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Speech Understanding and Listening Effort in Noise with a New Speech Processing Algorithm

Abigail Elizabeth Compton

A dissertation submitted to the Graduate Faculty of

JAMES MADISON UNIVERSITY

In

Partial Fulfillment of the Requirements

for the degree of

Doctor of Audiology

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ABSTRACT

This study examined the effect of a new speech processing strategy (SpeechZone2) in a commercially available hearing aid on speech understanding in noise and self-reported listening effort. Seven adult, experienced hearing aid users (2 males, 5 females; mean age = 64.6 years) with mild to severe, sloping sensorineural hearing loss participated in this study. Binaural Unitron Flex receiver in the ear style hearing aids with closed domes were used to provide the manufacturer prescribed amplification for each participant. The hearing aids were programmed with two separate memories: 1) omnidirectional microphone without SpeechZone2 processing, and 2) adaptive directionality with SpeechZone2 processing. The participants were seated in the center of a five loudspeaker fixed array. HINT scores (dB SNR required for 50% speech understanding) with the speech source at 0° , 90° , 180° , and 270° azimuths were measured for each program while uncorrelated speech babble noise was presented simultaneously from four speakers. The participants were also asked to fill out a short questionnaire on listening effort after each condition. Results showed that the new speech processing algorithm (adaptive directionality with SpeechZone2) did not improve speech understanding in noise compared to the omnidirectional microphone condition ($F_{(1,6)} = 1.723$; $p = 0.237$). Pairwise comparison with Bonferroni corrections ($\alpha = 0.0125$) indicated that there was a significant improvement only when speech was presented from 270° degree azimuth ($p = 0.002$). The ANOVA also revealed a significant effect of the speech source location ($F_{(3,18)} = 5.62$; $p = 0.02$). Regardless of the directionality and speech processing, the participants performed better when speech was presented from the sides (90° and 270°). A Wilcoxon signed-rank nonparametric test showed that there was no significant difference in the self-reported scores in the listening effort questionnaire ($Z = 0.637$, $p = 0.39$). There was a large intersubject variability noticed in this small sample size.

CHAPTER I

INTRODUCTION AND LITERATURE REVIEW

In environments with poor signal-to-noise ratios (SNRs), understanding speech in the presence of background noise can be difficult even for people with normal hearing. For those with hearing loss, it becomes an even greater issue. Moore (1996) provides a review of the effects of cochlear hearing loss on aspects of speech understanding such as audibility, intelligibility by examining the relationship of cochlear hearing loss with loudness perception, frequency selectivity, and temporal resolution.

Hearing Loss and Audibility

“Normal hearing” thresholds typically range from -10 to 20 dB HL across a range of frequencies tested. Hearing loss is noted when thresholds fall outside of that 20 dB HL range. Hearing loss may occur in only one part of the frequency range, but speech understanding may still be impacted, as speech sounds fall at various intensity levels across the frequency range. For normal hearing listeners, all speech sounds are suprathreshold and audible. However, with hearing-impaired listeners, since their thresholds are poorer than normal, the amount by which speech is suprathreshold, as well as the proportion of the speech spectrum that is suprathreshold, is less than for normal listeners (Moore, 1996). This creates an imbalanced perception of the speech spectrum and therefore affects speech intelligibility. Additionally, if any part of the speech spectrum is masked by noise, this will create a loss of information, also affecting intelligibility (Moore, 1996).

Several studies have examined the effects of hearing loss in speech intelligibility. Turner et al. (1992) examined both speech detection threshold (SDT) and speech reception threshold (SRT) in the presence of background noise for both normal hearing and flat moderate cochlear loss. They reported that while the SDT occurred at roughly the same signal to noise ratio (SNR) between the groups, the

subjects with hearing loss required a higher SNR in order to achieve the given speech reception threshold (SRT). This indicated that the issue with understanding speech in noise may not be due to a deficit in *detecting* the signal of interest, but rather a deficit in the ability to utilize audible speech cues presented over a broader frequency range (Moore, 1996). These findings support the idea that audibility is a primary factor in speech intelligibility: speech sounds may be detected even with access to a limited portion of the speech spectrum; however, speech recognition and understanding requires the information to be audible over a wider frequency range, making it difficult to hearing-impaired listeners to understand speech in noise on the same level as normal hearing listeners.

