberation Techniques on Speech Understanding in Different Reverberant Environments" has been approved by the thesis committee as satisfactorily completing the thesis requirement for the degree of Doctor of Audiology, Au.D.

Nirmal Srinivasan, Ph.D. | 5/4/18
Chairperson, Thesis Committee | Date

Saradha Ananthakrishnan, Ph.D. | 5/4/18
Committee Member | Date

Peggy Korczak, Ph.D. | 5-4-18
Committee Member | Date

Janet Delany | 5-21-18
Dean of Graduate Studies | Date

Acknowledgements

I wish to thank the faculty of the audiology department of Towson University. I don't think I could say enough how much you mean to me. I would also like to express gratitude to my supervisor Nirmal Srinivasan for his unwavering support, guidance and many hours helping me throughout this research project. Thanks also to my thesis committee members Peggy Korczak, Saradha Ananthakrishnan, and Tricia Ashby for valuable help in the preparation of this thesis document and for keeping me inspired and motivated. Thank you to my family for their unconditional love and amazing support through the journey of becoming a future audiologist. I would also like to include a special note of thanks to Catherine DiSanti, my mother, for always listening and keeping me calm during the many challenges faced over the course of this program.

ABSTRACT

Reverberation and background noise is present in almost all everyday listening situations. Reverberation times in real rooms vary widely from nearly several milliseconds up to several seconds based on the absorptive properties of the materials present in a room. In a reverberant listening environment, listeners receive speech through three main components: direct sound, early reflections, and late reflections. Older adults with hearing loss are more susceptible to the negative effects late reflections have on speech. On the other hand, previous studies have indicated that early reflections are associated with improved speech understanding. With that being said, it is surprising that even with the availability of de-reverberation processing strategies, most hearing aid manufacturers do not implement these processing strategies in their devices. The effects of a dereverberation processing strategy on speech understanding in two reverberant environments (T_{60} = 1s and T_{60} = 2s) were evaluated in this experiment. Target-to-Masker (TMR) identification thresholds from the participants in this study were analyzed in terms of 3 different conditions. These conditions include: three different reverberation times (i.e., T_{60} = 0s, T_{60} = 1s, and T_{60} = 2s); two different spatial configurations (colocated and spatially separated), and three different dereverberation processing techniques (i.e., correct, over, and underestimation). The results of experiment 1 demonstrated that the listener's ability to identify the target call-sign in a multi-talker environment improved with spatially separating the target speaker from the masker speakers. Consequently, as reverberation increased speech understanding was significantly reduced. In experiment 2, the underestimation Binaural Room Impulse Response elicited the lowest TMR identification thresholds. The conditions were ordered:

Under, Correct, and Overestimation (going from the lowest to the highest TMR). Multiple regression analyses predicting the amount of Spatial Release from Masking using age and PTA indicated that only PTA was significant in predicting SRM in the correct estimation condition in the reverberant condition of 2 seconds. The results of this study provide evidence that adding a dereverberation program utilizing an underestimation processing strategy as an option for hearing aid users could improve speech understanding and reduce listening effort in reverberant environments.

TABLE OF CONTENTS

Approval Page i

Acknowledgements ii

Abstract ..... iii-iv

List of Tables vii

List of Figures viii

Key to Abbreviations ix

Chapter 1: Introduction 1-2

Chapter 2: Review of The Literature 5-41

- **Hearing Loss: Definition and Core Features** 7-9
- **Masking** 10-12
  - **Energetic Masking** 11
  - **Informational Masking** 12
- **Spatial Release from Masking in Speech Perception** 12-21
  - **Effects of Hearing Loss on Spatial Release from Masking** 15-17
  - **Effects of Aging on Spatial Release from Masking** 19-22
- **3 Primary Cues for Localization** 22-23
- **Localization** 24-31
  - **Sound Localization in Free Field** 26
  - **Sound Localization Under Headphone Presentation** 27
  - **Sound Localization in Reverberation** 28
  - **Precedence Effect** 29
- **Reverberation** 32
  - **Binaural Room Impulse Response** 33
  - **Early Reflections** 34-36
  - **Late Reflections** 36-39
- **Effects of Reverberation on SRM** 39-40
- **Effects of Hearing Loss and Age on the Benefit of Spatial Release from Masking in a Reverberant Environment** 40-41

Purpose Statement 41

Chapter 3: Methodology 42-52

Participants 42-43

Procedure 43-44

Pure-Tone Behavioral Audiometric Test Protocol 44-45

Stimulus 45

Binaural Room Impulse Response Modeling Procedure 46-49

Procedure 50

Threshold Estimation 50-51

Data Analysis 52

Chapter 4: Results 53-73

Identification Thresholds in Anechoic Listening Condition 54

Identification Thresholds of Anechoic (0s) and Reverberant Environments (1s & 2s) 55-56

Repeated Measures ANOVA (3 BRIR X 2 Reverb Times X 2 Spatial Conditions) 60-67

Multiple Regression and Correlation Models Predicting SRM Based on Various Potential Predictors 67-73

Chapter 5: Discussion 74-81

Experiment 1: Identification Thresholds in Anechoic (0s) and Reverberant Environments (1s & 2s) 74-77

Experiment 2: Identification Thresholds in Two Different Reverberant Environments (1s & 2s) Based on Three Different BRIR Estimation (Correct, Over, And Under Estimation) 77-81

Chapter 6: Conclusion 82-84

References 85-93

Appendices 95-99

Appendix A: IRB Approval 96

Appendix B: Consent Form 97

Appendix C: Flyer 98

Appendix D: Case History Questionnaire 99

Appendix E: VASLUMs Cognitive Screener 100

Appendix F: Incentive Form 100

Curriculum Vitae 101-102

LIST OF TABLES

Table 1: Paired Sample T-Test for Spatial Release from Masking for the Anechoic Condition 55

Table 2: Descriptive Statistics for Identification Thresholds for Experiment 1 59

Table 3: Paired Sample T-Test for Spatial Release from Masking for Experiment 1 59

Table 4: Descriptive Statistics for Identification Thresholds for Experiment 2 64

Table 5: Descriptive Statistics for Spatial Release from Masking for Experiment 2 66

Table 6: Descriptive Statistics and Correlation of Identification Thresholds and Spatial Release from Masking of Age and Pure-tone Averages 71

Table 7: Descriptive Statistics and Correlation of Identification Thresholds and Spatial Release from Masking of Age and Pure-tone Averages Based on Two Different Reverberant Listening Environments...72

Table 8: Multiple Regression Models Predicting Spatial Release from Masking...........74

LIST OF FIGURES

Figure 1: Illustration of Better-Ear Advantage 17

Figure 2: Illustration of Angles of Azimuth 23

Figure 3: Illustration of Original and Reverberant Speech Signal 37

Figure 4: Average thresholds for all participants across 250-8000 Hz (dB HL)................ 43

Figure 5: Binaural Room Impulse Response Modeling Procedure..................................... 46

Figure 6: Block Diagram of Signal Processing Technique... 49

Figure 7: Experiment 1 Results Based on Target-to-Masker Identification Thresholds.. 56

Figure 8: Experiment 1 Results Based on Spatial Release from Masking.......................... 58

Figure 9: Experiment 2 Results Based on Target-to-Masker Identification Tresholds.... 63

Figure 10: Experiment 2 Results Based on Spatial Release from Masking........................ 65

Figure 11: Scatterplot Generated from SPSS for Experiment 2 69

KEY TO ABBREVIATIONS

ANSI: American National Standards Institute

BRIR: Binaural room impulse response

dB: Decibel

dB HL: Decibel hearing level

Hz: Hertz

HRTFs: Head-related transfer functions

IRB: Institution of Research Board

OHI: Older hearing-impaired

ONH: Older normal hearing

PTA: Pure tone average, average of thresholds at 500, 1000, 2000, and 4000 Hz.

SRM: Spatial release from masking

TMR: Target-to-Masker Ratio

YNH: Younger normal hearing

CHAPTER 1

Introduction

Frequently people encounter complex auditory environments where it proves difficult to attend to one speaker in the presence of multiple talkers. For successful communication in these auditory settings, the listener must process the target speech of interest while simultaneously ignoring the sources of masking (Alain, 2007; Bremen & Middlebrooks, 2013). When the sources of masking are spatially separated from the signal of interest, a listener can take advantage of acoustic cues arising from the spatial separation. This benefit of spatial separation between the target signal and the source of masking is known as spatial release from masking (SRM) (Brungart, Simpson, Ericson, & Scott, 2001; Freyman, Balakrishnan, & Helfer, 2001; Freyman, Helfer, McCall, & Clifton, 1999; Hawley, Litovsky, & Culling, 2004).

Reverberation is present in almost all everyday listening situations. The effects of reverberation on speech intelligibility has been thoroughly investigated in normal hearing individuals (Breitsprecher, 2011; Sudirga, 2014) and in individuals with age-related hearing loss (Helfer, 1992; Helfer & Wilber, 1990; Marrone, Mason, & Kidd, 2008; Srinivasan, Stansell, & Gallun, 2017) A few prior studies have provided some evidence that older individuals with hearing impairment have greater difficulty understanding speech in moderate amounts of reverberation, whereas, moderate amounts of reverberation had no significant impact on speech intelligibility for younger normal hearing individuals (Breitsprecher, 2011; Marrone et al., 2008). However, the results of previous studies indicated that the individual effects of aging on SRM are so small it could not be detected in individuals with hearing loss (Glyde et al., 2013; Marrone et al., 2008). Srinivasan, Jakien, and Gallun (2016) used smaller spatial separations (i.e., between target signal and masker signals) to isolate the effects of age from hearing loss on SRM. Srinivasan and colleagues (2016) found that in order to measure the effects of aging independent from the effects of hearing loss on SRM, small separations between the target signal and masker signals should be utilized. More recently, Srinivasan et al. (2017) studied the effects of reverberation on spatial release from masking for listeners varying in age and hearing ability. Collectively the results of these studies indicate that older adults with hearing loss are more susceptible to the negative effects reverberation has on speech intelligibility.

Srinivasan et al. (2017) theorized that dereverberation can improve speech intelligibility by removing the noise and late reflections of the room impulse response. To date, few studies have been conducted to investigate the effects of dereverberation algorithms to improve speech intelligibility. Therefore, there is the need to investigate a Dereverberation program to improve speech intelligibility and the perceived reduction in listening effort in these reverberant environments. Thus, the purpose of this study is to evaluate a dereverberation processing technique based on estimating the binaural room impulse response (BRIR) of the listening environment on spatial release from masking.

In the proposed research, the effects of BRIR on speech understanding in older hearing impaired (OHI) participants will be investigated. In this environment, the spatial cues will be absent. BRIR is a representation of the acoustical behavior of a room and is largely determined by reverberation time. Given that reverberation is thought to be a significant factor in the reduction effect on SRM, it is important to include the investigation of direct signals, early reflections, and late reflections. We reasoned that as the reverberation increased, the benefit received from spatially separating sources would decrease. We are also interested in determining the effects of reverberation in the context of age-related hearing loss.

Reverberation times in real rooms vary widely from nearly zero up to several seconds based on room size and absorptive properties of the room surfaces and materials. The proposed study, however, is concerned primarily with smaller room listening environments which are more representative of everyday listening environments in which the vast majority of listening takes place. Therefore, in the proposed study, a dereverberation processing technique will be used to assess speech intelligibility in two different reverberant environments (T_{60} = 1 and 2 s). These effects will be studied in three different experimental conditions which include: underestimated effects of reverberation, overestimated effects of reverberation, and correct estimation of reverberation. The dereverberation technique uses non-individualized head related transfer functions (HRTFs) to spatially render the target signal and the masker signals for each ear. This simulation method has been found to accurately reproduce BRIRs measured in a real room (Zahorik, 2009).

Some hearing aid manufacturers utilize proprietary dereverberation algorithms in their hearing aid devices [i.e., Echoblock (Phonak), EchoShield (Cigna)]. Some of the disadvantages of the available techniques are that the dereverberation algorithms need the original clean signal to perform the dereverberation.

However, in reality, we do not always have access to the real signal. The aim of this research is to aid in the development of dereverberation algorithms for hearing aids which may help improve speech recognition in reverberant listening conditions for all populations with hearing loss.

CHAPTER 2

Review of the Literature

Speech Perception in Everyday Life

The process of understanding speech in challenging listening situations presents a complex problem for the auditory system. In various listening situations, the speaker's voice arrives at the listener's ear often mixed with other sounds. These sounds are referred to as sources of masking and include distracting noise, simultaneous talkers, and reverberation. These potential sources of masking are numerous and can negatively affect a listener's ability to perceive and comprehend a speech signal (Gelfand, 2010, Chapter 10). Most normal-hearing individuals can understand a conversation partner in reverberant listening environments without much difficulty (Bronkhorst, 2015). The ability for a typical listener to concentrate, direct one's attention to their conversation partner, and make sense of speech in a mixture of competing sources relies on several processes (Gatehouse & Akeroyd, 2006). These auditory processes include sound localization and lateralization, auditory discrimination, temporal aspects of audition (i.e., resolution, masking, ordering), auditory performance with competing signals, and auditory performance with degraded acoustic signals (ASHA, 2005). The sensory system for the sense of hearing uses multiple spatial and spectro-temporal cues to localize the position of sounds in the auditory scene. Similarly, localization of sound streams in an auditory environment is an important process that is used to segregate speech from noise (Bregman, 1990).

Spatially separating the target signal from the noise produces binaural difference cues that help the auditory system enhance the target signal while suppressing the responses to distracting noise in a complex environment (Arbogast, Mason, & Kidd, 2005; Breitsprecher, 2011; Gallun, Diedesch, Kampel, & Jakien, 2013). These acoustic cues include: fundamental frequency; intensity and timing differences; onsets and offsets; and spatial location of the target and masker signals (Alain, 2007; Breitsprecher, 2011; Srinivasan et al., 2017; Stecker & Gallun, 2012). Bronhorst (2015) indicated that the auditory system benefits significantly from the ability to process acoustic signals with two ears (i.e., binaural hearing). Binaural hearing allows the listener to take advantage of the multiple spatial and spectro-temporal cues, which in turn, results in improved speech perception and sound localization (Mason, 2011). Cherry (1953) was primarily interested in the challenges associated with understanding target speech in the presence of multiple-talker masking speech in a complex listening environment. Below is a description of the early studies conducted in this area.

History of the Cocktail Party Effect

Cherry (1953) was among one of the first researchers to study the effects of simultaneous talkers on an individual's ability to understand speech. The problem previously described is often referred as the "cocktail party effect". Cherry (1953) defined the "cocktail party effect" as a complex listening environment that contains multiple speech sources and often involves the difficulty of understanding one target talker while ignoring multiple simultaneous talkers. From this research, it seems evident that listeners with normal hearing can direct their attention to the talker of interest, and understand speech in these complex listening environments with little difficulty (Cherry, 1953). Consequently, the degree to which speech perception is impaired is strongly influenced by the age and hearing status of the listener (Breitsprecher, 2011; Gallun et al., 2013; Glyde et al., 2013; Marrone et al., 2008; Srinivasan et al., 2016). In order to understand why individuals with hearing loss have difficulty in these “Cocktail party” environments, it is essential to understand how the auditory system processes sound.

Hearing Loss: Definition and Core Features

Hearing loss is often variable, and there is no one single underlying cause. Hearing loss can be caused by a variety of risk factors, with the most common causes including noise-induced hearing loss and age-related-hearing loss (Hearing Loss Association of America [HLAA], 2017). The stages of the hearing pathway are extensive and will be reviewed briefly below in order to understand the impacts of age-related hearing loss on peripheral and central auditory function.

The auditory system creates a neural representation of the acoustic world based on spectral and temporal cues present at the listener’s ear, including cues that potentially signal the locations of sounds (Bremen & Middlebrooks, 2013). The auditory system acts to convey acoustic messages by transducing sound vibrations into electrophysiological signals (i.e., electrical energy) (Howarth & Shone, 2006). The pinna filters sound through the ear canal and directs the sound waves into the middle ear space. Sound pressure waves enter the middle ear space, vibrate the eardrum, and the mechanical vibrations are transmitted to the inner ear via the ossicular bones. These mechanical vibrations create a displacement of the fluid in the inner ear, which stimulates the basilar membrane. The result is a vibration pattern that generates a traveling wave along the basilar membrane. When the fluid is displaced, the basilar membrane is set into motion and a force is applied to the hair cells. The hair cells are situated on the basilar membrane in a fluid known as perilymph. The site where mechanical vibrations are transduced into electrophysiological energy is known as the cochlea. The cochlea performs a significant amount of signal analysis (i.e., frequency, intensity, and timing information). Hudspeth (2005) explained how this mechanoelectrical transduction by the hair cell is generated. When the hair cell bundle is deflected, this reaction opens transduction channels which produce depolarizations of the hair cells. The hair cells, when displaced, depolarizes and the auditory nerve generates an action potential (Hudspeth, 2005). The auditory nerve codes timing and frequency specific information that is sent via electrical signals from the cochlea to the cochlear nucleus in the brainstem (Howarth & Shone, 2006). The cochlear nucleus preserves the neural coding of the auditory information performed initially by the auditory nerve. From the cochlea to the auditory afferent nerve fibers, electrical energy is sent through the brainstem up to the auditory cortex which is interpreted into recognizable sounds (Howarth & Shone, 2006).

In an individual with normal hearing, a clear and audible signal is delivered to the higher structures in the auditory system for processing. However, in an individual with hearing loss, degraded signals produced by the hearing loss or environmental distortions (i.e., reverberation) are delivered to the higher levels of the auditory system. More specifically, reverberation produces an acoustic filtering effect that alters the original signal by smearing, eliminating, or distorting the frequency and spectral characteristics of the signal (Nabelek & Robinette, 1978; Nabelek et al., 1989). Individuals with hearing loss have (a) elevated hearing thresholds and (b) degraded temporal resolution (reduced spectral and temporal cues) (Bronkhorst & Plomp, 1992). Additionally, hearing loss can be caused by a variety of risk factors, with the most common causes including noise induced hearing loss and age-related hearing loss (The Hearing Loss Association of

America, 2016). Additional etiologic factors in the development of age-related hearing loss include: infections; ototoxic medications; heart disease, and diabetes (Taylor & Mueller, 2011, Chapter 3). However, it can prove challenging to isolate the exact cause of hearing loss in any given individual. The primary cause of hearing loss in the aging population has been referred to in the literature as "Presbycusis" (Helfer & Freyman, 2008). Characteristically, presbycusis involves bilateral high frequency sensorineural hearing loss that occurs gradually over time because of aging. This age-related process can be attributed to a genetic predisposition to the deterioration of hair cells in the cochlea. Consequently, the reduction in hearing sensitivity can be attributed to the weakening of the active processes of the ear's outer hair cells (OHCs) (Helfer & Freyman, 2008; Hudspeth, 2005).

When the OHCs are damaged two significant things occur:

- There is a mild-to-moderate loss of hearing. (which can be as great as 50 to 60 dB from OHC damage)
- The cochlea losses its ability for sharp frequency tuning. (Taylor & Mueller, 2011, pp. 64)

Similarly, individuals with hearing loss have difficulty focusing on one speaker and filtering out any unwanted acoustic information (i.e., sources of masking) (Gatehouse & Akeroyd, 2006). There are several reasons why individuals with hearing loss have difficulty understanding speech in noise. First, the background noise often exceeds the level of the target signal. When the signal-to-noise ratio approaches 0 dB, even people with normal hearing struggle to understand speech in an adverse listening environment (Taylor & Mueller, 2011, Chapter 3). Second, individuals with high frequency hearing loss miss soft sounds of speech that are critical to understanding speech. Finally, the central auditory pathways are susceptible to age-related changes, thus, the central auditory systems' ability to relay important acoustic information deteriorates (Weinstein, 2000, Chapter 4). These are some of the many reasons a damaged auditory system impairs a listener's ability to distinguish between a target speaker and unwanted acoustic information. To conclude, these ages related changes in the auditory system can affect any part of the auditory signal ranging from accurately *encoding the signal* to *processing of the signal*. The next section of this literature review is concerned with masking, and how the presence of one sound impacts the audibility of another sound.

Masking

Masking is a process that refers to the ability of one sound to block out or reduce the audibility of another sound (Gelfand, 2010, Chapter 10). Bregman (1990) indicated that in a typical masking experiment there are two sounds presented simultaneously, a target and a masker. The listener is instructed to listen for the target signal while ignoring the masker signal. The intensity of the masker is increased until the target can no longer be detected. The effectiveness of the masking can be influenced by many factors including: the intensity and the frequency; spectrum of the masker; spatial location of the target and masker signals; temporal structure of the signals, and amplitude modulation of the signal (Eggermont, 2013; Jones & Litovsky, 2011; Kidd, Mason, Deliwala, Woods, & Colburn, 1994). The subsequent sections will address the different classifications of masking and review the effects that the two different categories of masking have on the peripheral and central auditory system.

Energetic Masking

The perception of a target signal can be energetically masked by a competing sound source when they simultaneously occur within the same critical frequency band. This phenomenon occurs because of an overlap of their representations that originate in the peripheral auditory system (Arbogast et al., 2005; Breitsprecher, 2011; Brungart, Simpson, Ericson, & Scott, 2001; Fletcher, 1940; Ihlefeld & Shinn-Cunningham, 2008). Fletcher (1940) reported on which frequency elements contribute to the effectiveness of the masker signal. Fletcher indicated that the only components of the noise that have a significant masking effect on the target signal are the frequencies contained within the critical frequency band. Consequently, when a target signal and a competing sound source are presented simultaneously, the signals compete for representation at the level of the peripheral auditory system. As a result, the auditory system cannot filter the two signals.

One type of energetic masking often encountered is speech masking. Everyday listening environments are comprised of a target signal which frequently co-occurs in the presence of various sources of masking. Researchers have demonstrated that speech masking impacts intelligibility more than the presence of stationary noise (Breitsprecher, 2011; A. Bronkhorst, 2000; Hawley, Litovsky, & Culling, 2004; Jones & Litovsky, 2011). This effect occurs because the energetic component of speech is more complicated than stationary noise as speech fluctuates in frequency and amplitude over time. This in turn, results in a variation of energetic masking (Breitsprecher, 2011).

Informational Masking

In contrast to energetic masking, informational masking refers to a reduction in performance in higher processing levels of the auditory system even though the peripheral auditory system provides enough information to accurately encode the target and masker (Durlach et al., 2003). Brungart and colleagues (2005) suggested that informational masking occurs when the target signal and masker are both audible, but the listener is unable to separate the acoustic cues of the target signal from the acoustic cues of the masker. Durlach et al. (2003) and Watson (2005) reported that informational masking varies depending on factors such as (a) similarity of talkers and maskers and (b) trial-to-trial uncertainty. These factors in isolation are not enough to produce substantial informational masking. However, when combined, the listener's thresholds can be significantly elevated (Durlach et al., 2003). Informational masking is theorized to interfere with the ability for sound discrimination, ability to detect the signal, and the ability to focus attention on the talker of interest (Durlach et al., 2003; Ihlefeld & Shinn-Cunningham, 2008; Kidd, Mason, Deliwala, Woods, & Colburn, 1994).

Spatial Release from Masking in Speech Perception

Speech intelligibility in a cocktail party environment is dependent on numerous acoustical factors. The more relevant aspects include fundamental frequency (male vs. female), spectral fluctuations of multiple speech maskers (frequency and amplitude fluctuations), context, differences in the levels of the target and masker signal, and differences in the location and timing of the target and maskers in the environment (Alain, 2007; Bronkhorst & Plomp, 1992; Brungart et al., 2001). Sources of masking often come from different locations and distances in a complex listening environment.

Typically, SRM is assessed using two different paradigms: colocated and spatially separated. Colocated is when the target signal and masker signals are at the same location. In the colocated condition (i.e., masked) there is no difference between the signal-to-noise ratio (SNR) at each ear. When the masker and target signal are colocated, it can prove difficult for even normal hearing listeners to separate the acoustic cues of the target speech from the acoustic cues of the masking source (Bregman, 1990). In the spatially-separated condition (i.e., unmasked), the ear closest to the target signal obtains an enhanced SNR which is attributed to the head shadow effect. Furthermore, when the target and maskers are at different locations, there is an improvement in a listener's ability to discern between the target and masker. Several researchers have indicated that a listener can take advantage of acoustic cues arising from spatial separation (Brungart et al., 2001; Freyman, Balakrishnan, & Helfer, 2001; Freyman, Helfer, McCall, & Clifton, 1999; Hawley, Litovsky, & Culling, 2004). This benefit of spatial separation between the target speech signal and masker signal is known as spatial release from masking (SRM) (Bronkhorst, 2000; Brungart et al., 2001; Freyman et al., 1999, 2001; Hawley et al., 2004; Jones & Litovsky, 2011).

Hawley, Litsky, and Culling (2004) reported on the advantages of symmetrical maskers (i.e., multiple voice interferers) when compared to measurements with fewer maskers (i.e., single masker) or with other types of interference (i.e., spectrum shaped noise, speech-spectrum noise, and time reversed sentences). Speech Reception Thresholds (SRTs) were measured for 16 normal hearing participants with binaural presentation and 16 normal hearing participants with monaural presentation (n = 32 participants). Performance of the listeners from the monaural conditions was paired with listeners from the binaural condition for data analyses. In the Hawley et al. study (2004), the total advantage of separation for each listener in each condition was determined by subtracting the Speech Reception Threshold (SRT) from a given separated condition from that for the equivalent unseparated condition. For example, the change in SRT from the binaural unseparated speech condition (SRT: 14 dB) to the binaural separated speech condition (SRT: 8 dB), would produce a 6-dB benefit obtained from spatial separation of the target signal and the two speech masker signals. Then, the monaural advantage of separation for each listener was calculated using the same process as previously described. Finally, the binaural advantage was calculated by subtracting the monaural advantage from the total advantage of separation. The researchers indicated that for a single interferer, there was a binaural advantage of 2-4 dB for all interferer types. Consequently, for two to three speech maskers, the advantage was 6-7 dB for speech and time-reversed speech. The researchers concluded that measurements with a single masker could underestimate the benefit of spatial release from masking (Hawley et al., 2004).

The separation of sound sources is investigated by measuring the threshold for the target signal in the presence of different sources of masking, specifically when the maskers are colocated (i.e., target and masker presented at 0° azimuth) and secondly when the maskers are spatially separated from the target signal (i.e., target signal presented at 0° and maskers symmetrically separated by ±90°). Spatial release from masking is produced primarily because of two factors: (a) spatial separation of target and masker provides an increase in the target-to-masker ratio at the ear closest to the original signal, and (b) the advantages of binaural processing (Gelfand, 2010). Furthermore, the benefit of spatial separation is affected by several factors. For example, the type of masker present has a different effect on the benefit of release from masking. The two types of masking that must be considered in an adverse listening condition are energetic and informational masking. Several studies have demonstrated that the spatial separation of a target signal and masker signal can reduce both energetic and informational masking (Arbogast, Mason, & Kidd, 2002; Arbogast et al., 2005; Brungart et al., 2005, 2001, Freyman et al., 2001, 1999; Hawley et al., 2004; Kidd et al., 1994; Kidd, Mason, Rohtla, & Deliwala, 1998). Additionally, the spatial separation of a target and masker produces target-to-masker ratios (TMR) across ears. The TMR changes with spatial location and is typically more significant at higher frequencies. When the TMR is different between the two ears, the auditory system can utilize the TMR at the ear closest to the original source to improve speech recognition.

Effects of Hearing Loss on Spatial Release from Masking

Subsequently, there has been an extensive amount of literature which has evaluated the effects of age-related hearing loss on spatial release from masking (SRM) in the presence of sources of speech masking (Gallun et al., 2013; Glyde et al., 2013; Marrone et al., 2008; Srinivasan, Jakien, & Gallun, 2016). Additionally, understanding the effects of aging on spatial release from masking independent of hearing is one of the leading obstacles researchers have faced in this area (Gallun, Diedesch, Kampel, & Jakien, 2013; Marrone, Mason, & Kidd, 2008; Srinivasan, Jakien, & Gallun, 2016). Marrone et al. (2008) conducted a study to investigate the effects of hearing loss and age on SRM and concluded that age of the subjects, independent of their hearing status, was not a significant predictor of spatially separated thresholds. Of an important note, Marrone et al., (2008) only investigated large spatial separations (±90°). Srinivasan and colleagues (2016) conducted a study using headphones and non-individualized Head Related Transfer Functions (HTRFs) and investigated the effects of smaller spatial separations between a target stimulus and different types of maskers. The purpose of their study was to predict the individualized contributions of age and hearing loss to SRM. The findings of the Srinivasan study revealed that the subjects' age was the main factor predicating SRM at smaller spatial separations; whereas the subjects' hearing loss was the primary contributing factor for predicting SRM at larger spatial separations (Srinivasan et al., 2016).

Reductions in speech intelligibility can also be attributed to environmental distortions such as background noise and reverberation. Speech intelligibility in distracting noise is dependent on the level of the target speech as well as the level of the background noise (i.e., signal-to-noise ratio) (Eggermont, 2013). The spatial separation of the target and the masker provides acoustic cues such as interaural timing (ITDs) and interaural level (ILDs) that provide different target-to-masker ratios for both ears (Freyman et al., 1999). When masking is primarily energetic in nature, improved speech understanding in noise for normal hearing listeners may result from attending to the ear with a higher signal-to-noise ratio (SNR) (Bronkhorst & Plomp, 1992; Freyman et al., 1999). For reference, the illustration below depicts the better-ear advantage and the benefit received from spatial release from masking. As seen in this illustration, the right ear obtains an enhanced SNR which can be attributed to the head shadow effect.

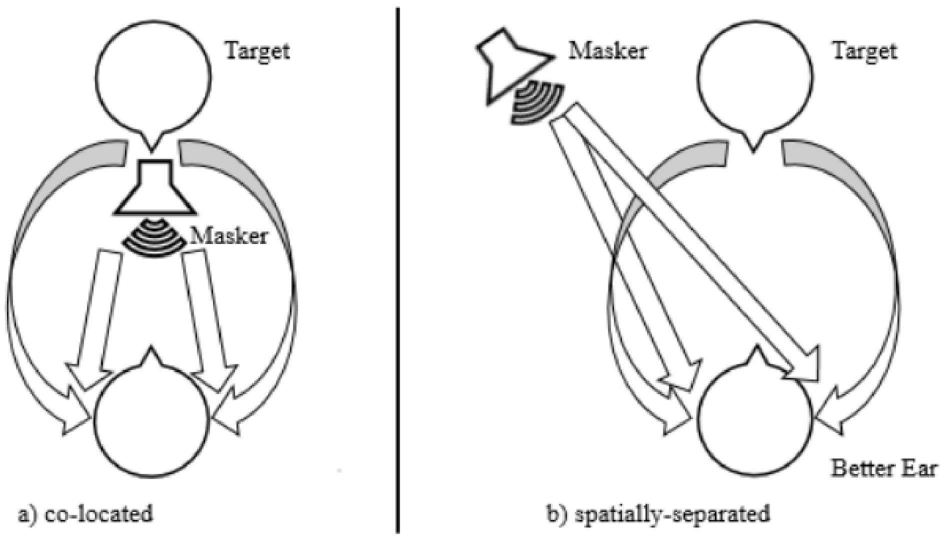

Figure 1. Illustration of the better-ear advantage. In the colocated condition (i.e., masked), there is no difference between the signal-to-noise ratio (SNR) at each ear. In the spatially-separated condition (i.e., unmasked), the right ear obtains an enhanced SNR which is attributed to the head shadow effect. *Source:* Based on Carlile (2014), *Speech intelligibility in noisy environments, Vol 42*.

Consequently, hearing loss results in a reduced ability to use these ITDs and ILDs and, in turn, diminishes the advantage of spatial separation for speech recognition in adverse listening conditions (Bronkhorst & Plomp, 1988). To understand speech equally as well as the normal hearing listener, a listener with hearing loss needs a 4.2-10 dB better signal-to-noise ratio for equal intelligibility (Bronkhorst & Plomp, 1992). Additionally, speech has the potential to produce informational masking. Furthermore, researchers have indicated that the benefit obtained from a spatial release from masking is less for individuals with hearing loss in comparison to individuals with normal hearing (Arbogast et al., 2002, 2005; Bronkhorst & Plomp, 1992; Gallun et al., 2013).

Arbogast et al. (2005) utilized the coordinate response measure (CRM) corpus of sentences and investigated the effect of informational and energetic masking on the spatial release from masking in normal hearing and hearing-impaired listeners. The hearing-impaired (HI) group and the age-matched normal hearing (NH) listener group ranged in age from 19 to 79 years of age. The study was conducted in a sound-treated booth, and the target stimuli were always presented directly in front of the listener (0° azimuth) and the speaker configuration of the masker stimuli varied (0° and 90° azimuth). The three different types of maskers that were utilized in this experiment included: different-band sentence (DBS) (primarily informational, different-band noise (DBN) (primarily informational), and same-band noise (SBN) (primarily energetic). The researchers concluded that the hearing-impaired (HI) group obtained similar spatial release from masking (1.5-2.0 dB) as the normal hearing (NH) group from energetic masking. However, the average SRM obtained from the informational masker was statistically more significant for the NH group (i.e., 15.3 dB) in comparison to the HI group (i.e., 9.5 dB). To summarize, larger SRM effects were obtained for the informational masker in comparison to the energetic masker condition, which indicates that the benefit obtained from spatially separating the target signal from the masker signal can be utilized by NH and HI listeners to decrease the detrimental effects of informational masking. Based on these findings, Arbogast and colleagues suggested that if these higher-level processes of the auditory cortex are intact, spatially separating the target signal from the masker signal can provide hearing-impaired listeners with improved speech intelligibility in a cocktail party environment. Nonetheless, this study did not examine the effects of aging on these higher-level processes (Arbogast et al., 2005). Therefore, additional research will be reviewed to determine if these higher-level processes in listeners with age-related hearing loss are equally as effective in processing informational masking as NH listeners.

Effects of Aging on Spatial Release from Masking

Hearing loss is one of the most common health problems, and the degree of impairment and prevalence increases with age (Howarth & Shone, 2006). One of the biggest challenges many older individuals encounter is difficulty understanding conversations in the presence of multiple simultaneous talkers. The impact of age on spatial release from masking is less understood than the effects of hearing loss on SRM. Researchers have indicated that there is a need to understand both the nature of age-related changes in the auditory system in older individuals with hearing loss and how these age-related changes influence their ability to understanding speech in adverse listening conditions (Gallun et al., 2013; Glyde, Cameron, Dillon, Hickson, & Seeto, 2013; Helfer & Freyman, 2008; Srinivasan et al., 2016). These teams of researchers have suggested that higher level processing abilities play an important role when trying to understand a target talker in the presence of competing speech (Gallun et al., 2013; Glyde et al., 2013; Helfer & Freyman, 2008; Srinivasan et al., 2016).

There are several processes that occur in the auditory system that can be impacted by age-related changes. Age-related changes in the auditory system can affect any part from accurately encoding sounds to the processing of the signal. These age-related changes are extensive and could include a reduction in the following cognitive abilities: working memory, executive function, ability to ignore irrelevant information, processing speech, selective attention, and lexical knowledge (Tun, Williams, Small, & Hafter, 2012). Helfer & Freyman (2008) investigated the influences, if any, of energetic masking, informational masking, as well as higher level cognitive processes on speech perception in an adverse listening condition. Helfer & Freyman (2008) used a four-type speech-on-speech masker design to examine the effects of hearing loss and aging on sentence recognition. This study included a younger group of listeners with normal hearing (11 women, 1 male) and an older group of listeners (9 women, 3 males) (average age of 72 years). Most listeners in the hearing-impaired group had essentially a mild to moderate high-frequency sensorineural hearing loss. For the purposes of this literature review, this discussion will focus on the older hearing-impaired group. Helfer and colleagues (2008) findings indicated that the older adult group had significantly poorer sentence recognition in the presence of all four types of maskers in comparison to the younger group. The most considerable difference between the younger and older group was observed for the Male Target Talker (MTT) masker when compared to other maskers. Glyde et al. (2013) conducted a subsequent study and concluded the effect of age on spatial-processing ability is less statistically significant than the effect of hearing loss. The researchers suggested that even a mild hearing loss diminishes one's ability to benefit from a spatial separation of target talker and masker. Similarly, Glyde et al. (2013) reported that further research is needed to understand the effects of aging on SRM.

In previous studies, researchers were unable to distinguish the effects of age from hearing loss on spatial-processing abilities due to issues of sample size (Abrogast et al., 2002, 2005; Glyde et al., 2013; Marrone et al., 2008). Therefore, a larger sample size is essential in order to detect a small effect size such as age. Researchers also indicated that it is important to examine further the main effects of hearing loss and aging on spatial release from masking and develop testing methods to reduce the influence of hearing loss on the outcome (Gallun et al., 2013). Gallun and colleagues (2013) conducted a study to investigate the effects of age on SRM. Four spatial configurations were utilized: 1) colocated (all three sentences presented from 0° azimuth), 2) 15° separation (target at 0°, maskers at ±15°), 3) 30° separation (target at 0°, maskers at ±30°), and 4) 45° separation (target at 0°, maskers at ±45°). Additionally, these researchers used a four talker-gender combination condition: 1) male/male (male target, male masker), 2) male/female (male target, female masker), 3) female/female (female target, female masker), and 4) female, male (female target, male masker). In experiment 2, the mean TMR thresholds for the colocated condition were 2.0 dB for same gender speech maskers and 6.9 dB for different gender maskers. Additionally, when the target and masker were spatially separated, there was a greater release from masking for the different genders (i.e., male target talker, female masker) (TMR -10.8 dB) when compared to the same gender (i.e., male target talker, male masker) (TMR -7.8 dB). Gallun et al. (2013) indicated that age for the spatially separated condition (r^2 = 0.55) and colocated condition (r^2 = 0.32) does have a statistically significant effect on the SRM independent of hearing impairment. These findings provided evidence that the impact of age on spatial release from masking is significant and is independent of hearing loss.

Srinivasan et al. (2016) conducted a subsequent study to examine the effects of smaller spatial separations of target and masker signals to further separate the effects of age on SRM from the effects of hearing loss. These researchers wanted to determine if older hearing-impaired individuals needed a more substantial spatial separation of the target and masker signals in order to obtain the increased benefit of SRM. In this study, SRM was investigated at eight spatial different configurations; for condition 1 the masker was colocated with the target at 0°, whereas for conditions 2-8, the masker was symmetrically separated from the target by 2°, 4°, 6°, 8°, 10°, 15°, or 30°, respectively.

Srinivasan and colleagues (2016) found that younger normal hearing listeners could obtain benefit from spatial separations between the target talker and symmetrically separated maskers as small as (2°-4°). Additionally, age as opposed to hearing loss, was the significant predictor at spatial separations of 4° and 6°. In contrast, older hearing-impaired individuals obtained minimal benefit from even the largest spatial separation tested of 30°. Consequently, older normal hearing listeners required an average spatial separation of at least 6° to demonstrate an improvement in TMR (target-masker-ratio). To summarize, the benefit obtained from SRM varied significantly for the three listener groups.

3 Primary Cues for Localization

In this section of the document, we explore several aspects of the perceived direction of sound (i.e., localization) and the differentiation between two sound sources from different locations (i.e., discrimination). Specifically, we will examine how sounds presented in a free field (i.e., speakers) and a sound presented under headphones can result in a different perception of the location of a sound. Sounds originating directly in front of the listener have an azimuth of (0°), where sounds originating directly behind the listener have an azimuth of (180°). Other angles of azimuth can also use positive (right) and negative (left) signs which indicate the position of the sound relative to the center of the listener's head. For example, a +45° and a -45° spatial separation indicate the listener is facing 0° azimuth with the speaker positioned 45° off center to the right and left, respectively. An illustration adapted from Gelfand (2010) is provided below to demonstrate the various ways of expressing angles of azimuth horizontally relative to the listener's head.

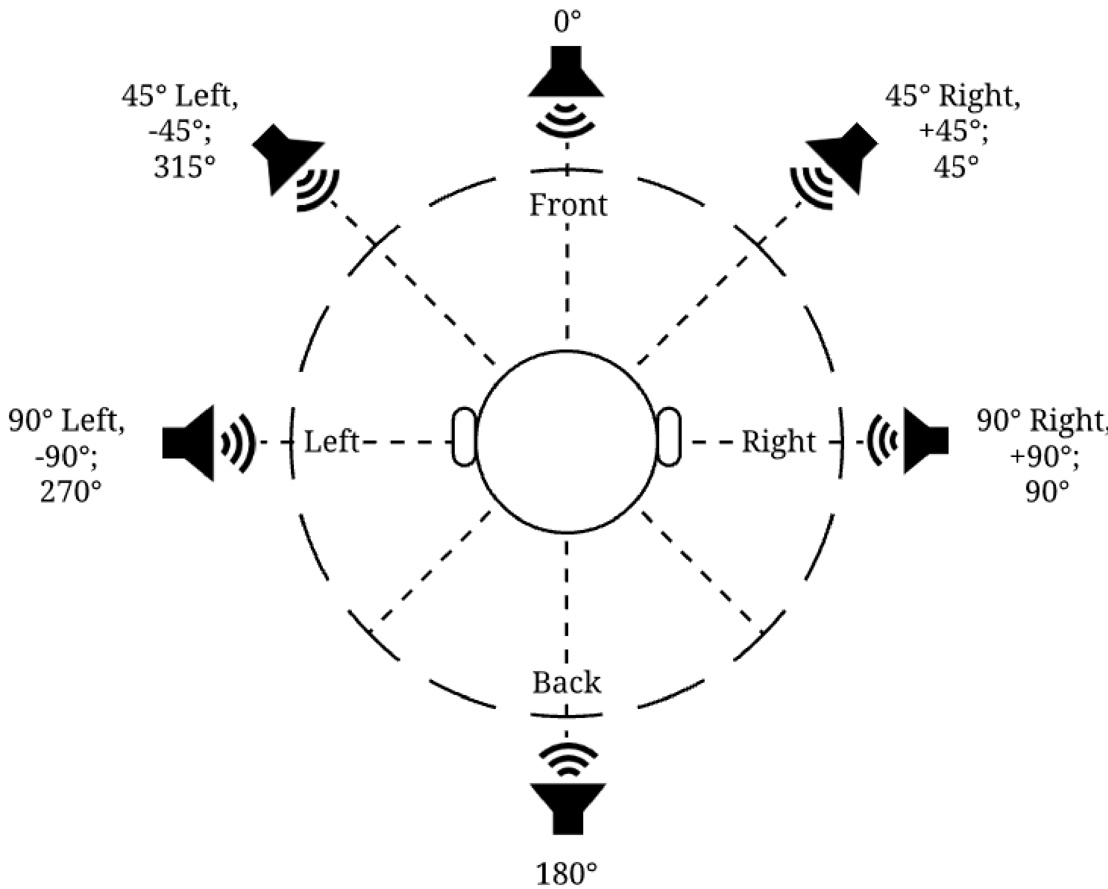

Figure 2. Illustration of the angels of azimuth. Based on "Hearing: An introduction to psychological and physiological acoustics," by S.A. Gelfand, 2010, *Binaural and Spatial Hearing, 5th edition*, p. 235. Copyright 2010 by Informa Healthcare.

Localization

The physical properties of the pinna, shoulders, and head produce time and intensity differences between the ears and a filtered response that provides the auditory system with the necessary information to localize sounds in space (Gallun, lecture, February 17, 2017). The human auditory system can localize sounds in three dimensions which are as follows: distance, azimuth (i.e., horizontal plane), and elevation (i.e., medial plane). For the purposes of this document, the primary focus will be on the horizontal plane. The auditory system depends upon three main acoustic cues known as interaural timing differences (ITDs), interaural level differences (ILDs), and spectral cues in determining the location of sound. The localization cues can also be categorized into monaural and binaural cues (Emanuel, Maroonroge, & Letowski, 2009).

Monaural spectral cues are produced by the anatomical properties of the outer ear. The spectrum of a sound that arrives at the tympanic membrane (i.e., eardrum) is the product of the pinna filtering and the spectrum of the sound source (Wightman & Kistler, 1997). The auditory system extracts these monaural cues by deconvolving the sound transduced at the eardrum, the separate contributions of the sound source, and the pinna filtering effects (Wightman & Kistler, 1997). Head Related Transfer function (HRTF) is a response that characterizes the difference between the sound that arrives at the listener and the sound that enters the ear canal. HRTF vary as a function of the position of the source in reference to the listener and with the frequency of the sound (Emanuel et al., 2009). The sound field to head-related transfer function also depends upon the location at which sound is oriented toward the listener. These HRTFs can be used to study the effects of how the amplitude spectrum of a sound is modified as it travels from a sound source in a free field to the listener's ear canal. For example, Shaw (1974) investigated transfer functions from the sound field to the eardrum. Shaw (1974) recorded measurements of the sound pressure level arriving at the right tympanic membrane when sound was presented at different azimuths. Shaw (1974) reported a consistent increase in sound pressure level in the 1500-7000 Hz frequency region when presenting the sound from a loud speaker from -45° azimuth (i.e., contralateral ear) to +45° azimuth (i.e., ipsilateral ear). Wanrooij and Van Opstal (2004) studied the degree to which unilateral deaf individuals depend on intensity and spectral cues to localize sounds. These researchers reported that a majority of monaural listeners in this study showed evidence of using spectral cues to localize sound-source azimuth. Surprisingly, all monaural listeners depended on the head shadow effect to localize sounds in the azimuthal plane. However, it is clear there is a significant degradation of azimuth localization performance of monaural listeners when compared with binaural listeners (Wanrooij & Van Opstal, 2004). Moreover, to determine the horizontal position of a sound, the auditory system relies mainly on binaural difference cues (Sudirga, 2014).