In addition to flat cochlear losses, cochlear losses above 3kHz only have been examined for their influences on speech intelligibility, often using simulations of hearing loss and speech processed through various high- and low-pass filters. Aniansson (1974) demonstrated that when speech sounds above 3 kHz were filtered out of a signal, speech intelligibility significantly decreased, particularly in the presence of background noise (Moore, 2016).

On the reverse side of audibility affecting speech intelligibility, when amplification was provided above 3 kHz for subjects with mild to moderate high-frequency hearing loss, speech intelligibility improved (Baer et al., 2002, Hornsby & Ricketts, 2006; Skinner et al., 1982; Skinner & Miller, 1983). Of note, in these studies, speech and noise were presented from the same degree of azimuth. If the speech and noise differ in their spatial origins, then the frequencies above 3 kHz that are carried in the signal become more important. Particularly if subjects are fit with binaural amplification, this extends the audible frequency range, which has been shown to improve speech understanding.

The rationale for higher-frequency information becoming more important when speech and noise are spatially separated involves the head shadow effect. This is a proven phenomenon, more effective for mid- to high-frequency signals (Shaw, 1974; Bronkhorst & Plomp, 1989) in which a noticeable intensity level difference is present between ears. Listeners can use this to their advantage if

the target signal, most likely speech, is on the opposite side of the head from the noise. As previously stated, this effect is more advantageous at higher frequencies, which supports the idea that an extended audible frequency bandwidth could improve speech intelligibility.

In the event of overlapping speech signals such as in the case of multitalker babble noise, it can be especially difficult to discriminate the target signal, particularly for listeners with cochlear hearing loss. This difficulty is known as informational masking (Brungart et al, 2001). Freyman et al. (1999, 2001) demonstrated that when the simultaneous speech signals are perceived to be spatially separated, this can help with focusing on the target signal, therefore reducing the effects of informational masking and leading to improved speech intelligibility (*cf* Moore, 2016).

These studies support the indication that a separation between the noise source and the signal source will improve speech intelligibility for those with mild to moderate high-frequency cochlear hearing loss, especially with the addition of amplification for frequencies above 3 kHz (Moore, 2016). This is due in part to the audibility principle: frequencies well above 3 kHz have been shown to contribute to speech intelligibility when the target signal is spatially separate from the source of the interfering noise. Audiometric thresholds from 6-10 kHz have the most influence on a listener's ability to exploit this spatial separation for better understanding of speech in noise; if these thresholds are elevated, the listener may not be able to take advantage of this phenomenon (Besser et al, 2015).

Hearing Loss and Loudness Recruitment

Loudness recruitment is an issue that emerges in cases of hearing loss, in which the range of intensity levels over which sounds are both audible and comfortably heard is reduced or narrowed. This range of levels is known as the dynamic range. In a normal hearing listener, there is a large difference between the level of the hearing threshold, at which the sound is just audible, and the level at which the sound is uncomfortable (UCL), resulting in a wide dynamic range.

As discussed above, hearing-impaired listeners have thresholds above the normal range. However, their UCL is likely to stay the same as a normal hearing listener, leading to a reduced dynamic range, otherwise known as recruitment. Several studies attempted to simulate the effects of loudness recruitment on the auditory system and on speech intelligibility. Villchur (1977) simulated the effects of recruitment on speech intelligibility for subjects with severe hearing loss by filtering a signal through 16 bands and expanding the range of levels in each band before recombining them. This was done to essentially reverse the fast-acting compression that occurs in a normal cochlea at each characteristic frequency of a stimulus. According to Moore's review of this study (1996), increased difficulty was found for intelligibility of words and sentences in the presence of quiet, white noise, and speech-shaped noise for simulated hearing losses. This indicates a difference in processing between normal hearing listeners and those with hearing loss. Normal hearing listeners are able to take advantage of the spectral and temporal dips of a signal, even in the presence of a single-talker masker. However, in order for "dip listening" (Moore, 1996) to be successful, the listener must have a wide dynamic range. As discussed above, hearing loss will narrow the dynamic range which, while also leading to recruitment, can indicate more difficulty hearing in noise. It is important to note that the type of background noise may influence speech intelligibility regardless of dynamic range: as Moore (1996) notes, "When a background of speech-shaped noise is used, dip listening is of much less importance, since the noise does not contain dips of sufficient magnitude or duration. Hence, speech intelligibility depends more on the higher-level portions of the speech, and these are less affected by dynamic range". This difference in speech intelligibility between various types of background noise was later explored by Moore et al. (1997) using simulated hearing losses in normal hearing subjects, as well as subjects with cochlear hearing loss.. It was noted that for the simulated hearing losses, speech intelligibility improved significantly for single-talker noise compared to speech-shaped noise. This was in contrast to the subjects with cochlear loss, in which there was very little difference in their discrimination abilities between the two types of