The two primary binaural cues are interaural timing differences (ITDs) and interaural level differences (ILDs). The binaural difference cues such as ITDs and ILDs are based on the comparison of the sound signal that arrives at the two ears (Emanuel et al., 2009; Stecker & Gallun, 2012; Sudirga, 2014). ITD is the difference in arrival time of the sound wave at the ipsilateral ear relative to the contralateral ear. ILD is the difference in sound pressure level arriving at the two ears. The energy of the sound arriving at the listener is affected by the outer ear, head, and the human body. A physical interaction occurs between the sound and the human body that attenuates and reflects some sounds away from the listener while other sounds are reflected toward the ear canal and enhanced (Emanuel et al., 2009). Consequently, a listener's ability to localize a sound source is dependent upon time differences between the ears at lower frequencies and level differences between the ears at high frequencies (Gelfand, 2010).

The head shadow effect occurs in the sound field where the presence of the head creates a region of reduced sound pressure on the contralateral (far) side from where the direct sound source is coming from (Emanuel et al., 2009). This head shadow effect is significant for frequencies > 1500 Hz because the wavelength of higher frequencies is smaller in comparison to the diameter of the head and they are attenuated in the arrival to the contralateral ear. This head shadow effect produces interaural level differences or interaural timing differences between the two ears. For lower frequencies, the wavelength is longer than the diameter of the head and the head does not impede the sound wave. ITDs are used to localize sounds at lower frequencies and ILDs are used to localize sounds at higher frequencies.

Sound Localization in Free Field

Early studies of localization were conducted in free-field environments to minimize any effects of reverberation by presenting sounds in an environment absent of any reflective boundaries (Stecker & Gallun, 2012). Stevens and Newman (1936) were among some of the first researchers to investigate the effects of localization in a free field. The study was conducted entirely in open air (i.e., free field) outside, to minimize the effects of reverberation. Stevens and Newman (1936) instructed participants to listen to a sound produced by a loudspeaker and state the perceived location of the sound. Researchers indicated that there is a significant relationship between accuracy of localization as a function of frequency (Stevens & Newman, 1936). The findings of this study revealed that frequencies below 1000 Hz and above 4000 Hz were localized with greater accuracy in comparison to sounds in the frequency range between 2000-4000 Hz. Additionally, low-frequency tones were localized based on phase-differences at the two ears and high-frequency tones (i.e., 3000 Hz) were localized based on intensity differences (Stevens & Newman, 1936).

Subsequent studies have utilized anechoic chambers to study localization. Sandel, Teas, Feddersen, and Feffress (1955) examined the ability of listeners to localize sounds in an anechoic room with speakers positioned at 0 degrees and 40 degrees. The results demonstrated that frequencies below 1500 Hz were accurately localized using ITD cues, these findings were in agreement with the results reported in Stevens and Newman (1936). Bronkhorst & Plomp (1988) conducted a study to investigate the effects of ITDs and ILDs on speech intelligibility in noise. When maskers were spatially separated and ILDs and ITDs were introduced, an increase in SRT occurs (Bronkhorst & Plomp, 1988). This increase in SRTs ranged from 1-8 dB for normal hearing listeners and 1-6 dB for hearing-impaired listeners.

Sound Localization under Headphone presentation

Sounds that are presented from earphones are perceived to be inside the head and can be lateralized along the horizontal plane (Gelfand, 2010). In experimental studies utilizing headphones, these localization cues could be altered to investigate these cues individually or to analyze the interaction of a combination of different cues (Yost, 1974; Yost et al., 1974; Yost & Dye, 1988. Also, researchers utilized the virtual auditory space techniques to study directional hearing (Wightman and Kistler, 1992; Macpherson & Middlebrooks, 2002).

Head-related transfer functions (HRTFs) are utilized for spatial representation of the direct path of the original signal and the reflections. A binaural room impulse response model (BRIR) can then be used to describe the acoustic properties of a room in terms of sound propagation and reflections (Zahorik, 2009). These acoustic properties of the room are characterized by the room size, absorptive properties of the floor, ceiling, and walls, the position and orientation of the listener in reference to the speaker, and the location of other sound reflective objects in the room (Srinivasan, Stansell, & Gallun, 2017; Zahorik, 2009). These HRTFs can be used to recreate the spatial representation of the direct path and early reflections of the sound source over headphones. Several studies on localization under headphones by varying interaural timing and level differences

Additionally, these acoustic cues that we use to localize sounds can be degraded by reverberation. The localization cues of ITDs and ILDs are often degraded by acoustical reflections from boundaries in an enclosed room (Boothroyd, 2004; Bronkhorst & Plomp, 1988).

Sound Localization in Reverberation

The main acoustic cues used for sound localization in a free-field environment and under headphones have been reported in the previous section. Sound localization in a reverberant environment is based off the same set acoustic cues. However, reverberation may degrade these cues. In this section of the literature review, we will discuss the degradation of these acoustic cues for sound localization and how the auditory system processes allow us to localize in these reverberant environments. There has been an extensive amount of literature that has studied how we localize sounds accurately in a free field and under headphones. When a single direct sound arrives at a listener's ear, sound localization cues provide information about a sound's origin. However, when the sound travels from a source to a listener in a reverberant environment, the acoustic characteristics of sound are modified and reflected by the surfaces and boundaries in these environments. The sound waves are reflected off the boundaries in many different directions and arrive at the listener's ears later than the direct sound. The content of reverberant sound in relation to the content of the original sound depends on the room size, reflective and absorptive properties of its boundaries, and the directionality of the source (Boothroyd, 2004; Shinn-Cunningham, 2013). Although the direct sound is followed by multiple reflections that can degrade the spatial information provided by the direct sound, the auditory system utilizes several neural mechanisms to process the direction of the earlier-arriving signal (Gelfand, 2010).

Interestingly, there is a behavioral phenomenon that refers to a listener's ability to judge the direction of a sound source and localize it accurately in these reverberant conditions and is referred to in the literature as the precedence effect (Shinn-Cunningham, 2013). This effect will be described in the next section of this literature review.

Precedence Effect

In reverberant listening environments, listeners can extract important localization cues about a sound despite receiving both direct and reflected sounds that potentially can degrade the spatial information provided by the direct sound. In these reverberant conditions, the listener depends more on the spatial information at the onset of direct sound in comparison to the spatial information of the latter arriving reflected sounds. In a simple precedence experiment, two sounds are presented. One speaker presents an initial brief click, while the other speaker presents an identical click after a time delay. The listeners are asked to judge whether they perceive one signal or two. Although the precedence effect can be discussed as a single psychophysical phenomenon, there are several different mechanisms that contribute to the perceived dominance of early spatial information over the later arriving information.

A key function of the peripheral auditory system is its ability to receive auditory information that is presented to both ears and combine the sounds into a fused auditory perception. Research has demonstrated that when the time delay between the lead signal and the lag signal is less than 5ms, the lead and lag signals are combined into a fused auditory perception. This convergence contributes to the perceived direction of the fused signal and provides the peripheral auditory system important information on differences in these neural signals, such as differences in time of arrival and intensity differences. This mechanism is known as summing localization. For relatively short delays (i.e., 1-5 msec) the two signals remain perceptually fused. However, as the time delay between the two signals increases, the lagging signal becomes audible as a separate auditory event. Researchers have demonstrated in an ideal listening environment (i.e., anechoic chamber), as the click is delayed beyond 5 msec, listeners can differentiate the lead and lag signal as a separate event.

Precedence effect is thought to be involved in resolving the competition for perception and localization between the direct sound and a reflection (Gelfand, 2010). Additionally, in enclosed spaces, listeners also receive speech via reverberation. For speech, the precedence effect has a longer time course for the ongoing signal because speech contains multiple onsets due to local energy (Zahorik, 2009). Researchers have suggested that the precedence effect produces a perceived separation of a target signal and a masker signal in reverberant spaces (Freyman et al., 2001, 1999; Hirsh, 1950). Freyman et al. (1999) wanted to determine if the precedence effect could be used to produce a perceived separation of a target source and two different sources of masking which could potentially improve speech recognition in a reverberant space. The researchers conducted two different experimental models to measure the degree to which the benefit of SRM is degraded by energetic and informational masking (Freyman et al., 1999). The study was conducted in an anechoic chamber and the signals were presented from two speakers, one speaker positioned directly in front of the listener (i.e., 0 degrees) and the other speaker positioned to the right of the listener (i.e., 60 degrees). In the experiment, six different speaker configurations were used for the presentation of the target stimuli and masker stimuli. In conditions 1-4, the target stimuli were delivered to the listeners as a single source from the front speaker while the speaker configuration of the masker consisted of a single source and a 4-ms delayed reflection.

The precedence effect conditions involved presenting speech-spectrum noise (i.e., energetic masking) or female masking speech (i.e., informational) from two speakers with a 4-ms delay in the presentation of the signal from one speaker in comparison to the second speaker. Due to the precedence effect, the listener perceived the direction of the masker either colocated or spatially separated from the target signal, which was determined by the leading signal. Similarly, the researchers developed two additional experimental conditions in which the target and masker stimuli were presented from both speakers with added reflections to simulate a more accurate representation of an everyday listening environment. An interesting finding in this study is that the illusion of spatial separation of two talkers created by the precedence effect provides normal hearing listeners with a release from masking that improves their speech intelligibility in reverberant spaces (Freyman et al., 1999). However, the findings of this study demonstrated that the benefit provided by SRM is much smaller for energetic masking in comparison to informational masking. Additionally, reverberation reduced the benefit of spatially separating the target speech from energetic masking from 8 dB to 1 dB, while for informational masking, the benefit of SRM was reduced from 14 dB to 9 dB (Freyman et al., 1999).

Reverberation

A complex listening environment is comprised of the following components: direct source (i.e., target signal), distance, reverberations, and noise (Boothroyd, 2004). In this section, we will focus on the concept of reverberation. Reverberation is the persistence of sound in an enclosed space after the direct sound has stopped and it is caused by a collection of acoustic reflections from the ceiling, floor, and other surfaces in the environment (Boothroyd, 2004). Boothroyd (2004) indicated that room acoustics have a significant impact on the transmission of speech from a speaker to the listener's ears. There are two main ways to control the sound characteristics of a room, which includes passive and active acoustics. Passive acoustics are the materials and boundaries in a room such as carpet, walls, and acoustic tiles. These materials have absorptive properties and reflective properties. The absorption coefficient represents the ability of the materials or boundaries to absorb the sound energy, which results in the reduction in reverberation time. Conversely, the reflective coefficient represents the pattern in which sound energy is reflected and redirected. These absorptive and reflective properties contribute to the reverberation time in a room. For example, in large rooms with many reflective materials and boundaries, there is an increase in the reverberation time. In contrast, in smaller rooms with more absorptive materials, there is a decrease in the reverberation time. Additionally, there is a mathematical relationship between the absorptive and reflective properties of the materials and boundaries in a room, the size of the room, and the reverberation time. The reduction of reverberation energy is measured in seconds. The calculation of reverberation energy is T_{60}, which refers to the duration of time needed for reverberation energy to decrease by 60 dB at the offset of the direct sound source (Boothroyd, 2004).

Binaural Room Impulse Response

Active acoustics can consist of speakers, amplifiers, and electronic equipment. Virtual acoustic techniques can be utilized to simulate a reverberant environment (Zahorik, 2009). Zahorik (2009) reported that one of the most common and useful methods for representing the acoustics of a room is by a transfer function known as Binaural Room Impulse Response (BRIR). In this section, we introduce the concept of BRIRs in addition to the three components of the BRIR, which include: direct sound (DS), early reflections (ER), and late reflections (LR).

The BRIR contains information about the size and materials of the room, the physical characteristics of the listener (i.e., head, torso, and shoulders) and the position of the listener in relation to the sound source. This transfer function allows researchers to take any recording, convolve with the BRIR, and present it over headphones to determine if the BRIR is a physical and perceptual approximation of the BRIR measured in the room the recording took place. Furthermore, researchers can manipulate the reverberation over headphones by changing the dimensions, the materials, and position of the sound source. Once a binaural room impulse response has been measured, the effects of simulated reverberation on speech can be directly studied (Zahorik, 2009).

Early Reflections

In a reverberant environment, listeners receive speech through three main components: direct sound (DS), early reflections (ER), and late reflections (LR) (Srinivasan, 2017). Researchers have indicated that reverberation can have a positive or an adverse effect on speech intelligibility (Bradley, Sato, & Picard, 2003; Breitsprecher, 2011; Lavandier & Culling, 2007, 2008; Marrone et al., 2008; Nábělek & Robinette, 1978; Srinivasan et al., 2017; Warzybok, Rennies, Brand, Doclo, & Kollmeier, 2013). As sound travels from the source to the listener, the acoustical energy is spread throughout the enclosed space. When the listener is close to the sound source, the level of direct source is greater than the reverberant sound. When a listener is far from the source, the reverberant sound energy masks the direct sound (Boothroyd, 2004).

The energy from early reflections arrives at the listener's ear shortly (<50 ms) after the original speech sound is generated (Bradley & Sato, 2003). Nabelek & Robinette (1978) investigated the effect of a single early reflection on word identification and found that normal-hearing listeners and listeners with hearing impairment benefit equally from a single early reflection. Additionally, word recognition was unaffected by reflections arriving between 0-20 ms after the direct sound. In a subsequent study, it was demonstrated that the energy from ER could increase the Signal-to-Noise (SNR) by up to

9 dB for normal hearing listeners and hearing-impaired listeners (Bradley & Sato, 2003). Warzybok's (2013) expanded upon the existing models that predict binaural speech intelligibility based on the temporal integration of a single early refection. Warzybok et al. (2013) conducted several experimental models to measure the degree to which a single reflection impacts speech intelligibility. The spatial configurations of the sources utilized in this experiment varied between the experimental models. The target speech was always presented directly in front of the listener (0 degrees), while the noise was presented frontally (0 degrees), laterally (135 degrees), or diffusely. The azimuth of the single reflection of either the target speech or the noise signal varied in 45-degree steps.

In the first experimental model, the target stimuli were presented to the listener with a frontally delayed reflection which was a copy of the target stimuli. The time delays of the target signal were 0, 10, 25, 50, 75, 100, or 200 ms. The two-different speaker configuration of the noise signals consisted of a direct source with or without an added reflection. The time delay of the single reflection of the noise signal were 10, 25, and 50 ms. Additionally, the influence of amplitude in reference to the target stimuli was also considered. The findings of the first experimental model demonstrated that the influence of a single reflection on speech intelligibility in noise for delays up to about 50 ms was not impacted by different reflection amplitudes related to the target signal or by adding a single reflection to the noise signal. Consequently, speech intelligibility in noise was reduced by a single reflection with a time delay of 200 ms.

In the remaining experimental models, speech and noise were not colocated, and no reflection was added to the noise source. Additionally, several different factors were also considered and varied within the experimental models. These factors included: the single reflection azimuth (øR), the type of noise (i.e., dichotic, laterally located, diffuse), and the time delay of the reflection (Δt). In the second experimental model, Warzybok et al. (2013) found that a single frontal reflection can be completely integrated with the direct speech signal up to 50 ms for dichotic and laterally located noise and 25 ms for diffuse noise. Consequently, in all noise conditions, SRTs increased when the time delay for the reflections was larger than 50 ms. The researcher's findings suggested that the amount of reduction in speech intelligibility becomes greater as the time delay of single reflection increases above 50 ms, because the later-arriving reflection may not effectively integrate with the direct signal. Moreover, spatial separation of the direct sound and reflection in lateral and diffuse noise was also investigated. Spatial release from masking was greater for the laterally located noise condition (5.2 dB) than in diffuse noise condition (2.5 dB). Therefore, spatial separation of the direct speech signal from the later-arriving reflection substantially reduces the masking effect contributed to the late reflection. To summarize, the findings from Warzybok and colleagues are in agreement with the findings of Nabelek and Robinete (1978) and Bradley and Sato (2003). Collectively these studies showed that early reflections improve speech intelligibility by increasing the loudness of the direct sound. Furthermore, unlike early reflections, late reflections may decrease a listener's ability to accurately identify speech.

Late Reflections

Late reverberation is the multiple delayed and attenuated reflections that reach a listener's ears (Srinivasan et al., 2017). Late reverberation degrades speech intelligibility due to different masking effects (Srinivasan, Stansell, & Gallun, 2017; Nabelek & Robinete, 1978). Nabelek and colleagues (1978) described the effects of reverberation as a function of two main components: self-masking and overlap-masking. *Self-masking* refers to the distortion occurring within each phoneme by reverberation. *Overlap-masking* is the distortion that occurs when acoustic information from previous phonemes spills over into the subsequent speech components. Both self-masking and overlap-masking combine to distort the spectral and temporal cues important for speech perception. For example (see Fig. 1.3 below), the reverberant sound energy masks the original speech signal by producing a smearing effect of the speech envelope. Similarly, reverberation results in alteration of the Spectro-temporal characteristics of the individual phonemes (Nabelek & Robinette, 1978; Nabelek et al., 1989). Finally, reverberation produces frequency variability, which decreases the listener's sensitivity to binaural difference cues, and leads to the deterioration of the amplitude spectrum of the speech signal (Marrone et al., 2008; Srinivasan et al., 2017).

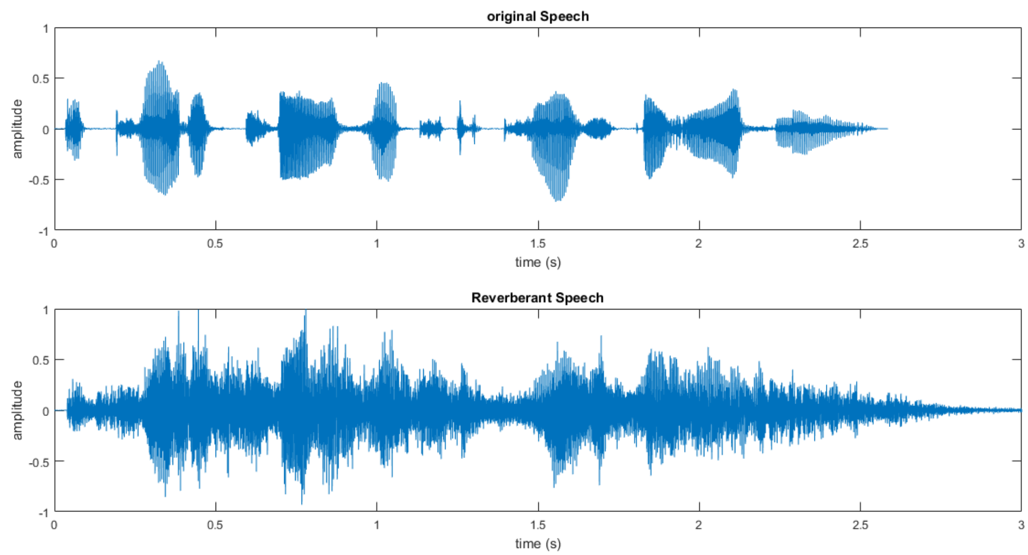

Figure 3. Illustration of an original and reverberant speech signal. The original speech signal (top row) shows separable and distinct syllables with silent gaps between the speech sounds. Note in the reverberant speech signal (bottom row) how the reverberant energy smears the original speech signal and the silent gaps between syllables are filled with noise. As reverberation energy increases speech intelligibility decreases.

Researchers have theorized that the later sound reflections arriving at the listeners' ears degrade the interaural coherence of the target signal, which can negatively impact spatial release from masking (Lavandier & Culling, 2007, 2008). Researchers have defined coherence as the direct-to-reverberant energy ratio arriving at the listeners' ears, and can vary depending on the source, reverberation characteristics of the room, and the listener location. Lavandier and Culling (2007) conducted four experimental models to determine the detrimental effects of reverberation on the spatial release from masking in a simulated reverberant space produced under headphones. In experiments 1 and 2, the researchers explored the influence of interaural coherence on the interferer by manipulating the following characteristics, which included: 1) increasing the distance of the interferer and 2) by manipulating the room absorption coefficient for the interferer. The findings of the first two experimental models revealed that SRTs increased as the interferer signal distance increased, and the absorption coefficient decreased, with effects ranging from 2-4 dB. Moreover, the findings of experiments 3 and 4 demonstrated that SRTs increased as the target signal distance increased and as the absorption coefficient decreased. Researchers concluded that speech intelligibility depends on the azimuth of separation of sound sources in addition to the interaural coherence and the direct-to-reverberation ratio energy arriving at the listener.

In a subsequent study, Lavandier et al. (2008) used comparable methods described in the previous study to investigate the influence of reverberation on speech intelligibility for spatially separated target and masking sources. The binaural room impulse response utilized allowed for the manipulation of the absorption coefficient of the room characteristics and the distance of the sources. Therefore, the researchers could modify the direct-to-reverberant ratio and the interaural coherence of the sources of masking and the target signal. Although the experimental design is not an accurate representation of a real-world listening environment, the study design allowed the researchers to observe the individual effects of the interferer and target signal reverberation characteristics independent of each other. The researchers concluded that the later reflections diminished speech intelligibly initially by reducing the correlation of the sources of masking at the listener's ear. Therefore, the effectiveness of the masker was increased. The second effect demonstrated in the experiment was the degradation of the intelligibility of the target signal. To summarize, researchers suggested that the interaction observed between the effects of reverberation on the target signal and the sources of speech masking required further investigation.

Effects of Reverberation on SRM

The reduction in spatial release from energetic masking due to reverberation is demonstrated in the study conducted by Kidd, Mason, Brughera, and Hartmann (2005). Kidd, Mason, Brughera, and Hartmann (2005) utilized a similar CRM design as Arbogast et al. (2002) to investigate the spatial release from masking in three room conditions (i.e., FOAM, BARE, and PLEX) that produced acoustic differences ranging from least reverberant to most reverberant respectively. The target was always presented at 0°. The maskers were presented at 0° (i.e., target 0°, masker at 0°) and spatially separated from the target (i.e., target 0°, masker 90°). For the same band noise (SBN) (primarily energetic speech masking) condition, the group average SRM reduced from 7.9 dB in the FOAM to 2.0 dB in the PLEX condition. Additionally, for the different band speech (DBS) (primarily informational speech masking) the SRM was larger when compared to energetic masking. The SRM for the DBS condition was 14.9 dB, 16.7 dB, and 16.0 dB for the FOAM, BARE, and PLEX conditions respectively. In the less reverberant condition, the acoustic advantage that occurs due to the head shadow effect provided the listener with a binaural release from masking. However, the authors indicated that the reduction in threshold occurs because the energy produced by reverberation significantly reduced the ITDs and ILDs for the PLEX condition relative to the other conditions. The authors concluded that for the SBN energetic masker, reverberation reduced the benefit of spatial separation between the target signal and the maskers.

Effects of Hearing Loss and Age on the Benefit of Spatial Release from Masking in a Reverberant Environment

Marrone et al. (2008) examined the interaction between hearing loss, aging, and reverberation on a large spatial separation (±90°) from masking. A three-talker interferer either presented colocated with the target (0°) or positioned at a (±90°) azimuth on the horizontal plane. The results indicated that on average, listeners with hearing loss demonstrated less benefit from SRM than normal hearing listeners, in both a reverberant and non-reverberant environment (Marrone et al., 2008). Of important note, Marrone et al. (2008) indicated that most of the spatial release from masking occurs between 0°-15°. However, Marrone et al. (2008) were unable to distinguish the effects of aging from hearing loss due to a small sample size. To conclude, the results indicated that the listener’s hearing loss was the primary factor influencing the reduced benefit of spatial separation. In a subsequent study, Srinivasan et al. (2016) differentiated the contributions of aging and hearing loss on SRM at small spatial separations. These findings prompted further research for various spatial separations in a reverberant environment.

Srinivasan et al. (2017) examined SRM and the role of early and late reflections at various separations between the target and maskers and concluded that hearing loss and age were significant predictors of SRM in the late reverberation condition. These results suggest that aging plays a role in speech intelligibility in reverberant listening environments when all early reflections are removed. Specifically, it reduces the effective signal-to-noise ratio between the direct speech and reverberation and thus makes the signal less intelligible.

Purpose Statement

The purpose of this current study is to utilize three deconvolving techniques to obtain speech recognition thresholds and to calculate the amount of spatial release from masking. Older adults with and without hearing impairment will participate in this study to investigate the effects of reverberation, aging, and hearing loss on spatial release from masking. This study will specifically evaluate the effects of the binaural room impulse response on speech understanding when spatial cues are absent. We will utilize three different experimental conditions, which include: underestimated effects of reverberation, overestimated effects of reverberation, and correct estimation of reverberation. We reasoned that if the correct estimation of reverberation were calculated and de-reverberated, then we would expect a decrease of the deterioration effects of reverberation on spatial release from masking.

CHAPTER 3

Methodology

Participants

A total of 15 participants, aged 47 to 77 years (*M* = 59 years), with various hearing capabilities were recruited from Towson University Hearing and Balance Center (TU-HBC) based on age and hearing status using a flyer (*See Appendix B*). An effort was made to have equal number of male and female participants, but more females (*n* = 10) participated than males (*n* = 5). All participants signed a consent form (*See Appendix A*). All participants were older adults with varying degrees of hearing capabilities. Six participants had pure-tone averages (PTAs) ≤ 20 dB hearing level (HL) [hearing level re: ANSI 3.6-2004 (ANSI, 2004)]. Five participants had PTAs ≥ 20 dB HL. PTA values are the averaged thresholds of the octave frequencies of 500, 1000, 2000, and 4000 Hz. The PTA values were averaged across both ears for all subsequent analysis. The average hearing loss of all participants at the tested audiometric frequencies between 250 Hz and 8000 Hz (i.e., 250, 500, 1000, 2000, 4000, 6000, 8000 Hz) are shown in figure 4. In addition, all listeners had air conduction and bone conduction thresholds of ≤ 45 dB HL or better at octave frequencies of 2000 Hz and below, air conduction and bone conduction thresholds of ≤ 65 dB HL at 4000 Hz, and air conduction thresholds that do not exceed 75 dB HL at 6000 and 8000 Hz.

All listeners had symmetrical hearing with no differences exceeding 10 dB at more than one frequency, and no differences exceeding 15 dB at any tested frequency. Four participants did not meet the hearing criteria for the study due to an asymmetrical hearing loss as indicated by audiometric thresholds that differed more than 15 dB across ears. Listeners that participated in the study were reimbursed for their time (*See Appendix E*).

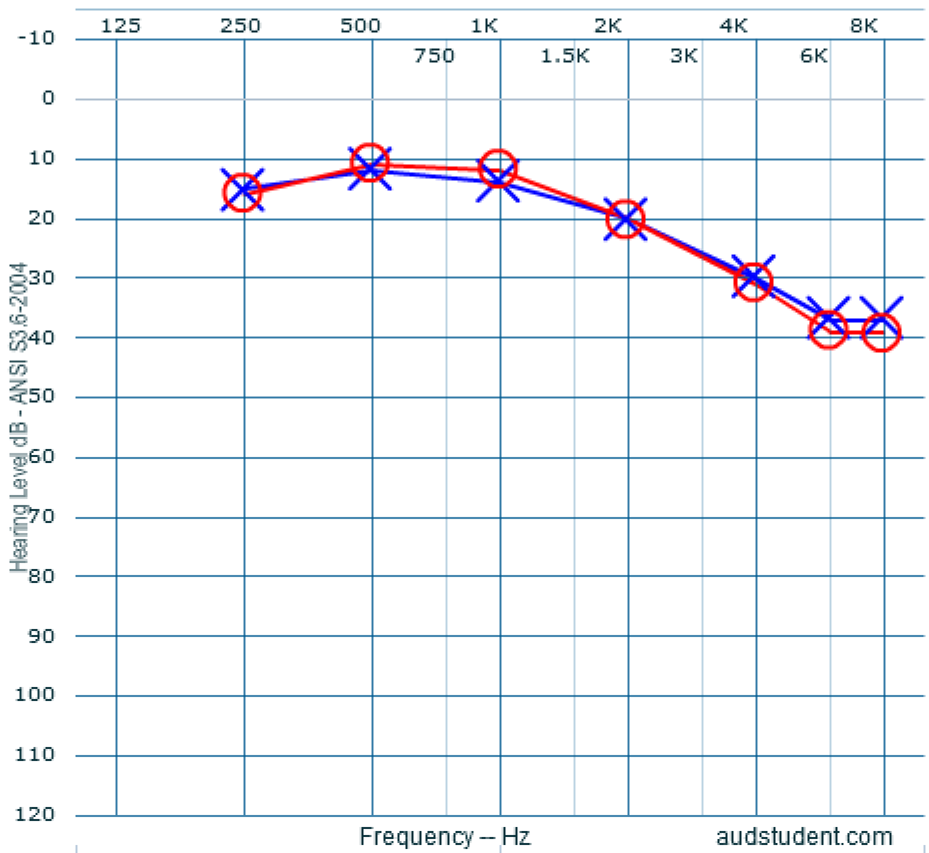

Figure 4. The figure represents the average thresholds for all participants (*n* = 11) across the audiometric frequencies between 250-8000 Hz (dB HL).

Procedure

Each trial began with an otoscopic examination to evaluate the participant's ear canal to determine if any impacted cerumen or foreign objects were present. Tympanometry was administered to evaluate the participant's middle ear function. Normal middle ear function in each ear was defined as follows: Middle ear pressure ranging from +50 to -150 daPa and static compliance values ranging from 0.2-1.5 ml (Hunter & Sanford, 2015). No air-bone gaps greater than 10 dB were present at octave frequencies from 500 Hz to 4000 Hz.

All participants were in good health and had no self-reported history of otologic disorders as reported in the case history form completed prior to testing (*See Appendix C*). Researchers have indicated that the Mini-Mental State Examination (Folstein et al., 1975) is commonly used as a screening tool to detect dementia (Tariq, Tumosa, Chibnall, Perry, & Morley, 2006). However, Tariq et al. (2006) reported findings which indicated that the Saint Louis University Mental Status Examination (SLUMS) is more sensitive to mild cognitive impairments. Therefore, the Veterans Affairs Saint Louis University Mental Status examination (VASLUMS) was administered prior to testing to the three groups to measure cognitive impairment (*See Appendix D*). The VASLUMs examination is an 11-item screening tool that was used to evaluate cognitive functions including, but not limited to: attention, immediate and delayed recall, language, ability to follow simple commands, calculation, and orientation (Tariq, Tumosa, Chibnall, Perry, & Morley, 2006). A 30-point scale is used to assess the impact of the cognitive impairment. High scores correlate closely with normal cognition. All participants had scores of 25 or higher on the Saint Louis University Mental Status Examination (Tariq et al., 2006) which ruled out dementia or any other mild cognitive impairment.

Pure-Tone Behavioral Audiometric Test Protocol

All responses for conventional pure-tone audiometry were obtained utilizing an Audio Star Pro audiometer that was calibrated to ANSI 2004 standards. Audiometric thresholds were obtained in accordance with Modified Hughson-Westlake procedure using pulsed pure-tone stimuli at all octave frequencies between 250 Hz and 8000 Hz. Each participant received a set of instructions in accordance with ANSI S3.21-2004: “(1) Indicate the purpose of the test, to find the faintest tone that can be heard. (2) Indicate the need to respond whenever the tone is heard, no matter how faint it may be. (3) Indicate the need to respond overtly as soon as the tone comes on and to respond overtly immediately as the tone goes off. (4) Indicate that each ear is to be tested separately" (ANSI, 2004, p. 4). Testing was administered using TDH-49 Supra Aural headphones.

Stimulus

Sentence intelligibility and spatial release from masking were evaluated utilizing the Coordinate Response Measure (CRM; Bolia et al., 2000), which is a database of simple sentences with the form "Ready [CALL SIGN] go to [COLOR] [NUMBER] now". The CRM measures the participant's ability to correctly identify target messages in the presence of a competing speech. In the present study, all available phrases for three of the four male talkers in the CRM will be used for the target and speech maskers. Srinivasan et al. (2016) reported that the speaking rate of the fourth talker was slower when compared to the other three speakers and hence will be excluded from the current study. CRM corpus has eight possible call signs (Arrow, Baron, Charlie, Eagle, Hopper, Laker, Ringo, and Tiger), four colors (Blue, Red, White, and Green) and eight numbers (1-8) (Bolia et al., 2000). All available phrases, bandpass filtered from 80 Hz to 8000 Hz, were delivered via HD650 Circum-aural headphones. For future reference, the sources of masking will be referred to as '*speech maskers'*. Reader(s) can review Jakien et al. (2017) for a complete description of the test-retest reliability of the headphone-based spatial release from masking task with two speech maskers used in this study.

Binaural Room Impulse Response Modeling Procedure

Simple room-acoustic models were used to simulate the reverberant environment with the following dimensions: Length: 5.7m; width: 4.3m; height: 2.6m. The broadband (125 – 4000 Hz) reverberation times (T_{60}) of 1 and 2 secs were simulated utilizing techniques described in Zahorik (2009). A synthesized description of room simulation techniques based on acoustic models of room environments are described below. Reader(s) can review Zahorik (2009) for a complete description of the room simulation techniques.

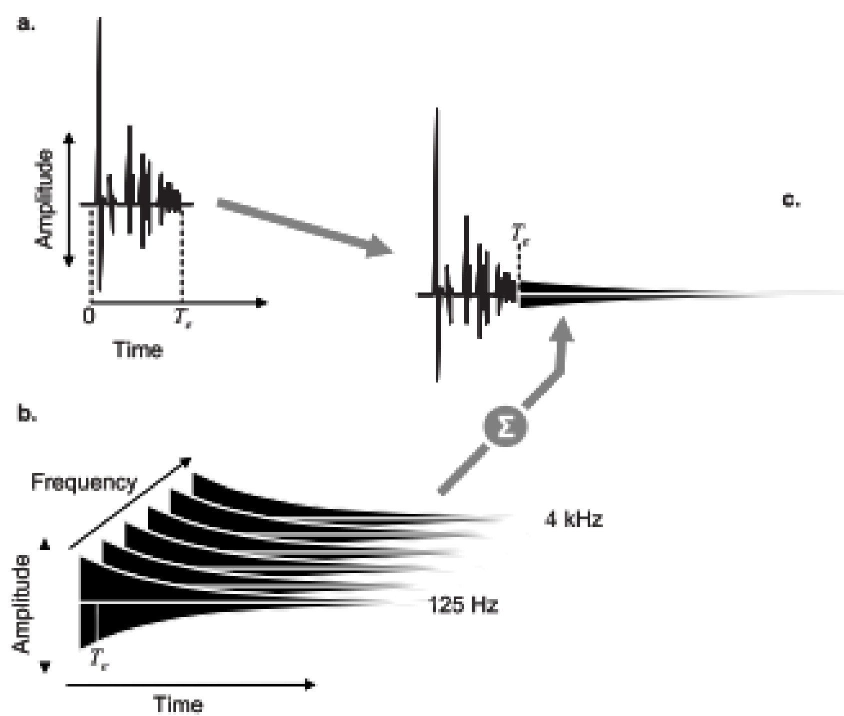

Figure 5. Is an illustration of the BRIR modeling procedure described by Zahorik (2009). The (a) estimated early reflections modeling procedure, (b) late reverberation modeling procedure, and (c) combined early and late responses modeling procedure are described in further detail below. *Source*: From Zahorik (2009).

a. *Early response modeling*: this simulation method utilizes an image model (Allen and Berkley, 1979) to compute a three-dimensional image-model, which includes: directions, delays, and attenuations of early reflections. Then, the direct source and the early reflections were spatially rendered with non-individualized head-related transfer functions (HTRFs).

b. *Late response modeling*: Late reverberant energy was simulated statistically utilizing exponentially decaying independent Gaussian noise samples in octave bands from 125 – 4000 Hz for each ear. The decay functions were derived from the Sabine equation (Sabine, 1922; as referenced in Zahorik, 2009).

c. *Combining early and late reflections*: The resulting late responses for each ear were summed with the early responses for each ear to generate an estimated BRIR.

Overall, this simple method of room simulation was determined to produce BRIRs that closely approximate the physical and perceptual characteristics of BRIRs measured in a real room (Zahorik, 2009).

Signal processing techniques were implemented to create two different reverberant environment conditions (i.e., T_{60} of 1 and 2 secs). The two-room simulations had identical dimensions of 5.7m x 4.3m x 2.6m but differed in absorptive properties. For this model, average absorption coefficient values were estimated based on published absorption coefficient values for common building materials (Moulder, 1991; as referenced in Zahorik, 2009). All three parts of the BRIR (DS, ER, and LR) were retained for both reverberant environment conditions (Srinivasan et al., 2017).

The binaural room impulse response was calculated using Zahorik's (2009) simulation techniques, in order to study the effects of the simulated reverberation on speech. A virtual space array (VSA) was presented over Sennheiser HD650 Circum-aural headphones to simulate a multi-talker speech signal presented from loudspeakers in a reverberant room. The two-different simulated reverberant rooms resemble the signals that would occur if the target signal and speech masker signals had been presented from loudspeakers and the listener was positioned at a given location in a real room. Within each simulated room, the listener was positioned as if they were in the center of the room facing toward the front loudspeaker. The distance between the target speech signal and the listener was approximately 1.4 meters. The two spatial configurations that were used are as follows: colocated (all three sentences presented from 0° azimuth) and spatially separated (target signal presented at 0° azimuth, two different speech maskers presented at $\pm 30°$).

Next, the reverberant sample of the target and masker signals was produced by convolving the speech sample (i.e., CRM phrase) with the calculated room impulse response (RIRs) of the two different conditions for the appropriate location relative to the listener (Srinivasan et al., 2017). Consequently, the same reverberant sample of speech can be deconvolved. In short, inverse filtering of room transfer functions was utilized to deconvolve reverberant speech. Then, the left BRIR and right BRIR were deconvolved under three different reverberation conditions: 1) underestimating the effects of reverberation, 2) overestimating the effects of reverberation, and 3) correct estimation of reverberation. For instance, *underestimation* T_{60} is 0.5 s less than actual (i.e., 0.5s and 1.5s for 1s and 2s reverberant rooms, respectively), *overestimation* T_{60} is 0.5 s more than actual (i.e., 1.5s and 2.5 s for 1s and 2s reverberant rooms, respectively), *Correct estimation* T60 is same as actual (i.e., 1s and 2s for 1s and 2s reverberant rooms, respectively). The block diagram below illustrates the signal processing techniques incorporating convolution and deconvolution of the binaural room impulse response.

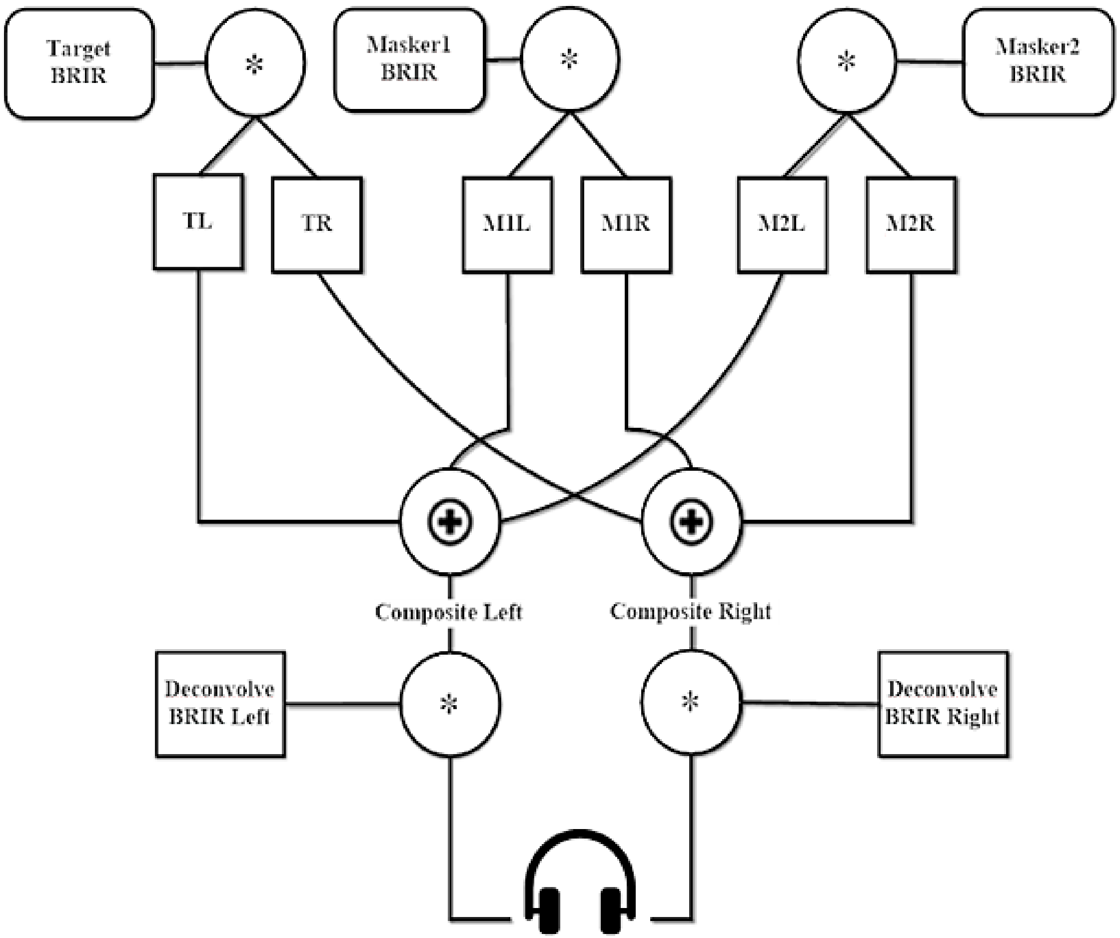

Figure 6. A block diagram of the signal processing techniques implemented to create different reverberant environment conditions. The target signal was convolved with the BRIR for the target signal and two speech masker signals. Signal type was indicated by the terms target or masker. Spatial conditions are as follows: Colocated (BRIR same for T, M1, and M2). Spatially separated (BRIR different for T, M1 and M2). The convolved signal for the right and left composite is then deconvolved and presented over headphones. BRIR = binaural room impulse response; T = target; M = masker.

Procedure

Participants were tested in a double-walled sound-attenuated booth made by Industrial Acoustics Company located at the Towson University Van Bokkelen Hall. Participants listened to speech stimuli presented over Sennheiser HD650 circumaural headphones. For every trial, the listener was presented with a set of three simultaneous CRM sentences. The objective was to attend to the sentence identified by the callsign "Charlie" and ignore the two speech masking sentences. The target and speech maskers varied randomly from trial to trial. Brungart (2001) investigated the intelligibility of each of the call signs spoken by the eight talkers and concluded that there is no significant advantage to a listener when instructed to identify any of the individual CRM talkers. However, to avoid additional spatial cues arising from temporal modulations of the speech maskers, it was determined that all the speech masker signals are to be of the same gender as the target talker (Brietsprecher, 2011).

Threshold Estimation

The threshold estimation procedure implemented the use of equal sensation level (SL) to obtain the level of the target sentence. This was achieved by adding 20 dB SL to the listener's average PTAs (i.e., 500, 1000, and 2000 Hz) to obtain the level of the target sentence, which was always fixed during the experiment. The two masker sentences were presented at certain fixed levels relative to the target sentence and were appropriately scaled in SL to obtain required TMRs (Srinivasan et al., 2017).

Then, detection thresholds in the anechoic environment were estimated using a progressive tracking procedure (Gallun et al., 2013). This progressive tracking procedure involves a descending presentation technique involving presenting 20 trials, specifically two presentations at each TMR starting at 10 dB TMR and ending at -8 dB TMR (i.e., decreasing in steps of 2 dB after two trials). The TMR was determined after the completion of 20 trails by subtracting the number of correct responses from 10 dB. For instance, if the listener reported all the target sentences correctly the TMR threshold would be an estimated -10 dB (i.e., 10 dB – 20 correct responses). If the listener reported all the target sentences incorrectly the TMR would be roughly 10 dB (i.e., 10 dB – 0 correct response). Previous research has shown that this progressive tracking procedure can be used to rapidly and accurately estimate thresholds in both the colocated condition and the spatially separated condition in order to assess the spatial release from masking that a listener can achieve in each of these conditions (Gallun et al., 2013; Srinivasan et al., 2017).