background noise. The findings of this study pointed to the possibility that other factors outside of audibility and recruitment, may be affecting speech intelligibility in those with cochlear hearing loss.

Effects of Hearing Loss on Frequency Selectivity

Many studies examining effects of hearing loss on speech intelligibility use the speech reception threshold (SRT) as the dependent variable. It has been noted that even among subjects with similar degrees of hearing loss, the SRT for each subject varies. This indicates that processes occurring at levels above absolute threshold, such as frequency selectivity and temporal resolution, may be responsible for a significant proportion of the variance in SRTs that occur among subjects with similar degrees of hearing loss (Moore, 1996). In other words, for subjects with moderate or greater cochlear losses, the variability in SRT cannot be explained solely by the subjects' absolute thresholds. Several correlational studies, including Glasberg and Moore (1989) examined the effects of bilateral cochlear hearing loss on SRT compared to normal hearing, both in quiet and in speech-shaped noise. As expected, the SRTs for the hearing-impaired group were higher in both conditions, but it was noted that in the same 75 dB level of speech-shaped noise, the SRT was raised significantly more for the hearing impaired subjects than for the normal hearing subjects. However, the SRT in noise had a lower correlation to pure tone average (PTA) for the hearing impaired group compared to the SRT in quiet, indicating that the variance in SRT was not entirely due to variations in absolute threshold (Moore, 1996). In fact, it was found that the SRT in the 75 dB noise condition correlated more closely not with the PTA, but with the three measures of suprathreshold discrimination used in the psychoacoustic tests given to both groups of subjects, assessing frequency discrimination of both pure and complex tones, as well as gap detection in bands of noise. This indicates that in a high level of background noise, the ability to understand speech is determined more by these suprathreshold abilities than by absolute threshold sensitivity.

Frequency selectivity as a factor for understanding speech makes sense when the speech spectrum is considered. Speech sounds fall along a wide range of frequencies. If the natural auditory filters of the basilar membrane are working correctly, these sounds are easier to discriminate. However, with hearing loss comes a broadening of the auditory filters, creating less specific frequency selectivity. This was demonstrated in a study by Thibodeau and van Tasell (1987) in which subjects with both normal hearing and moderate flat cochlear hearing loss were asked to discriminate the syllables /di/ and /gi/ both in broadband noise, and in noise with a 2000Hz spectral notch. Subjects' frequency selectivity was estimated by measuring the subjects' ability to detect a 2000Hz sinusoid as a function of the width of a 2000Hz spectral notch in noise. It was noted that subjects with poorer frequency selectivity (a lower percent-correct for detecting the 2000Hz sinusoid) also showed worse discrimination between the two syllables. However, Moore (1996) notes that this study used stimuli that were restricted to a certain frequency range; natural speech contains information over a much broader frequency range, making it difficult to demonstrate the effects of reduced frequency selectivity. This is true with vowels as well as consonants, though to a lesser extent, as demonstrated by Turner and Henn (1989), whose results indicated that for subjects with moderate cochlear hearing loss, vowel identification is often not affected. This is due to several factors, the most important of which is that the spectral differences between vowels are much larger than between consonants, making it easier to identify different vowels even if frequency selectivity is impaired. Additionally, when vowels are naturally produced rather than synthetically, they contain temporal cues which may help to identify them even in cases of impaired frequency selectivity (Moore, 1996).