Performance was measured in twelve different conditions: colocated and spatially separated for all three reverberation estimations in two different simulated reverberant environments. The target sentences were presented at a fixed intensity, while the level of the speech masker varied. Specifically, we used a one-up/one-down procedure to estimate the speech reception threshold (SRT) (i.e., TMR required to understand the target phrase (i.e., color number combination) 50% of the time (Levitt, 1971). Responses were recorded on a computer monitor located in front of the listener. Feedback was delivered after each presentation in the form of “Correct” or “Incorrect”. Data collection was self-paced, and listeners were instructed to take breaks as needed. The procedures were approved by the Towson University Institutional Review Board and all listeners were monetarily compensated for travel and their time in the form of a gift card. All stimulus presentation and data collection were completed utilizing MATLAB; statistical analysis of the recorded data were performed using SPSS Version 23 (IBM CORP, Armonk, NY, USA) and Microsoft Excel version 2016.

Data Analysis

The data was analyzed using SPSS Version 23 to perform several multiple regression analyses and four repeated measure analyses of variance. Multiple regression involves investigation of the effect of each predictor variable on a dependent variable while simultaneously considering the effects of other predictor variables. Repeated measure analyses of variance involved a statistical method used to test differences between two or more means for correlated samples. Additionally, the descriptive statistical analysis was performed to obtain Mean, Standard Deviation, and Standard error values. The predictor variables that were analyzed are the three different reverberant conditions, age, and hearing loss. Finally, correlation and multiple regression analyses were conducted to examine the relationship between SMR values and hearing loss (as defined by PTA) and age (in years).

CHAPTER 4

Results

Analyses focus on participants' Target-to-Masker (TMR) identification thresholds to the 12 different conditions in which listeners were presented with a set of three simultaneous CRM sentences over headphones. Performance was measured in terms of 3 different conditions. These conditions include: three reverberation times (i.e., T_{60} = 0s, T_{60} = 1s, and T_{60} = 2s); two different spatial configurations (colocated and spatially separated), and three different dereverberation processing techniques (i.e., correct, over, and underestimation). In the first experiment, target and masker sentences were either colocated (target and two speech maskers located at 0° azimuth) or spatially separated (target located at 0°; two speech maskers located at ±30°) for all three simulated reverberant environments (i.e., T_{60} = 0s, T_{60} = 1s, and T_{60} = 2s). In the subsequent experiments, target and masker sentences were either colocated or spatially separated for two different reverberation times (T_{60} = 1s and T_{60} = 2s) and three different BRIR estimations were used to remove the effects of reverberation on presented speech.

A total of four repeated measure analysis of variance (ANOVAs) were performed on the data to determine the effects of the within-subject factors of BRIR Estimation (i.e., Correct, Over, and Under Estimation), Reverberation Times (i.e., T_{60} = 0s, T_{60} = 1s, T_{60} = 2s), and Spatial Conditions of the target talker and maskers (0° and ±30°) on TMR identification thresholds. In addition, correlation and multiple regression models were performed to investigate the relationship between SRM and various potential predictors.

Identification Thresholds in Anechoic Listening Condition

First, we review the test-retest reliability of the headphone based spatial release from masking task performed in the anechoic condition (T_{60} = 0s). TMR identification thresholds were analyzed for 8 trials in the anechoic condition. Two participant's data were excluded from this analysis because they did not complete the 4 trials at the end of the study due to time constraints. Mean values were calculated for the 4 trials performed at the beginning of the study and were then compared to the 4 trials at the end of testing (n = 9 participants). This allowed us to examine, for example, whether participants TMR thresholds improved as a result of a learning effect. A paired sample t-test was performed on the data [i.e., anechoic condition] to determine whether there was a statistically significant mean difference between the TMR thresholds obtained at the start of the testing session and at the conclusion of the testing session. No statistically significant differences in TMR identification thresholds were observed between the two trials. As shown in Table 1, colocated TMR identification thresholds were not statistically significantly different in the before trial (M = 2.33, SE = 0.32) when compared to the after trial (M = 2.77, SE = 0.40), [t (17) = -1.054, p = .307, r = .247]. Similarly, separated TMR identification thresholds were not statistically significantly different in the before trial (M = 0.50, SE = 0.70) when compared to the after trial (M = -.1667, SE = 0.91), [t (17) = .753, p = .462, r = .273]. Therefore, we can infer that there is no learning effect associated with the identification task. Data from the two conditions were thus combined together for all subsequent analysis. The results described above can be found in Table 1.

Table 1

Paired Sample T-Test for Spatial Release of Target Talker from Masker Talkers (in dB) for Anechoic Listening Environment Three Different Listening Environments

TMR (dB)		*M*	*SD*	*SE*	95% CI LL	95% CI UL	*t*	*df*	*Sig.*
Pair 1	Colocated Threshold Before - Colocated Threshold After	-0.44	1.78	0.42	-1.33	0.44	-1.054	17	.307
Pair 2	Separated Threshold Before - Separated Threshold After	0.66	3.75	0.88	-1.20	2.53	0.753	17	.462

Note. M = mean. *SD* = standard deviation; *SE* = standard error of mean; CI = confidence interval; LL = lower limit; UL = upper limit; *df* = degrees of freedom.

Identification Thresholds of Anechoic (0s) and Reverberant Environments (1s & 2s)

A 2 x 3 repeated measures ANOVA was run to determine the effect of three different reverberation times (T_{60} = 0s, T_{60} = 1s, and T_{60} = 2s) and two different spatial conditions (colocated = target and masker sentence located at 0°; spatially separated = target sentences located at 0° and masker sentences located at ±30°) on TMR identification thresholds. Mauchly's test indicated that the assumption of sphericity had not been violated for the main effect of Reverberation time, $\chi(2) = .079$ $p = 0.961$. Therefore, degrees of freedom did not need to be corrected. Next, we report the two effects from this analysis as follows:

As expected, the mean TMR thresholds increased as reverberation time increased from 0s to 2s. There was a significant main effect of reverberation time, [F (2, 20) = 57.553, $p < .001$, partial $\eta^2 = 0.852$] on TMR identification thresholds indicating that anechoic condition T_{60} = 0s yielded better (i.e., lower) identification thresholds than the reverberation conditions T_{60} = 1s and T_{60} = 2s. There was a statistically significant increase in TMR identification thresholds based on the listening conditions from the anechoic (*M* = 1.18 dB) condition when compared to the T_{60} = 1s (*M* = 4.06 dB) and the T_{60} = 2s (*M* = 5.21) conditions, p < .05 (see figure 7)

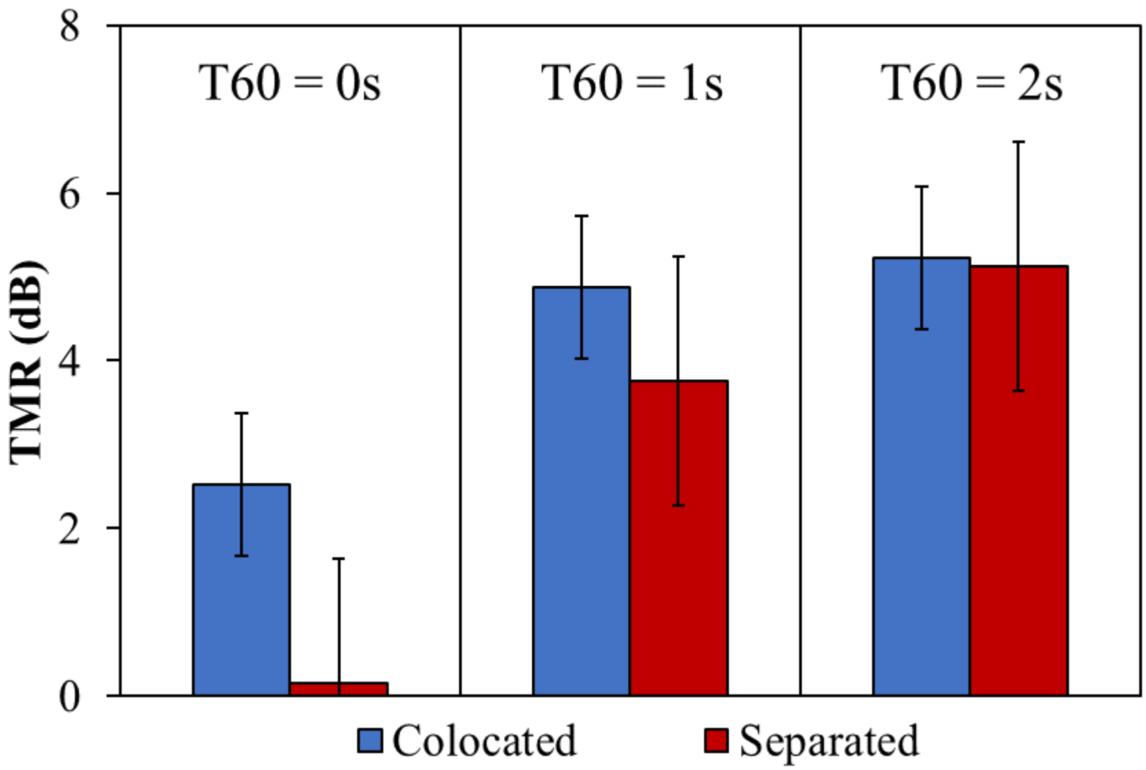

Figure 7. Mean values representing TMR identification thresholds (dB) for each listening environment and spatial condition. The left column shows the average TMR identification thresholds for the anechoic (T_{60} = 0s) condition. The center column and right column show the average TMR identification thresholds for the reverberant listening environments of T_{60} = 1s and T_{60} = 2s, respectively. Within each panel, the spatial conditions of colocated (0°) are denoted by blue bars and spatially separated (±30°) by red bars. Error bars indicated ±1 SEM.

Post hoc analysis pairwise comparisons with Bonferroni adjustments revealed significant differences in identification thresholds between listening environments with the anechoic condition T_{60} = 0s having elicited statistically significantly lower TMR identification thresholds than the T_{60} = 1s listening condition (M_{diff} = 2.906 dB, 95% CI [-4.070, -1.741 dB], *SE* = 0.406, *p* < .001), and T_{60} = 2s (M_{diff} = -4.056 dB, 95% CI [-5.170, -2.94 dB], *SE* = 0.388, *p* < .001). Identification thresholds in the T_{60} = 1s listening condition were statistically significantly different than TMR thresholds in the T_{60} = 2s condition (M_{diff} = -1.151 dB, 95% CI [-2.26, -.076 dB], *SE* = 0.374, *p* = .035).

Furthermore, there was a significant main effect of spatial condition [*F* (1, 10) = 17.455, *p* = .002, partial η^2 = 0.636] on TMR identification thresholds indicating that the spatially separated condition (M = 2.96) elicited lower identification thresholds when compared to the colocated condition (*M* = 4.086) than the colocated condition. *Post hoc* analysis pairwise comparisons with Bonferroni adjustments revealed significant difference in identification thresholds between spatial conditions with the spatially separated condition having statistically significantly lower TMR identification thresholds than the colocated condition (M_{diff} = -1.179 dB, 95% CI [-1.807, -0.550 dB], *SE* = 0.282, *p* = .002).

There was a borderline significant interaction between reverberation time and spatial separation, [*F* (2, 20) = 3.409, *p* = .053, partial η^2 = 0.254]. To break down this interaction, *Post hoc* analysis using three paired-sample t-tests with Bonferroni adjustments were used to determine whether there was a statistically significant mean difference between the TMR thresholds obtained in the colocated condition compared to the spatially separated condition in the three different reverberant environments. A paired sample t-test indicated that there was a significant difference between the TMR identification threshold in the colocated condition (*M* = 2.36, *SD* = 1.14) and the spatially separated condition (*M* = 0.00 dB, *SD* = 2.71 dB, *SE* = 0.81) in the anechoic listening environment; [*t* (10) = 3.46, *p* = .006, *r* = .738)]. However, there was no statistically significant difference between colocated and spatially separated thresholds in the 1s and 2s conditions [T_{60} = 1s: *t* (10) = 1.94, *p* = .081, *r* = .523; T_{60} = 2s: *t* (10) = .027, *p* = .979,

r = .01]. To summarize, there was a significantly better spatial release from masking obtained in the anechoic condition when compared to the T60 = 1s condition, $p < .05$. There was a notable decrease in SRM from the T60 = 1s condition when compared to the T60 = 2s condition, but this difference was not deemed statistically significant, $p > .05$. Figure 2 depicts the mean values based on the spatial release from masking in the three different listening environments. Descriptive statistics can be found in Table 2 and the results for paired sample t-test are provided in Table 3.

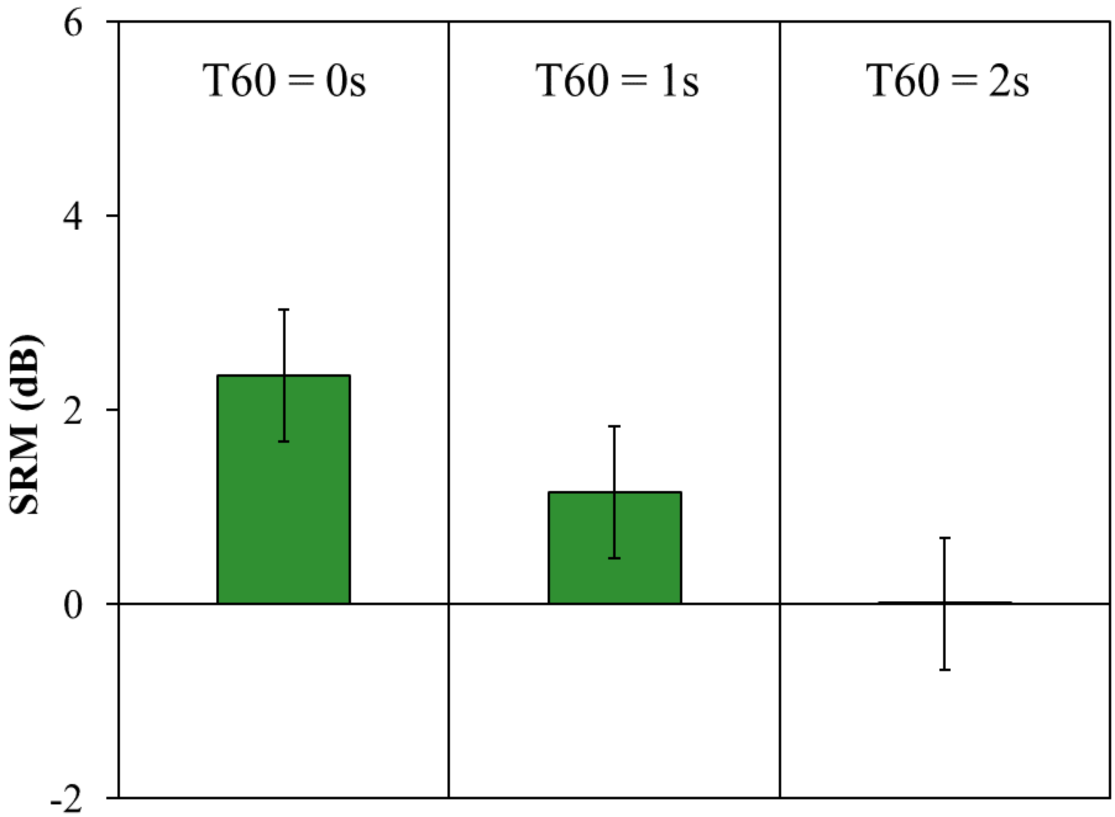

Figure 8. This figure shows the average SRM (as indicated by the difference of colocated TMR identification thresholds and spatially separated TMR identification thresholds) as a function of three different reverberant listening environments for the individual listeners. The left column shows the average SRM for the anechoic (T60 = 0s) condition. The center column and right column show the average SRM for the reverberant listening environments of T60 = 1s and T60 = 2s, respectively. Error bars indicate ±1 SEM.

Table 2.

Descriptive Statistic Values for TMR Identification Thresholds (in dB) of Anechoic (T_{60} = 0s) and Reverberant (T_{60}s = 1 and 2s) Listening Environments

Anechoic and Reverberant Listening Environments									
Spatial Condition	Anechoic T_{60} = 0s			Reverberation Time T_{60} = 1s			Reverberation Time T_{60} = 2s		
	n	*M*	*SD*	*n*	*M*	*SD*	*N*	*M*	*SD*
Colocated (0°)	11	**2.36**	1.14	11	**4.66**	1.53	11	**5.24**	1.41
Separated (±30°)	11	**0.00**	2.71	11	**3.50**	1.96	11	**5.23**	1.49

Note. Mean values are in boldface. *SD* = standard deviation.

Table 3

Paired Sample T-Test for Spatial Release of Target Talker from Masker Talkers (in dB) for Three Different Listening Environments

	Paired Differences							
				95% CI				
Condition	*M* (dB)	*SD*	*Std. Error M*	LL	UL	*T*	*df*	*Sig.*
0s	2.36	2.26	0.81	0.84	3.88	3.46	10	.006
1s	1.16	1.98	0.59	-0.17	2.49	1.94	10	.081
2s	0.01	1.57	0.38	-1.04	1.07	.027	10	.979

Note. CI = confidence interval; LL = lower limit and UL = upper limit, *df* = degrees of freedom; *M* = mean, *SD* = standard deviation.

Repeated Measures ANOVA (3 BRIR x 2 Reverb Times x 2 Spatial Conditions)

A 3 x 2 x 2 repeated measures ANOVA on mean TMR identification thresholds were conducted with Binaural Room Impulse Response estimations (correct: de-reverberation same as target BRIR's T_{60}; overestimation: de-reverberation T_{60} is 0.5s more than target BRIR's T_{60}, and underestimation: de-reverberation T_{60} is 0.5s less than target BRIR), spatial separations (0° and ±30°), and reverberation times (T_{60} = 1s and 2s) as within-subject factors. Mauchly's test indicated that the assumption of sphericity had not been violated for the main effect of BRIR Estimation, $\chi(2) = 1.969$ $p = 0.374$. Assumption for reverberation times and spatial conditions are already met since there are only two levels. Therefore, degrees of freedom did not need to be corrected.

There was a significant main effect of the type of BRIR estimation on TMR identification thresholds, [F (2, 20) = 6.540, p = .009, partial η^2 = 0.395]. There was no significant main effect of the type of spatial separation, [F (1, 10) = 2.789, p = .126], or reverberation time, [F (1, 10) = 4.883, p = .052, partial η^2 = 0.328], on TMR identification thresholds. To explore this main effect of BRIR further, *Post hoc* comparisons were conducted to evaluate pairwise differences among participants TMR identification thresholds with the use of Bonferroni adjustments. Pairwise comparisons revealed that participants obtained statistically significantly lower (better) TMR thresholds in the under estimated condition when compared to the correct estimation condition (M_{diff} = -0.700 dB, 95% CI [-1.223, -.175], SE = 0.183, p = .010) and the overestimation condition (M_{diff} = -0.776 dB, 95% CI [-1.509, -.043], SE = 0.255, p = .037). There was no statistically significant difference between the correct estimation and the over estimation BRIR conditions (M_{diff} = -.077 dB, 95% CI [-.832, -.679], SE =

0.263, $p > .05$). Additionally, pairwise comparisons revealed a sizeable difference between TMR thresholds with the T_{60} = 2s condition having elicited higher TMR thresholds than the T_{60} = 1s (M_{diff} = 0.422 dB, 95% CI [-0.004, 0.847], SE = 0.191, p = .052, partial η^2 = 0.328). However, this effect cannot be considered a statistically significant increase in TMR identification thresholds. There was no significant interaction effect found for any of the three factors BRIR and spatial separation: [F (2, 20) = 2.092, p = 0.150, partial η^2 = .173]; BRIR condition and reverberation time: [F (1, 10) = .524, p = .600, partial η^2 = .050]; reverberation time and spatial condition, [F (1, 10) = .468, p = .510, partial η^2 = .045]; and BRIR condition, reverberation time, and spatial separation, [F (2, 20) = .130, p = .878, partial η^2 = .013].

An additional 2 x 3 repeated measures ANOVA was conducted with spatial separations (0° and 30°) and Binaural Room Impulse Response conditions (correct estimation, over estimation, and under estimation) as within-subjects factors to evaluate the effects of these variables on the two different reverberation times (i.e., T_{60} = 1s; T_{60} = 2s). There was a significant main effect indicated for the BRIR condition for a reverberation time of 1 second, [F (2, 20) = 4.083, p = .033, partial η^2 = .290], and for a reverberation time of 2 seconds, [F(2, 20) = 4.695, p = .021, partial η^2 = 0.329]. Conversely, no significant main effect was indicated for the spatial separation condition for a reverberation time of 1 second, [F (1, 10) = 4.467, p = .061, partial η^2 = .309], and for a reverberation time of 2 seconds [F (1, 10) = 1.352, p = .455, partial η^2 = .057]. These results indicated that the BRIR conditions affected the TMR identification thresholds differently. *Post hoc* analysis with Bonferroni adjustment revealed that there was no statistically significant difference between TMR thresholds between any of the

BRIR estimation conditions for the reverberant environment of 1 second, $p > .05$. Although this may be true, there was a notable difference in TMR identification thresholds between the under-estimation condition when compared to the correct estimation condition [M_{diff} = -0.685 dB, 95% CI [-1.422, .051 dB, $p = .07$], but this difference was not deemed statistically significant. In addition, there was no statistically significant difference between the remaining BRIR estimations, [Correct estimation vs. Over estimation: (M_{diff} = 0.069 dB, 95% CI [-0.721, 0.860 dB], *SE* = 0.275 $p > .05$); Over estimation vs. Under estimation: (M_{diff} = 0.616 dB, 95% CI [-.130 dB, 1.363 dB], *SE* = 0.260, $p > .05$).

The pairwise comparisons for the T_{60} = 2s condition revealed statistically significant differences between BRIR estimation conditions in which the under-estimation condition yielded statistically significantly lower TMR thresholds than the overestimation condition (M_{diff} = 0.935 dB, 95% CI [-1.864, -.007 dB], $p = .048$). There were no significant differences between the remaining BRIR estimations, [Correct estimation vs. Over estimation: (M_{diff} = -0.222 dB, 95% CI [-1.158 dB, 0.714 dB], *SE* = .326, $p > .05$); Correct estimation vs. Under estimation: (M_{diff} = 0.713 dB, 95% CI [-.168 dB, 1.594 dB], *SE* = 0.307, $p = .127$). Figure 3 depicts the mean values based on the TMR spatial release from masking in the two-different reverberant listening environments based on the three different BRIR estimations used to remove the effects of reverberation on the presented speech.

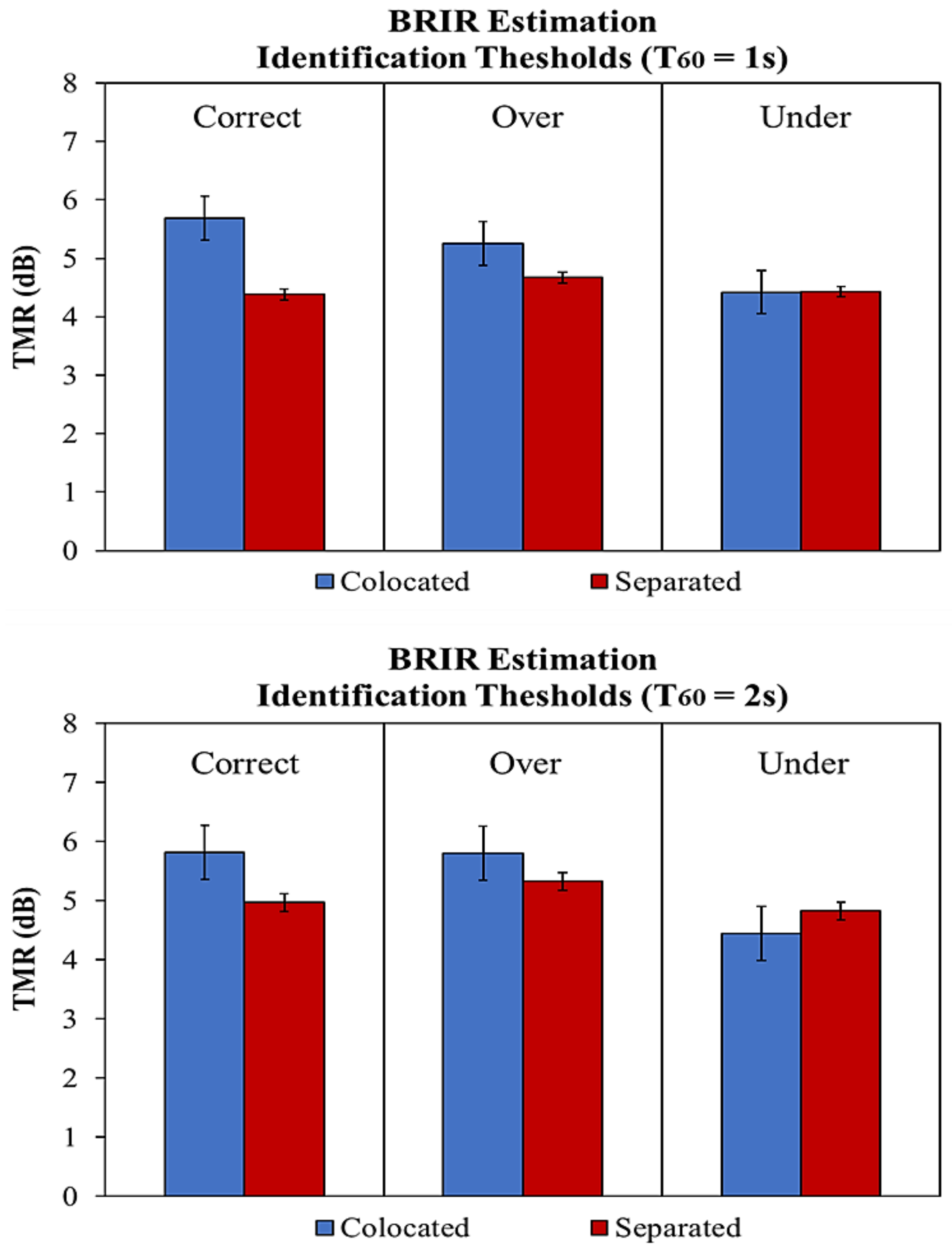

Figure 9. The top panel shows the average TMR identification thresholds for the three BRIR estimation conditions based on the reverberation listening condition of T60 = 1s. The bottom panel shows the average TMR identification thresholds for the three BRIR estimation conditions based on the reverberation listening condition of T60 = 2s. The left columns show the average TMR thresholds for the Correct Estimation of Binaural Room Impulse Response (BRIR) (de-reverberation T60 same as target BRIR's T60) for two different reverberation times (T60 = 1s and 2s). The center columns and right columns show the average TMR thresholds for the Over Estimation (de-reverberation T60 0.5s more target BRIR's T60) and Under Estimation (de-reverberation T60 is 0.5 less target BRIR's T60) conditions, respectively. Within each panel, the TMR identification thresholds obtained in the colocated condition are denoted by the red bars and the spatially separated condition by the blue bars. Error bars indicate ± 1 Standard error mean (SEM).

Table 4

Descriptive Statistics for Target-to-Masker (TMR) Identification Thresholds (in dB) Obtained in Three Different Binaural Room Impulse Estimations (BRIR), Two Reverberation Conditions (in seconds) and Two Different Spatial Conditions (Colocated and Spatially Separated)

Reverberation Time (T60)	Spatial Condition	Correct Estimation			Over Estimation			Under Estimation		
		n	*M*	*SD*	*n*	*M*	*SD*	*n*	*M*	*SD*
T60 = 1s	Colocated	11	**5.64**	0.96	11	**5.22**	1.05	11	**4.42**	1.04
	Separated	11	**4.37**	1.84	11	**4.66**	1.65	11	**4.23**	1.33
T60 = 2s	Colocated	11	**5.82**	1.43	11	**5.80**	1.25	11	**4.43**	1.07
	Separated	11	**4.87**	2.71	11	**5.33**	1.70	11	**4.83**	1.50

Note: Mean values are in boldface. *SD* = standard deviation.

There was no significant interaction effect found between BRIR and spatial separation for a reverberation time of 1 second, [F (2, 20) = 1.295, p = 0.296, partial η^2 = .115], and for a reverberation time of 2 seconds, [F (2, 20) = 1.352, p = .281, partial η^2 = .119]. Over vs. Under estimation: (M_{diff} = 0.616 dB, 95% CI [-.130 dB, 1.363 dB], *SE* = 0.260, p = .118)]. To clarify, there was a no statistically significant difference between spatial conditions for the correct estimation [Colocated vs. Spatially Separated: (M_{diff} = 0.958 dB, 95% CI [-.931 dB, 2.847 dB], *SE* = .285, p > .05)], over estimation [Colocated vs. Spatially Separated: (M_{diff} = 0.470 dB, 95% CI [-.940 dB, 1.80 dB], *SE* = 0.633, p > .05)], and for the under estimation condition [Colocated vs. Spatially Separated: (M_{diff} = -0.392 dB, 95% CI [-1.294 dB, .510 dB], *SE* = 0.405, p > .05)]. Figure 4 depicts the mean values based on the spatial release from masking in the two-different reverberant listening environments based on the three different BRIR estimations used to remove the effects of reverberation on the presented speech. Table 5 illustrates the descriptive statistics for spatial release from masking based on the two-different reverberant listening environments and the three different BRIR estimation conditions.

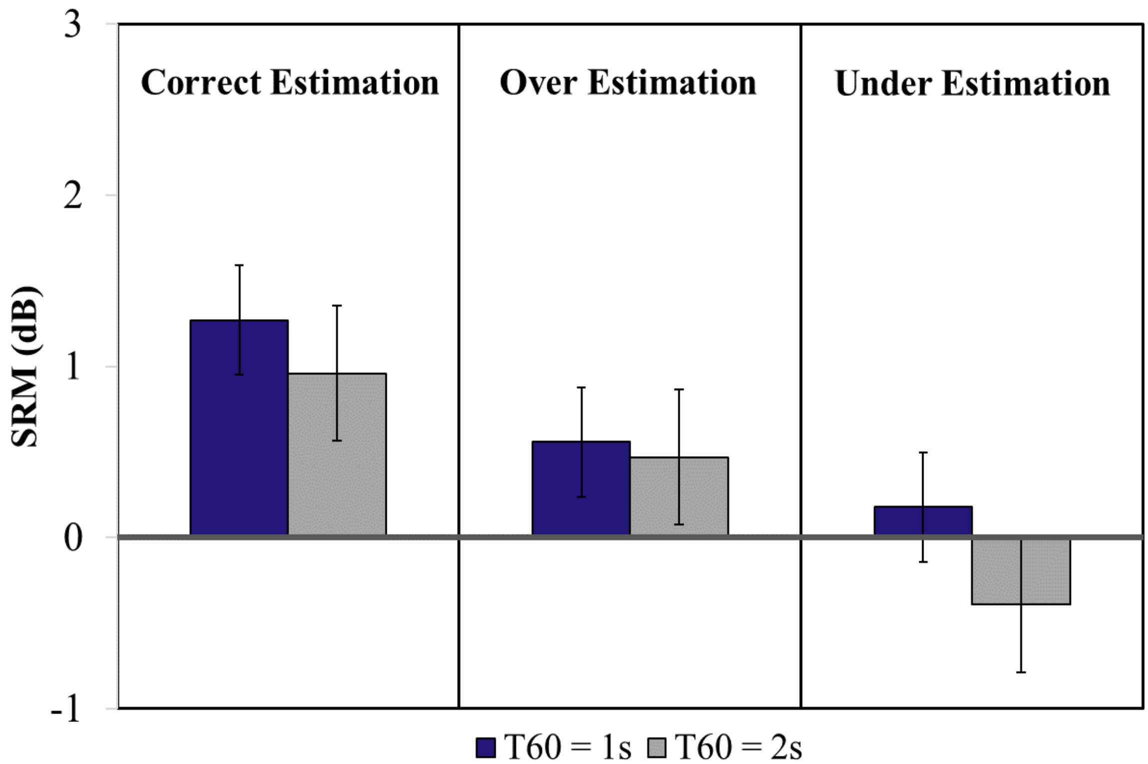

Figure 10. This figure shows the average SRM as a function three different BRIR conditions and two different reverberation times for individual listeners. The left column shows the average SRM for the Correct Estimation of Binaural Room Impulse Response (BRIR) (de-reverberation T60 same as target BRIR's T60) for two different reverberation times (T60 = 1s and 2s). The center column and right column show the average SRM for the Over Estimation (de-reverberation T60 0.5s more target BRIR's T60) and Under Estimation (de-reverberation T60 is 0.5 less target BRIR's T60) conditions, respectively. Within each panel, the calculated SMR values for the T60 = 1s condition are denoted by the blue bars and the T60 = 2s by the gray bars. Error bars represent ±1 SEM.

Table 5

Descriptive Statistics for Spatial Release from Masking (Colocated TMR Thresholds – Separated TMR Thresholds) (in dB) Based on Three Different Binaural Room Impulse Estimations (BRIR), and Two Reverberation Conditions (in seconds)

	Reverberation Time (T60)											
	T60 = 1s						T60 = 2s					
					Range						Range	
SRM (dB)	*n*	***M***	*SD*	*SE*	*Min*	*Max*	*n*	***M***	*SD*	*SE*	*Min*	*Max*
Correct	11	**1.27**	1.34	.41	-1.23	2.80	11	**0.96**	2.81	.85	-4.55	5.47
Over	11	**0.56**	1.97	.60	-2.07	3.54	11	**0.47**	2.01	.63	-3.06	4.90
Under	11	**0.18**	1.66	.50	-2.82	2.49	11	**-0.39**	1.34	.40	-2.97	1.40

Note: Mean values are in boldface. *SD* = standard deviation; *SE* = Standard error of mean; *Min* = minimum; *Max* = maximum.

The two correct estimation conditions utilized in the two-different reverberation listening environments were compared to determine if there was a significant difference between both conditions. Ideally, there should be no difference between each condition given that the estimation is the same as the reverberation time. A two-way repeated measures ANOVA was conducted to determine the effects of listening environment and spatial condition on TMR identification thresholds. There was a statistically significant main effect of listening environment on TMR identification thresholds, [F (2, 20) = 73.601, p < .001, partial η^2 = .880, and for spatial separation on TMR identification thresholds, [F (1, 10) = 8.156, p = .017, partial η^2 = .449. There was no significant interaction between listening environment and spatial separation, [F (2, 20) = 2.232, p = .133, partial η^2 = .182]. All simple pairwise comparisons were run between the different listening environments and spatial separations. A Bonferroni adjustment was applied. There was a statistically significant mean difference between: the anechoic listening condition and reverberation listening condition of T_{60} = 1s using a correct BRIR estimation (95% CI, [-5.164, -2.496], p < .001, SE = .461) and the anechoic listening condition and the reverberant listening condition of T_{60} = 2s using a correct BRIR estimation (95% CI, -5.250, -3.078, p < .001, SE = .378). There was no statistically significant mean difference between the T_{60} = 1s reverberant listening environment using correct estimation (M = 5.01, SE= 0.394) and the T_{60} = 2s reverberant listening environment using correct estimation (M = 5.345, SE = 0.359), p > .05. Pairwise comparisons with Bonferroni adjustments indicated a statistically significant difference between spatial separations with the spatially separated condition yielding significantly lower TMR identification thresholds than the colocated condition (M_{diff} = -1.529, 95% CI (-2.722, -.366 dB), p = .017, SE = 0.535). There was no significant interaction found between BRIR estimation and Spatial Condition for a Reverberation Time of 1 second, [F (2, 20) = 2.232, p = .133, partial η = .182].

Multiple Regression and Correlation Models Predicting SRM Based on Various Potential Predictors

To further investigate the trends suggested by the main effects of the BRIR estimation conditions, age and hearing loss were examined as continuous as opposed to categorical statistical variables. Figure 6 shows the relationship between age, PTA, and SRM (dB) for three BRIR estimation conditions in this study (i.e., correct estimation, overestimation, and underestimation). Hearing loss was quantified by the Pure Tone Average (PTA) of audiometric thresholds (in dB HL) for the octave frequencies 500, 1000, 2000, and 4000 Hz. Since all PTA averages between ears were within 5 dB HL, those values were averaged together to obtain an average PTA of both ears. Visual inspection of this scatterplot indicated a linear relationship between Spatial release from masking and Age only in the Under Estimated condition for 2 seconds of reverberation. Additionally, there was a linear relationship indicated between hearing loss (as indicated by PTA) and SRM in the correct estimated for 2 seconds of reverberation and the underestimated condition for one second of reverberation. In both cases, as hearing loss and/or age increased, the benefit (in dB) obtained from spatially separating (Target speaker located at 0° azimuth; Masker speakers located at ±30°azimuth) the target and masker signals decreased. In all other conditions the R^2 values are not significant to insinuate that there is an association between lower SRM values and the listeners PTAs and/or age. Correlations among age, PTA, and identification thresholds for the three different BRIR estimation conditions are indicated in the top right hand corner of each plot.

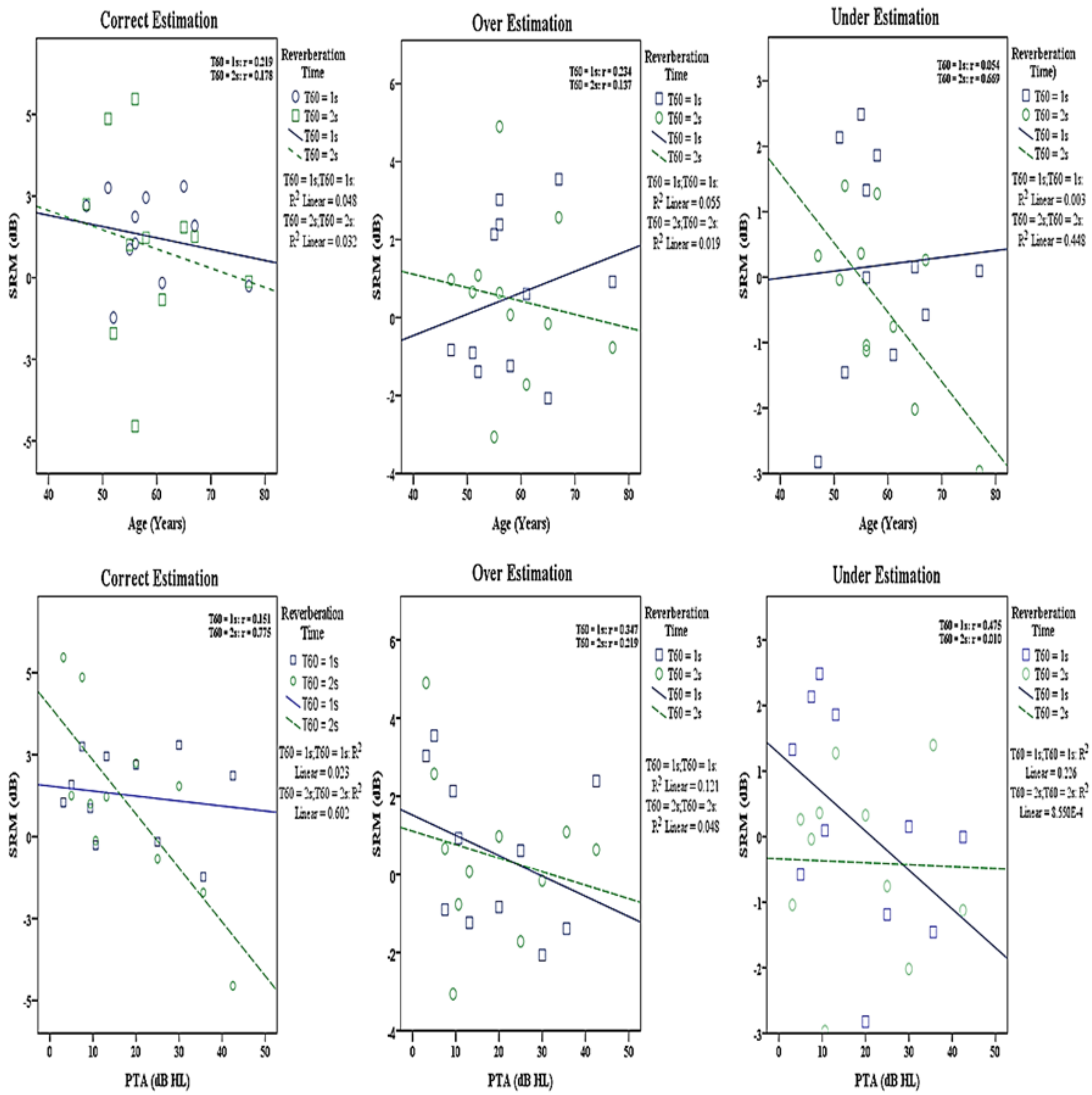

Figure 11. SRM for the three different BRIR estimation conditions plotted as a function of age (top row) and PTA (dB HL) (bottom row) for individual participants. Within each panel, the reverberation conditions of T60 = 1s are denoted by purple squares and T60 = 2s by green circles. Correlation values are indicated in the top right corner based on the two different reverberation times.

Correlation and multiple regression analyses were conducted to examine the effect of age and hearing loss on TMR identification thresholds and the SRM. Correlations between Spatially Separated TMR identification thresholds and PTA were statistically significant ($p < .05$) in all BRIR estimation conditions with correlation values ranging from .417 to .633. Correlations between Colocated TMR identification thresholds and PTA were statistically significant in the Under estimated BRIR condition [r (21) = 0.387, p = .038]. In addition, correlations between Age and Colocated TMR identification thresholds were statistically significant in the Overestimated condition [r (21) = 0.489, p = .010]. Finally, correlations between SRM and PTA were statistically significant in the Correct BRIR estimation condition [r (21) = -.539, p < .05]. Similarly, correlations between SRM and PTA were statistically significant in the Correct BRIR estimation condition [r (21) = 0.489, p = .010], in the T_{60} = 2s condition. Moreover, correlations between PTA and SRM were statistically significant in the correct estimated condition [r (8) = -.776, p = .002], in the T_{60} = 2s condition. Finally, correlations between age and SRM were statistically significant in the under estimated condition [r (8) = -0.670, p = .012]. All other correlations between the remaining predictor variables were not statistically significant, p > .05.

Table 6

Descriptive Statistics and Summary of Multiple Regression Analysis to Examine the Relationship Between Target-to-Masker Thresholds (as indicated by dB) and Spatial Release from Masking (as indicated by the difference between Colocated and Separated TMR in dB) and Various Potential Predictors

			Age		**Hearing Loss**	
Colocated TMR	*Mean (dB)*	*SD*	Correlation	*p* value	Correlation	*p* value
Correct Estimation	5.73	1.19	-.033	.443	.126	.288
Over Estimation	5.51	1.16	**.489**	**.010**	.118	.301
Under Estimation	4.42	1.02	-.264	.118	**.387**	**.038**
Separated TMR						
Correct Estimation	4.62	1.98	.176	.217	**.633**	**.001**
Over Estimation	5.00	1.66	.291	.095	**.417**	**.027**
Under Estimation	4.53	1.41	.087	.349	**.564**	**.003**
SRM						
Correct Estimation	1.11	2.15	-.179	.212	**-.539**	**.005**
Over Estimation	.517	1.98	.042	.426	-.281	.103
Under Estimation	-.103	1.50	-.263	.118	-.269	.113

Note. **Bolded** values indicate the correlations that are significant based on a *p* value < .05.

Table 7

Descriptive Statistics and Summary of Multiple Regression Analysis to Examine the Relationship Between Spatial Release from Masking (as indicated by the difference between Colocated and Separated TMR in dB) and Various Potential Predictors Based on Two Different Reverberation Times (T_{60} = 1s and T_{60} = 2s)

SRM			**Age**		**Hearing Loss**	
T60 = 1s	*Mean (dB)*	*SD*	Correlation	*p* value	Correlation	*p* value
Correct Estimation	1.26	1.34	-.218	.259	-.150	.330
Over Estimation	0.56	1.97	.235	.243	-.349	.147
Under Estimation	0.18	1.65	.054	.438	-.475	.070
T60 = 2s						
Correct Estimation	0.96	2.81	-.178	.301	**-.776**	**.002**
Over Estimation	0.47	2.10	-.139	.342	-.218	.260
Under Estimation	-0.39	1.34	**-.670**	**.012**	-.029	.446

Note. **Bolded** values indicate the correlations that are significant based on a *p* value < .05.

In order to separate the overall performance differences among the three different BRIR estimation conditions from the effects of reverberation on the presence and ability to use spatial cues, the colocated and spatially separated thresholds were subtracted to produce a calculation of SRM for the three BRIR conditions. SRM for the three BRIR estimation conditions in two different reverberant environments (T_{60} = 1s and T_{60} = 2s) as a function of age and PTA is shown in the left panel. The multiple regression model predicting SRM at 30 degrees was significant in the Correct estimated condition for a reverberation time of 2 seconds and accounted for 70.6% of the variance in SRM with R^2 = .706, [F (2, 8) = 9.614, p = .007]. Listeners with greater hearing loss (higher PTAs) tended to have less benefit obtained from spatially separating the target signal and masker signals, *r* (8) = -.776, *p* = .002.