Reduced frequency selectivity can also have deleterious effects on localization skills. The ability to localize a sound depends on the information obtained when a signal reflects off the listener's outer ear. This change in the signal will alter the sound spectrum that actually reaches the eardrum and is processed through the rest of the auditory system (Moore, 2016). Gardner & Gardner (1973) found that

this change in spectrum is more dramatic for signals higher than 3 kHz. Therefore, for high-frequency losses, e.g., elevated thresholds above 3 kHz, localization will likely be adversely affected, as the ability to use pinna cues will have decreased in combination with the reduced frequency selectivity that comes as a consequence of hearing loss. This decrement in localization ability will make it more difficult to detect where sounds are coming from; both the target signal and the interfering noise. The impaired ability to localize and discriminate sounds can lead to further difficulties in speech intelligibility due to a lack of spatial-separation advantage (Moore, 2016).

Effects of Hearing Loss on Temporal Resolution

Temporal resolution of a signal depends on two elements: the envelope of a signal, which contains relatively slow fluctuations in signal amplitude over time; and the temporal fine structure, or the rapid oscillations within a signal, a rate closer to the center frequency of the signal (Moore, 1996). Speech, as a broadband signal, is naturally band-pass filtered by the cochlea along the basilar membrane according to place coding. Therefore, we can examine the role of both envelope cues and fine-structure cues in speech by simulating the band-pass filtering performed by the basilar membrane. The speech signal was filtered into multiple adjoining frequency bands. Each band was processed in a way that preserved only either the envelope or the temporal fine structure cues (Moore, 1996). It was found that when speech was separated into 4-16 bands, envelope cues on their own provided high intelligibility of speech in quiet for both normal hearing subjects and hearing-impaired subjects. However, even for normal hearing listeners, speech intelligibility suffers when an envelope-filtered signal is presented in a fluctuating background sound, such as a single-talker masker, or amplitude-modulated noise. Moore (1996) suggests that this is due to the fact that envelope cues alone do not allow for effective listening in the “dips” of a fluctuating background sound. Furthermore, in order to effectively use the “dips” in a fluctuating background noise, the listener’s auditory system must be able to determine whether the

signal that persists during the dips is dominated by the target speech or the background sound. In a normal auditory system, this is done by using the information derived from neural phase locking to the temporal fine structure of a signal. Changes in the rate of phase locking when a dip occurs indicate that the target speech is present in the dip.

Therefore, it may be that hearing impaired listeners are unable to take advantage of the information from the temporal fine structure of a signal, which can add to their difficulty understanding speech in noise: they are unable to “listen in the dips” (Moore, 1996). This was shown in a study by Lorenzi et al. (2006) in which speech was processed to remove envelope cues while preserving the fine structure cues. Even in quiet, hearing-impaired subjects performed more poorly compared to normal hearing subjects when asked to identify nonsense syllables in quiet. These same hearing impaired subjects were later tested with steady-state and modulated background noise as well. The data support the idea that listening in the dips depends on the use of temporal fine structure information, and those with cochlear hearing loss are not as able to take advantage of this.

The loss of ability to take advantage of temporal fine structure information and “listen in the dips” may be due either to reduced phase locking precision or else to a reduced ability to extract information from the phase locking patterns (Moore, 1996). It was originally thought that a cochlear hearing loss would only affect the filtering and the nonlinearity of the basilar membrane, not the temporal integrator itself. This was supported by Drullman, Festen, & Plomp (1994) who examined the effects of “temporal smearing” on speech intelligibility. The incoming signal was split into frequency bands, with the envelope of each band being low-pass filtered with a different cutoff frequency. As the cutoff frequency reduced, the envelope became more “smeared” in the time domain, to simulate the loss of temporal resolution of the signal. It was found that envelopes with higher cutoff frequencies did not affect the SRT in the presence of speech-shaped noise. However, cutoff frequencies of 4 or 8 Hz significantly elevated the SRT. In other words, when very rapid fluctuations of 16 Hz or greater were

filtered out, there was very little effect on intelligibility, even in noise, compared to when slow fluctuations were filtered out. This suggests that reduced temporal resolution does not have a significant adverse effect on speech intelligibility for most people with cochlear hearing loss, as these subjects could easily detect modulation rates up to 16 Hz (Moore, 1996).

In summary, it should be considered that reduced frequency selectivity still has an effect on speech intelligibility even when stimuli are presented at suprathreshold levels. Linear amplification will be unable to improve upon this. Additionally, those with severe to profound cochlear losses will experience difficulty listening to speech in noise for reasons other than pure loss of audibility. Due to their reduced dynamic range, they will experience loudness recruitment and they will be unable to use temporal and spectral cues to “listen in the dips” of background noise. However, with the advent of digital signal processing hearing aids, as well as directional microphones, it has become easier to address some of these concerns.