The resulting regression model was: SMR = 3.988 + (-.109*Age) + (-.177*PTA), indicating that the SRM of the listener decreased by -.109 dB for every year in Age after keeping PTA constant as its mean. PTA did not statistically significantly predict SRM (as indicated by the difference between colocated thresholds and separated TMR identification thresholds) in the correct estimated condition for a reverberation time of 1 second, [F (2, 8) = .371, p = .701]. Age alone did not predict SRM in any of the estimation conditions, p > .05. Although the multiple regression model was not significant [F (2, 8) = 3.567, p =.078] Age statistically significantly predicted SRM for the underestimation for a reverberation time of 2 seconds. The resulting regression model was: SMR = 6.364 + (-.697*Age) + (-.016*PTA), indicating that the SRM of the listener decreased by -.697 dB for every year in Age after keeping PTA constant as its mean. Nonetheless, this multiple regression model was not statistically significant, p =.078. All other multiple regression analyses were not statistically significant in predicting SRM for the remaining BRIR estimations based on the two reverberation times, $p > .05$. These results suggest that hearing loss plays more of a significant role in speech understanding in reverberant environments when target signals and maskers are spatially separated for the correct estimation condition. The three tables below show the proportion of variance accounted for and standardized regression coefficients for the predictor variables (age and PTA) for the multiple regression analyses predicting colocated and spatially separated thresholds at three BRIR estimation conditions. The multiple regression model predicting spatially separated thresholds TMR are as follows:

Table 8

Multiple Regression Models Predicting SRM (as indicated by the difference between the Colocated and Separated Conditions in dB) between BRIR Estimation Conditions and Various Potential Predictors Based on Two Different Reverberant Listening Environments (T_{60} = 1s and T_{60} = 2s)

Reverberation Listening Environment (T_{60} = 1s)									
	Colocated			Separated			SRM		
BRIR Estimation	R^2	Age	PTA	R^2	Age	PTA	R^2	Age	PTA
Correct	**.546**	.115	**.751**	.298	.245	.534	.085	-.254	-.196
Over	.470	**.657**	.346	.357	.205	.599	.152	.178	-.316
Under	.010	-.057	.071	.433	-.003	**.657**	.227	-.033	-.481
Reverberation Listening Environment (T_{60} = 2s)									
	Colocated			Separated			SRM		
BRIR Estimation	R^2	Age	PTA	R^2	Age	PTA	R^2	Age	PTA
Correct	.082	-.094	-.288	**.807**	**.362**	**.825**	**.706**	-.328	**-.835**
Over	.202	.455	.115	.393	.562	.396	.080	-.184	-.251
Under	**.575**	-.339	**.620**	.399	.379	.578	.471	**-.697**	-.155

Note. Table 8 adapted from "The role of early and late reflections on spatial release from masking: Effects of age and hearing loss." by N. K. Srinivasan, 2017, *The Journal of the Acoustical Society of America, 140*, p. EL189. Copyright 2017 by the American Psychological Association. Significant models and their significant contributors are indicated in bold font.

CHAPTER 5

DISCUSSION

The experiments presented in this thesis investigated the effect of reverberation on spatial release from masking (SRM) for speech-on-speech masking. To perform this task, the listener must separate the target speech from the two competing masker speech signals. This study focused on the benefit of spatial information in reverberant environments by keeping other factors constant. Experiment 1 examined the effect of spatial separation on spatial release from masking (in three different listening situations (anechoic: T_{60} = 0s; reverberant environment: T_{60} = 1s: reverberant environment: T_{60} = 2s). Experiment 2 examined the effect of three dereverberation techniques (correct: de-reverberation T_{60} same as target BRIR's T_{60}; overestimation: de-reverberation T_{60} is 0.5s more than target BRIR's T_{60}, and underestimation: de-reverberation T_{60} is 0.5s less than target BRIR) on SRM in two different reverberant listening environments (T_{60} = 1s and T_{60} = 2s).

Experiment 1: Identification Thresholds in Anechoic (0s) and Reverberant Environments (1s & 2s)

One critical aspect of the automated test of spatial release from masking is evaluating the changes in performance as a function of test repetition (Jakien et al., 2017). Jakien et al. (2017), reported that the relative difference between the first run and subsequent runs averaged across participants for this program was 2.00 dB for the colocated condition and 2.55 dB for the spatially separated condition. This also held true for our experiment as the performance was consistent for this program, regardless of the number of times participants performed the test. Specifically, the difference between the four trial runs at the beginning (average of 2.33 dB) of the study, and the conclusion (average of 2.77 dB) of the study was not statistically significantly different. These results provided evidence that there was no learning effect associated with the identification task.

The results of experiment 1 demonstrated that the listener's ability to identify the target call-sign in a multi-talker environment improved with spatially separating the target speaker from the masker speakers. Listeners on average, achieved a 2.36 dB release from masking by spatially separating the target from the maskers in the anechoic listening condition. Consistent with previous research, when sources of masking are spatially separated from the signal of interest, a listener can take advantage of acoustic cues arising from the spatial separation (Brungart et al., 2001; Freyman et al., 1999; 2001; Gallun et al., 2013; Hawley, Litovsky, & Culling, 2004; Jakien et al., 2017; Marrone et al., 2008; Srinivasan et al., 2016; 2017). The more relevant acoustic cues include fundamental frequency (male vs. female), spectral fluctuations of multiple speech maskers (frequency and amplitude fluctuations), context, differences in the levels of the target and masker signal, and differences in the location and timing of the target and maskers in the environment (Alain, 2007; Bronkhorst & Plomp, 1992; Brungart et al., 2001). However, the magnitude of the benefit of spatial separation between target and the two masker speakers was smaller for older listeners with hearing loss when compared to previous research (Gallun et al., 2013; Jakien et al., 2017; Srinivasan et al., 2016).

According to Jakien and colleagues (2017), the calculated amount of SRM averaged across participants with various ages and hearing capabilities was roughly 6 dB for a spatial separation target speech signal (located at 0° azimuth) and two masker speech signals of (located at ±30° azimuth) in an anechoic listening environment.

Similarly, Gallun et al. (2013) indicated that when the target and masker speakers are similar genders (i.e., male talker, male maskers) mean TMR thresholds for the colocated conditions were 2.13 dB and -4.30 dB for the spatially separated condition for older listeners with hearing loss. In reference to Gallun et al. (2013) findings, the calculated binaural release from masking of 6.43 dB is quite similar to the SRM reported in the Jakein et al. (2017) study. These SRM values are better (i.e., higher) than the SRM obtained in this experiment. Nonetheless, a possible explanation for the findings presented in the Gallun et al. (2013) study is because a ±45° spatial separation of the target signal and masker signals were used, which may have improved performance for many of the hearing-impaired participants. Of important note, Jakien et al. (2017), indicated that the benefit of SRM for a ±45° spatial separation of target and masker speakers using headphone presentation, and ±30° spatial separation of target and speech maskers in an anechoic environment using sound field presentation was not statistically significantly different. Another important study for comparison to the current results is that of Srinivasan et al. (2016). In the conditions that are most relevant to the current experiment (two same-sex maskers either colocated with the target at 0 degrees or symmetrically separated at ±30 degrees), spatially separating the maskers led to an estimated 2.0 dB improvement in the TMR identification thresholds for the older hearing-impaired group. This trend in performance was similar to the findings of this study.

Another key point revealed in the data was that the mean SRM obtained in the anechoic condition reduced significantly as reverberation increased. In fact, for listeners with hearing loss, their performance when the target and masker speakers were spatially separated was no different than when the target and masker speakers were colocated.

Listeners with hearing loss showed a benefit of spatial separation of 1.16 dB in the reverberant listening condition of one second. Similarly, there was even less of a difference between colocated and spatially separated TMR identification thresholds for a reverberation time of two seconds. The average SRM obtained for older listeners with hearing loss was 0.13 dB in the reverberant listening condition of two seconds. Consistent with previous findings, reverberation significantly reduced the benefit obtained from spatially separating the target and masker signals for older listeners with hearing loss (Marrone et al., 2008; Srinivasan et al., 2017). Srinivasan and colleagues (2017) concluded that the older-hearing impaired listener group obtained an estimated average SRM of 1 dB when all reverberant reflections and when late reverberant reflections were present. In summary, the results presented in the previous literature indicate that there is a significant reduction in benefit obtained from binaural release from masking when late reverberant reflections are present for older listeners with hearing loss.

Experiment 2: Identification Thresholds in Two Different Reverberant Environments (1s & 2s) Based on Three Different BRIR Estimation (Correct, Over, and Under Estimation)

TMR identification thresholds from the participants in this study were analyzed in terms of 3 different conditions. These conditions include: two reverberation times (i.e., T_{60} = 1s, and T_{60} = 2s); two different spatial configurations (colocated and spatially separated), and three different dereverberation processing techniques (i.e., correct, over, and underestimation). Overall, the conditions were ordered: Under, Correct, and Overestimation (going from the lowest to the highest TMR threshold). Each of these different experimental conditions will be discussed separately below. Overall, the underestimation BRIR elicited the lowest TMR thresholds primarily in the colocated condition. More specifically, in the reverberant environment of one second, participants mean TMR thresholds in the underestimation condition (average of 4.42 dB) were significantly lower when compared to the correct estimation (average of 5.64 dB) and the overestimation condition (average of 5.22 dB). Thresholds in the separated condition were comparable between the three different BRIR estimations (correct estimation = 4.37 dB; overestimation = 4.66 dB; underestimation = 4.23 dB) for a reverberation time of one second.

However, when reverberation increased (i.e., T_{60} = 2s) the TMR identification thresholds were notably lower in the colocated condition for the underestimation when compared to the correct estimation and the overestimation conditions. Also, the overestimation BRIR yielded the highest TMR identification thresholds in the spatially separated condition for a reverberation time of 2 seconds. Specifically, the TMR identification thresholds for the overestimation in spatially separated condition were higher than the overestimation colocated and underestimation condition. There was a more substantial spatial release from masking in the correct estimation condition when compared to the overestimation condition and the underestimation condition for a reverberation time of 1s. In a similar trend, the correct estimation generated a larger SRM when compared to the overestimation condition and the underestimation condition.

Multiple regression analyses predicting the amount of SRM using age and PTA indicated that only PTA was significant in predicting SRM in the correct estimation condition in the reverberant condition of 2 seconds. However, age and hearing loss did not statistically significantly predict identification thresholds in the correct BRIR estimation condition for a reverberation time of 1s. To summarize, these effects indicate that the correct estimation was equally effective in both reverberant environments for all listeners, but that it was potentially not accurate enough to remove all reflections to simulate an anechoic listening condition.

The findings of this study gave added insight to the potential mechanisms that may contribute to the effect reverberation has on SRM indicated by other researchers (Freyman et al., 1999; 2001; Marrone et al., 2008; Srinivasan et al., 2017; Lavandier et al., 2007; 2008). Similar to previous research, the effect of age on spatial-processing ability is less statistically significant than the effect of hearing loss at a ±30° spatial separation between target and masker signals (Jakein et al., 2017; Marrone et al., 2008; Gallun et al., 2013; Glyde et al., 2013). Also, Glyde et al., (2013) concluded that there is no significant relationship between age and spatial processing abilities. In this study, we were unable to indisputably distinguish the effects of age and hearing loss, possibly due to issues of sample size. Another possibility is the variability of hearing status of the listeners used in this experiment. To clarify, the small sample in the current study included some participants with a mild to moderately-severe hearing loss and other participants with only a mild high-frequency hearing loss. The general trend in this study was that thresholds varied as age increased. Additionally, our results suggest that while there is no significant difference in SRM found in the underestimation condition, performance in the underestimation condition was significantly better in the colocated condition when compared to the other conditions. More specifically, participants TMR thresholds were essentially 1.388 dB and 1.366 dB lower than the correct and overestimated conditions, respectively. Of important note, Correct estimation yielded a significantly greater SRM by 1.11 dB when compared to the overestimation (average of 0.56 dB) and the underestimation condition (average of 0.18 dB)

Perhaps the effect observed in this study could be attributed to the early reflections increasing the amplitude of the target signal. To explain, it is possible that the underestimation BRIR condition maintained early reflections that increased the amplitude of the target signal and therefore improved speech recognition towards the target located. Overall the poorer performance in the spatially separated condition could be attributed to the decrease in interaural coherence of the target and masker signals (Lavandier et al., 2007; 2008).

It was theorized that spatial separation of the direct target speech from the later arriving reflections would substantially reduce the masking effects contributed to the late reflections. However, in this study, it is possible that the separation of the masker speech signals created additional later sound reflections that degraded the interaural coherence of the target signal, which negatively impacted the spatial release from masking, as suggested by previous researchers (Lavandier & Culling, 2007, 2008). Similarly, Freyman et al. (1999; 2001) indicated that the reduction in speech intelligibility is also dependent upon the reflection direction, time delay, and type of interferer. The results found in this study were consistent with previous studies, with decreased effects of SRM ranging from 2-5 dB (Freyman et al., 1999; 2001; Marrone et al., 2008; Srinivasan et al., 2017).

Overall, underestimation elicited the best TMR thresholds primarily in the colocated condition in this study. Multiple regression analyses of the amount of SRM using age and PTA indicated that PTA was a significant predictor predicting SRM in the correct estimation conditions. All other models were not significant in predicting SRM. This result suggests that PTA plays a role in speech intelligibility in listening environments when the later reflections are removed from the signal.

CHAPTER 6

CONCLUSION

Reverberation times in everyday listening environments vary widely from nearly a few hundred milliseconds up to several seconds based on the room size and absorption properties of the materials present in the environment. It is also evident in the literature that speech understanding is reduced significantly for older listeners with hearing loss in reverberant listening environments. With that being said, it is surprising that even with the availability of de-reverberation processing strategies, most hearing manufacturers do not implement these processing strategies in their devices. With more research and the advancement of hearing aid technology, it is possible that more complex signal processing techniques will develop with greater accuracy in reducing the effects of reverberation. Therefore, adding an evidenced-based dereverberation program as an option for hearing aid users could improve speech understanding and reduce listening effort in reverberant environments. Dereverberation algorithms implemented in hearing aids would give the user an additional option to increase their ability to improve speech understanding in adverse listening conditions. Consequently, this leads to the question of how hearing aid technology will incorporate de-reverberation algorithms to improve speech understanding in environments with reverberation.

Limitations of the study conducted could potentially be the environment in which the study was conducted. A soundproof booth (i.e., simulated reverberant environment) in many ways does not accurately reflect the typical characteristics of reverberation found in everyday listening situations. Additionally, a larger sample size could tease out the difference of age and hearing loss on predicting SRM. These results indicate that more evidence is needed to determine the individual impact that aging has on speech understanding when removing later reverberation reflections from the speech signal in a multi-talker environment. Future experiments could be performed to evaluate the effect of these BRIR estimations with various spatial separations. Additionally, this study utilized non-individualized head-related transfer functions to spatially render the target signal and masker signals for each ear. Although, this simulation method has found to be accurate enough to reproduce BRIRs measured in real rooms, this method can still impact the spatial release from masking obtained for each individual listener (Zahorik, 2009). Likewise, the same BRIR estimation was used for both the target and the masker signals. On a positive note, these results are promising in several ways. First, the improvements were observed for most participants in this study utilizing the underestimation BRIR. This solution could reduce the negative effects of reverberation on speech and promote easier listening in complex adverse listening environments. Furthermore, this solution could provide a more comfortable hearing experience in a variety of environments with reverberation. It is important to realize that in reality audiologists can precisely approximate the physical and perceptual characteristics measured in a real room, however it is a cumbersome process. For that reason, it can prove difficult to use a correct BRIR estimation strategy to remove the late reverberant energy which can be detrimental for speech understanding. For this reason, if listeners used an underestimation processing strategy in hearing aids, speech intelligibility may be improved in reverberant environments.

To conclude, the ideal processing technique would be a correct BRIR estimation strategy. Underestimation is more effective in improving speech understanding in reverberant environments than an overestimation BRIR strategy. Nonetheless, getting the correct estimation of reverberation time is difficult. Hence, if hearing aid manufacturers implement a dereverberation algorithm for hearing aids, underestimating the T_{60} would improve speech intelligibility more so than a overestimating the T_{60}. To conclude, further research is needed in reverberant spaces to investigate this processing technique.

References

Alain, C. (2007). Breaking the wave: Effects of attention and learning on concurrent sound perception. *Hearing Research*, *229*(1–2), 225–236. doi:10.1016/j.heares.2007.01.011

American National Standards Institute (2004). American national standard specification for audiometers. ANSI 3.6-2004

Arai, T. (2017). *Improving speech intelligibility in reverberant environments*. [web page]. Retrieved from http://www.splab.net/sok_en.htm

Arbogast, T. L., Mason, C. R., & Kidd, G. (2002). The effect of spatial separation on informational and energetic masking of speech. *The Journal of the Acoustical Society of America, 112*(5), 2086–2098. doi:10.1121/1.1510141

Arbogast, T. L., Mason, C. R., & Kidd, G. (2005). The effect of spatial separation on informational masking of speech in normal-hearing and hearing-impaired listeners. *The Journal of the Acoustical Society of America, 117*(4), 2169–2180. doi:10.1121/1.1861598

American Speech-Language-Hearing Association (ASHA). (2005). *Central Auditory Processing Disorders: The role of the audiologist*. Retrieved from http://www.phon.ucl.ac.uk/courses/spsci/audper/ASHA%202005%20CAPD%20statement.pdf

Boothroyd, A. (2004). Room acoustics and speech perception. *Seminars in Hearing, 25*(2), 155–166.

Bradley, J. S., Sato, H., & Picard, M. (2003). On the importance of early reflections for speech in rooms. *The Journal of the Acoustical Society of America, 113*(6), 3233. doi:10.1121/1.1570439

Bregman, A. S. (1990). *Auditory scene analysis: The perceptual organization of sound*. Cambridge, MA: The MIT Press.

Breitsprecher, C. (2011). *Effects of reverberation on speech intelligibility in normal-hearing and hearing-impaired listeners* (Master's thesis, Technical University of Denmark, Kongens Lyngby, Denmark). Retrieved from http://www.eriksholm.com/Research%20Areas/Advanced-Algorithms/spatial-hearing-and-spatial-processing/the-effect-of-reverberation-on-speech-recognition.aspx

Bremen, P., & Middlebrooks, J. C. (2013). Weighting of spatial and spectro-temporal cues for auditory scene analysis by human listeners. *PLoS ONE, 8*(3), 1–13. doi:10.1371/journal.pone.0059815

Bronkhorst, A. (2000). The cocktail party phenomenon: A review of research on speech intelligibility in multiple-talker conditions. *Acta Acoustica, 86*(1), 117–128.

Bronkhorst, A. (2015). The cocktail-party problem revisited: Early processing and selection of multi-talker speech. *Journal of Attention, Perception, and Psychophysics, 77*(5), 1465-1487. doi:10.3758/s13414-015-0882-9

Bronkhorst, A. W., & Plomp, R. (1988). The Effect of Head-Induced Interaural Time and Level Differences on Speech-Intelligibility in Noise. *The Journal of the Acoustical Society of America, 83*(4), 1508–1516. doi:10.1121/1.395906

Bronkhorst, A. W., & Plomp, R. (1992). Effect of multiple speechlike maskers on binaural speech recognition in normal and impaired hearing. *The Journal of the Acoustical Society of America, 92*(6), 3132–3139. doi:10.1121/1.404209

Brungart, D. S., Simpson, B. D., Darwin, C. J., Arbogast, T. L., & Kidd, G. (2005). Across-ear interference from parametrically degraded synthetic speech signals in a dichotic cocktail-party listening task. *The Journal of the Acoustical Society of America, 117*(1), 292–304. doi:10.1121/1.1835509

Brungart, D. S., Simpson, B. D., Ericson, M. A., & Scott, K. R. (2001). Informational and energetic masking effects in the perception of multiple simultaneous talkers. *The Journal of the Acoustical Society of America, 110*(5), 2527–2538. doi:10.1121/1.1408946

Carlile, S. (2014). Active Listening: Speech Intelligibility in Noisy Environments. *Acoustics Australia*, *42*(2), 90–96.

Cherry, E. C. (1953). Some experiments on the recognition of speech, with one and with two ears. *The Journal of the Acoustical Society of America, 25*(5), 975–979. doi:10.1121/1.1907229

Durlach, N. I., Mason, C. R., Shinn-Cunningham, B. G., Arbogast, T. L., Colburn, H. S., & Kidd, G. (2003). Informational masking: Counteracting the effects of stimulus uncertainty by decreasing target-masker similarity. *The Journal of the Acoustical Society of America*, *114*(1), 368–379. doi:10.1121/1.1577562

Eggermont, J. (2013). Speech understanding in noise. In J. Eggermont (Ed.), *Noise and the brain: Experience dependent development and adult plasticity* (pp. 155–179). San Diego, CA: Elsevier.

Emanuel, D., Maroonroge, S., & Letowski, T. R. (2009). Auditory Function. In C. Rash (Ed.), Helmet-mounted displays: sensation, perception, and cognition issues (pp. 307–332). Fort Rucker, AL.

Folstein, M. F., Folstein S. E., & McHugh, P. R. (1975). Mini-Mental State. A practical method for grading the cognitive state of patients for the clinician. *Journal of Psychiatric Research*, *12*, 189-198

Fletcher, H. 1940. Auditory patterns. *Reviews of Modern Physics. 12*, 47–65. Retrieved from http://journals.aps.org.proxy-tu.researchport.umd.edu/rmp/pdf/10.1103/RevModPhys.12.47

Freyman, R. L., Balakrishnan, U., & Helfer, K. S. (2001). Spatial release from informational masking in speech recognition. *The Journal of the Acoustical Society of America*, *109*(5), 2112–2122. doi:10.1121/1.1354984

Freyman, R. L., Helfer, K. S., McCall, D. D., & Clifton, R. K. (1999). The role of perceived spatial separation in the unmasking of speech. *The Journal of the Acoustical Society of America*, *106*(6), 3578–3588. doi:10.1121/1.428211

Gallun, F. J. (2017). *Developing portable, automated, rapid test of auditory processing ability* [PowerPoint slides]. Retrieved from http://www.ncrar.research.va.gov/aboutus/staff/gallun.asp

Gallun, F. J., Diedesch, A. C., Kampel, S. D., & Jakien, K. M. (2013). Independent impacts of age and hearing loss on spatial release in a complex auditory environment. *Frontiers in Neuroscience*, *7*(252), 1–11. doi:10.3389/fnins.2013.00252

Gatehouse, S., & Akeroyd, M. (2006). Two-eared listening in dynamic situations. *International Journal of Audiology*, *45*(1), 120–124. doi:10.1080/14992020600783103

Gelfand, S. A. (2010). Hearing: An introduction to psychological and physiological acoustics (5th ed.). New York, NY: Informa Healthcare.

Glyde, H., Cameron, S., Dillon, H., Hickson, L., & Seeto, M. (2013). The Effects of Hearing Impairment and Aging on Spatial Processing. *Ear and Hearing, 34*(1), 15–28. doi:10.1097/AUD.0b013e3182617f94

Hawley, M. L., Litovsky, R. Y., & Culling, J. F. (2004). The benefit of binaural hearing in a cocktail party: Effect of location and type of interferer. *The Journal of the Acoustical Society of America, 115*(2), 833–843. doi:10.1121/1.1639908

Hearing Loss Association of America. (2017). Basic facts about hearing loss [webpage]. Retrieved from http://hearingloss.org/content/basic-facts-about-hearing-loss

Helfer, K. S., & Freyman, R. L. (2008). Aging and speech-on-speech masking. *Ear and Hearing, 29*(1), 87–98. doi:10.1097/AUD.0b013e31815d638b

Helfer, K. S., & Wilber, L. A. (1990). Hearing loss, aging, and speech perception in reverberation and noise. *Journal of Speech, Language, and Hearing Research, 33*, 149-155. doi:10.1044/jshr.3301.149

Helfer, K. (1992). Aging and the binaural advantage in reverberation and noise. *Journal of Speech and Hearing Research, 37*, 1394-1401.

Hirsh, I. J. (1950). The relation between localization and intelligibility. *The Journal of the Acoustical Society of America, 22*(2), 196–200. doi:10.1121/1.1906588

Howarth, A., & Shone, G. R. (2006). Ageing and the auditory system. *Postgraduate Medical Journal, 82*(965), 166–171. doi:10.1136/pgmj.2005.039388

Hunter, L., & Sanford, C. (2015). Tympanometry and wideband acoustic immittance. In J. Katz, K. English, L. Hood, & K. Tillery (Eds.), Handbook of Clinical Audiology (pp. 137-164). Philadelphia, PA: Wolters Kluwer.

Hudspeth, A. J. (2005). How the ear's works work: Mechanical transduction and amplification of hair cells. *Journal of Neurosciences*, *328*, 155-162. doi:10.1016/j.crvi.2004.12.003

Ihlefeld, A., & Shinn-Cunningham, B. (2008). Spatial release from energetic and informational masking in a selective speech identification task. *The Journal of the Acoustical Society of America*, *123*(6), 4369–4379. doi:10.1121/1.2904826

Jakien, K. M., Kampel, S. D., Stansell, M. M., & Gallun, F. J. (2017). Validating a rapid, automated test of spatial release from masking. *American Journal of Audiology*, *26*(4), 507-518. doi:10.1044/2017_AJA-17-0013

Jones, G. L., & Litovsky, R. Y. (2011). A cocktail party model of spatial release from masking by both noise and speech interferers. *The Journal of the Acoustical Society of America*, *130*(3), 1463–74. doi:10.1121/1.3613928

Kidd, G., Mason, C. R., Deliwala, P. S., Woods, W. S., & Colburn, H. S. (1994). Reducing informational masking by sound segregation. *The Journal of the Acoustical Society of America*, *95*(6), 3475–3480. doi:10.1121/1.410023

Kidd, G., Mason, C. R., Rohtla, T. L., & Deliwala, P. S. (1998). Release from masking due to spatial separation of sources in the identification of nonspeech auditory patterns. *The Journal of the Acoustical Society of America*, *104*(1), 422–431. doi:10.1121/1.423246

Lavandier, M., & Culling, J. F. (2007). Speech segregation in rooms: Effects of reverberation on both target and interferer. *The Journal of the Acoustical Society of America*, *122*(3), 1713–1723. doi:10.1121/1.2764469

Lavandier, M., & Culling, J. F. (2008). Speech segregation in rooms: monaural, binaural, and interacting effects of reverberation on target and interferer. *The Journal of the Acoustical Society of America, 123*(4), 2237–2248. doi:10.1121/1.2871943

Levitt, H. (1971). Transformed Up-Down Methods in Psychoacoustics. *Journal of The Acoustical Society of America, 49*(2B), 467. doi:10.1121/1.1912375

Litovsky, R. Y., Colburn, H. S., Yost, W. A. & Guzman, S. J. (1999). The precedence effect. *Journal of the Acoustical Society of America, 106*, 1633-1654.

Marrone, N., Mason, C. R., & Kidd, G. (2008). The effects of hearing loss and age on the benefit of spatial separation between multiple talkers in reverberant rooms. *The Journal of the Acoustical Society of America, 124*(5), 3064–3075. doi:10.1121/1.2980441

Mason, T. (2011). *Importance of Binaural Hearing* [PowerPoint slides]. Retrieved from https://www.ihsinfo.org/IhsV2/Convention2011/Seminars_PDF/The%20Importance%20of%20Binaural%20Hearing.pdf

Nábělek, A. K., & Robinette, L. (1978). Influence of the precedence effect on word identification by normally hearing and hearing-impaired subjects. *The Journal of the Acoustical Society of America, 63*(1), 187–194. doi:10.1121/1.381711

Sandel, T. T., Teas, D. C., Feddersen, W. E., Jeffress, L. A. (1955). Localization of sound from single and paired sources. *Journal of the Acoustical Society of America, 27*, 842-852.

Shaw, E. G. (1974). Transformation of sound pressure level from the free field to the eardrum in the horizontal plane. *Journal of the Acoustical Society of America, 56*, 1848-1861.

Shinn-Cunningham, B. (2013). Auditory Precedence Effect. In D. Jaeger & R. Jung (Eds.), Encyclopedia of Computational Neuroscience (pp. 1–3). New York, NY: Springer New York. doi.org/10.1007/978-1-4614-7320-6_101-5

Srinivasan, N. K., Jakien, K. M., & Gallun, F. J. (2016). Release from masking for small spatial separations: Effects of age and hearing loss. *The Journal of the Acoustical Society of America*, *140*(1), EL73-EL78. doi:10.1121/1.4954386

Srinivasan, N. K., Stansell, M., & Gallun, F. J. (2017). The role of early and late reflections on spatial release from masking: Effects of age and hearing loss. *The Journal of the Acoustical Society of America*, *141*(3), EL185-EL191. doi:10.1121/1.4973837

Stecker, G. C., & Gallun, F. J. (2012). Binaural hearing, sound localization, and spatial hearing. In K. Tremblay & R. Burkhard (Eds.), Translational perspectives in auditory neuroscience: Normal aspects of hearing (pp. 383–433). San Diego, CA: Plural Publishing, Inc.

Stevens, S. S., & Newman, E. B. (1936). The localization of actual sources of sound. *The American Journal of Psychology*, *48*(2), 297–306. doi:10.2307/1415748

Sudirga, R. E. (2014). Effect of Reverberation Context on Spatial Hearing Performance of Normally Hearing Listeners. The University of Western Ontario.

Tariq, S. H., Tumosa, N., Chibnall, J. T., Perry, M. H., & Morley, J. E. (2006). Comparison of the saint louis university mental status examination and the mini mental state examination for detecting dementia and mild neurocognitive disorder-pilot study. *The American Journal of Geriatric Psychiatry*, *14*(11), 900-910.

Taylor, B., & Mueller, G. H. (2011). Basic anatomy and physiology of the ear. In B. Stach (Ed.), *Fitting and dispensing hearing aids* (pp. 45-70). San Diego, CA: Plural Publishing.

Tun, P. A., Williams, V. A., Small, B. J., & Hafter, E. R. (2012). The Effects of Aging on Auditory Processing and Cognition. *American Journal of Audiology, 21*(2), 344–350. doi:10.1044/1059-0889(2012/12-0030)

Warzybok, A., Rennies, J., Brand, T., Doclo, S., & Kollmeier, B. (2013). Effects of spatial and temporal integration of a single early reflection on speech intelligibility. *Journal of the Acoustical Society of America, 133*(1), 269–282. doi:10.1121/1.4768880

Watson, C. S. (2005). Some comments on informational masking. *Acta Acoustica, 91*(3), 502–512.

Weinstein, B. E. (2000). Aging of the outer, middle, and inner ear, and neural pathways. In A. Seils (Ed.), *Geriatric Audiology* (pp. 55-80). New York, NY: Thieme Medical Publishers.

Wightman, F. L., & Kistler, D. J. (1997). Monaural sound localization revisited. *Journal of Acoustical Society of America, 101*(2), 1050-1063. doi:10.1121/1.418029

Zahorik, P. (2009). Perceptually relevant parameters for virtual listening simulation of small room acoustics. *The Journal of the Acoustical Society of America, 126*(2), 776–791. doi:10.1121/1.3167842

Appendix A

IRB <irb@towson.edu> Jan 22

to me, Nirmal

The IRB has approved your protocol “How Accurate Dereverberation Should be? The Effects of Underestimating and Overestimating Binaural Room Impulse Response” **as expedited, effective 1/22/2018 and expiring 1/21/2019.**

Your IRB protocol can now be viewed in MyOSPR. **Student investigators: protocols can be viewed by your faculty advisor.** For more information, please visit: http://www.towson.edu/academics/research/sponsored/myospr.html

Please Note: *Formal approval letters are now provided upon request. If you would like to have one drafted, please notify the IRB staff.*

If you should encounter any new risks, reactions, or injuries to subjects while conducting your research, please notify IRB@towson.edu. If your research has been approved as expedited and will extend beyond one year in duration, you will need to submit an annual renewal notice. Should there be substantive changes in your research protocol, you will need to submit another application.

Regards,
Towson IRB

Appendix B

INFORMED CONSENT FORM

SUBJECT ID#: ___________

PRINCIPAL INVESTIGATOR: Michael DiSanti   EMAIL: mdisan1@students.towson.edu

Purpose of the Study:

The proposed study is designed to evaluate a dereverberation processing strategy based on estimating the reverberation times of listening environments which are representative of real rooms in which the vast majority of listening takes place (i.e., small office spaces). Dereverberation is the process of removing sound reflections and sound echoes in a listening environment to increase speech intelligibility and understanding. The purpose of the proposed research is to aid in the development of dereverberation processing techniques for hearing aids which may help improve speech intelligibility in reverberant listening environments for all populations with hearing loss.

Procedures:

Participants will be administered an audiometric screening which includes: otoscopy, tympanometry, pure tone air and bone conduction at Towson University Van Bokkelen Hall. Then, participants will be given an audiology probe questionnaire and the Veterans Affairs Saint Louis University Mental Status Examination) to evaluate cognitive function. Next, the audio recordings of the Coordinate Response Measure will be presented over headphones and performance will be measured in twelve conditions. The target sentences the participant will repeat involves identifying a color number pair followed by the target call sign "Charlie". For example, "Ready [CALL SIGN] go to [COLOR] [NUMBER] now. The target sentences will be fixed in volume while the competing speech interferes will be varied in volume in order to obtain the speech reception threshold (level required to understand the target phrase 50% of the time). The simultaneous competing speech sentences will have different call signs and color number combinations.

Risks/Discomfort:

The proposed research involves no more risk than what is associated with daily life. However, there is a potential for boredom and fatigue. This will be discussed prior to the beginning of the study and you will be informed that breaks are encouraged as needed.

Benefits:

It is hoped that the results of this study will aid in the development of dereverberation processing techniques for hearing aids which may help improve speech intelligibility in reverberant listening environments for all populations with hearing loss.

Alternatives to Participation:

Participation in this study is voluntary. You are free to withdraw or discontinue participation at any time. Refusal to participate in this study will in no way affect your standing with Towson University, the Kaplan Hearing Center, or the services received by patient at each location.

Cost Compensation:

Participation in this study will involve no costs or payments to you. Participants will be compensated for their time with a $30.00 Target gift card.

Confidentiality:

All information collected during the study period will be kept strictly confidential. You will be identified through identification numbers. No publications or reports from this project will include identifying information on any participant. You will be instructed to your assigned identification number on all of the completed forms. If you agree to join this study, please sign your name below.

_____ I have read and understood the information on this form.

_____ I have had the information on this form explained to me.

____ ___________________

Subject's Signature Date

____ ___________________

Witness to Consent Procedures Date

____ ___________________

Principal Investigator Date

If you have any questions about the project, you may contact me at mdisan1@students.towson.edu or (443)-867-6376, my faculty advisor of the Audiology Department, Dr. Nirmal Srinivasan at nsrinivasan@towson.edu or the Chairperson of Towson University's Institutional Review Board for the Protection of Human Participants, Dr. Elizabeth Katz, at (410) 704-2236.

THIS PROJECT HAS BEEN REVIEWED BY THE INSTITUTIONAL REVIEW BOARD FOR THE PROTECTION OF HUMAN PARTICIPANTS AT TOWSON UNIVERSITY.

**If investigator is not the person who will witness participant's signature, then the person administering the informed consent should write his/her name and title on the "witness" line.

Appendix C

TOWSON UNIVERSITY

PARTICIPANTS NEEDED FOR A HEARING RESEARCH STUDY!

Who: Adults at risk of hearing loss or living with hearing loss between the ages of 30 and 80

Where: Towson University Van Bokkelen Hall
8000 York Road Towson, MD 21252

Time: ~2.5 hours

What does this study involve?

- 5-10 minute hearing screening to determine eligibility to participate
- A 2.5-hour session at Towson University Van Bokkelen Hall, where participants will repeat sentences presented in background noise and in reverberation

Goal: This study aims to examine processes involved in improving speech understanding in adverse listening environments

We will administer a brief case history with you and discuss whether you are able to participate. All queries are confidential

Compensation will be provided ($30.00 Gift Card for completion of entire study)!

To participate, please contact Michael:
(443)-867-6376
mdisan1@students.towson.edu

Appendix D

Towson University Speech, Language, and Hearing Center
Institute for Well-Being
1 Olympic Place
Towson, MD 21252
Voice or TTY: 410-704-3095

SUBJECT ID#: ___________

<u>Audiology Probe Questionnaire</u>

1. **Hearing Loss:**
   a. Duration ______________________ Was it gradual/sudden/fluctuating
   b. Which ear(s): right/left/both

2. **Symptoms:**
   a. Otalgia(pain) Y/N r/l/b
   b. Aural fullness Y/N r/l/b
   c. Vertigo/Disequilibrium Y/N
   d. Discharge Y/N r/l/b
   e. Tinnitus Y/N r/l/b (ringing/hissing/buzzing/roaring)

3. **History:**
   a. Hx of Otitis Media (ear infections)? If so how often (i.e., chronic)
   ____
   b. Family HX of hearing loss? ____________________________________
   c. Medical HX: (illnesses/diseases/trauma/surgery) ___________________________
   ____
   d. Were you ever exposed to noise (i.e., work, military, gunshots, fireworks, etc.) Y/N. If so for how long and did you wear hearing protection? ______________________________

4. **Amplification:** Current/previous user? Y/N Make/model/ear
   ____

OBSERVATIONS

5. **Otoscopic examination:** ____

Appendix E

VAMC
SLUMS Examination

Questions about this assessment tool? E-mail aging@slu.edu.

Name____________________ Age ____________________

Is patient alert?____________________ Level of education ____________________

__/1 ❶ **1. What day of the week is it?**

__/1 ❶ **2. What is the year?**

__/1 ❶ **3. What state are we in?**

4. Please remember these five objects. I will ask you what they are later.

Apple Pen Tie House Car

5. You have $100 and you go to the store and buy a dozen apples for $3 and a tricycle for $20.

❶ **How much did you spend?**

__/3 ❷ **How much do you have left?**

6. Please name as many animals as you can in one minute.

__/3 ⓿ 0-4 animals ❶ 5-9 animals ❷ 10-14 animals ❸ 15+ animals

__/5 **7. What were the five objects I asked you to remember?** 1 point for each one correct.

8. I am going to give you a series of numbers and I would like you to give them to me backwards. For example, if I say 42, you would say 24.

__/2 ⓿ 87 ❶ 649 ❶ 8537

9. This is a clock face. Please put in the hour markers and the time at ten minutes to eleven o'clock.

❷ Hour markers okay

__/4 ❷ Time correct

__/2 ❶ **10. Please place an X in the triangle.**

❶ **Which of the above figures is largest?**

11. I am going to tell you a story. Please listen carefully because afterwards, I'm going to ask you some questions about it.

Jill was a very successful stockbroker. She made a lot of money on the stock market. She then met Jack, a devastatingly handsome man. She married him and had three children. They lived in Chicago. She then stopped work and stayed at home to bring up her children. When they were teenagers, she went back to work. She and Jack lived happily ever after.

❷ **What was the female's name?** ❷ **What work did she do?**

__/8 ❷ **When did she go back to work?** ❷ **What state did she live in?**

______ **TOTAL SCORE**

Department of Veterans Affairs

SAINT LOUIS UNIVERSITY

SCORING

High School Education		Less than High School Education
27-30	Normal	25-30
21-26	MNCD*	20-24
1-20	Dementia	1-19

* Mild Neurocognitive Disorder

SH Tariq, N Tumosa, JT Chibnall, HM Perry III, and JE Morley. The Saint Louis University Mental Status (SLUMS) Examination for Detecting Mild Cognitive Impairment and Dementia is more sensitive than the Mini-Mental Status Examination (MMSE) - A pilot study. Am J Geriatr Psychiatry 14:900-910, 2006.

Appendix F

Towson University
Graduate Student Association Award
Incentive Acknowledgement Form

This form must be filled out as part of your miscellaneous expense voucher for your research if offering gift card payment to subjects. Please keep copies of each participant's form for your records as well.

Project Information:

Title: ____

Researcher name and TU ID #: ______________________________

Participant Contact Information:

Full name: ____

E-mail:____

Amount Paid: ___________

Payment method:

Gift card ONLY Gift card #: _____________________________

_____________________________ _______________________

Participant Signature Date

CURRICULUM VITAE

MICHAEL DISANTI

[redacted]

Education

Towson University - Doctor of Audiology, Au.D. *Anticipated May 2019*

- GPA: 3.66
- Graduate Student Association Research Award

Towson University – Bachelor of Science in Speech-Language Pathology and Audiology *May 2015*

- GPA: 3.68

Clinical Practicum Experience

Nemours Alfred I. duPont Hospital for Children – Wilmington, DE
Graduate Student Intern

- Provided comprehensive pediatric audiologic diagnostic evaluations, which included: Conventional audiometry, visual reinforcement audiometry, conditioned play audiometry, and two-tester paradigm audiometry
- Performed Otoacoustic Emissions and Immittance testing
- Provided pediatric hearing aid services, which included: Fitting, orientation, follow-up, and repairs
- Performed hearing aid outcome measures, which included: Real Ear Measurements, Real Ear Coupler Difference, and Speech Mapping
- Conducted Electroacoustic Analysis testing on hearing aids to provide reliable data for quality control purposes

Ear, Nose, and Throat, Asthma and Allergy Specialty Group (ENTAA Care) – Annapolis & Columbia, MD
Graduate Student Intern *August 2017 - Present*

- Provided comprehensive audiological diagnostic evaluations
- Provided comprehensive vestibular diagnostic testing, which included: Videonystagmography (VNG) and Calorics Testing
- Performed auditory processing disorder screenings
- Performed Auditory Brainstem Response and Electrocochleography testing
- Provided hearing aid services, which included: Evaluations, fitting, orientation, follow-up, modifications and repairs

Johns Hopkins Bayview Medical Center – Baltimore, MD
Graduate Student Intern *May 2017- August 2017*

- Provided comprehensive audiological diagnostic evaluations for adults and special populations
- Provided comprehensive pediatric audiological diagnostic evaluations, which included: Conventional audiometry, visual reinforcement, and conditioned play audiometry
- Performed Otoacoustic Emissions and Immittance testing
- Provided hearing aid services, which included: Evaluations, fitting, orientation, follow-up, modifications and repairs

ENTAA Care – Odenton, MD
Graduate Student Intern *January 2017 - May 2017*

- Provided comprehensive audiological diagnostic evaluations for adults and special populations
- Provided comprehensive pediatric audiological evaluations, which included: Conventional audiometry, visual reinforcement, and conditioned play audiometry
- Provided hearing aid services, which included: Evaluations, fitting, orientation, follow-up, and repairs

Towson University Hearing & Balance Center (TU-HBC) – Towson, MD
Graduate Student Intern *September 2015 - December 2016*

- Provided comprehensive audiological evaluations for adults and special populations
- Performed Otoacoustic Emissions and Immittance testing
- Provided hearing aid and assistive listening device services, which included: Evaluations, fittings, modifications and repairs

- Routinely performed hearing aid outcome measures, which included: Real Ear Measurements (REM), Speech Mapping, and COSI inventory
- Conducted Electroacoustic Analysis (EAA) testing on hearing aids to provide reliable data for quality control purposes
- Provided hearing conservation evaluations and counseled employees on the importance and proper use of hearing protection devices
- Provided comprehensive vestibular diagnostic testing, which included: Rotary Chair Testing, VNG, and Caloric Testing

Research

Doctoral Thesis *In process*

- Titled: How accurate dereverberation should be? Effects of underestimating and overestimating binaural room impulse response
- Advisor: Dr. Nirmal Srinivasan, Ph.D., Towson University

Volunteer Experience

Maryland Special Olympics - Towson, MD
Healthy Hearing Volunteer *June 2016; June 2017*

- Screened hearing of athletes
- Counseled patients and families on hearing loss and provided appropriate follow-up recommendations

Leadership Skills

Student Academy of Audiology, Towson University Chapter
President *May 2016-2017*

- Created and disseminated an agenda for each meeting held
- Maintained regular contact with National Student Academy of Audiology (SAA) and collaborated with SAA to expand networking opportunities
- Arranged and managed group fundraising events
- Prepared and participated in philanthropy events with professors, alumni, and doctoral students

Work Experience

Kaplan Hearing Center - Columbia, MD
Administrative Assistant *June 2013 - August 2013*

- Assisted audiologists in patient care and management, performed general office duties, managed patient database, and scheduled appointments
- Worked with audiologists to communicate and assist in treatment and care of patients
- Participated in outreach programs for Senior Community Centers and Assisted Living Communities

Professional Organizations

Student Memberships

- Acoustical Society of America (ASA)
- Student Academy of Audiology (SAA), local chapter and national chapter
- Academy of Doctors of Audiology (ADA)

Skills

- Proficient in the use of multiple Microsoft Office programs, which include: Word, Excel, PowerPoint, and OneNote
- Proficient in the use of multiple hearing aid manufacturers' software, which include: Phonak, Unitron, Widex, Oticon, Starkey, GN ReSound, and Rexton