Digital Signal Processing and Noise Reduction in Hearing Aids

As opposed to analog hearing aids, which work by amplifying a continuous incoming acoustic waveform, digital hearing aids work by performing an analog to digital conversion, implementing algorithms in the processing core of the hearing aid to clean up the signal, and then converts from digital back to analog, in order to deliver the processed acoustic signal to the patient’s ear. As a result of this change in how the signal is analyzed, digital processing is able to handle a much wider range of frequencies, and as the signal is broken down into a binary signal stream made of discrete bits, digital processing makes it easier to individually adjust specific bands or channels of the signal.

These advancements in processing technology have made noise reduction in hearing aids much more sophisticated over the years. It is important to note that there is no program that can completely

remove background noise from a speech signal; all that the processing can do is reduce the noise relative to the signal of interest.

There are three main strategies when it comes to digital hearing aid processing and noise reduction: they work in the spatial domain, the spectral domain, and the temporal domain. The *spatial domain* works to figure out which direction the sounds of interest are coming from. Typically, hearing aids will designate the signal coming from the front (0 degrees azimuth) as the desired signal. This is also known as adaptive directionality, which has been one of the signature improvements of hearing aid processing. Adaptive directionality uses a system of twin omnidirectional microphones to determine the direction of the desired signal. As omnidirectional microphones are sensitive to sound originating from all directions relative to the person wearing the hearing aids, their sensitivity can be adjusted based on the direction of the signal of interest. With a system of two omnidirectional microphones, there is a built-in electrical delay that can be adjusted through computer programming. The length of the delay between the microphones will affect the level of directionality (focus) of the microphones. This is called “beamforming” technique. A longer delay (a larger “distance” between the microphones) will result in a greater focus toward the front of the speaker.

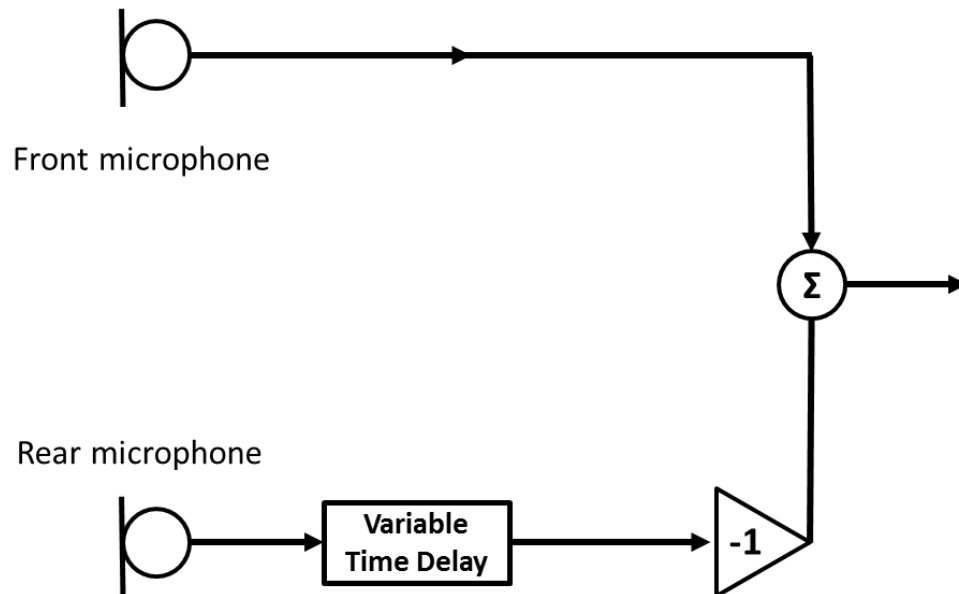


Figure 1. Functional block diagram of an adaptive directional microphone system. The variable electronic time delay is added to the signal from the rear microphone. The adaptive microphone system can change the directional sensitivity by changing the time delay.

The next strategy used in digital noise reduction involves the *spectral domain*, in which the processor determines which frequencies are desired. For instance, the speech intelligibility zone involves primarily the mid-frequencies, while steady-state background noise is typically more low-frequency. Filtering the signal correctly and applying frequency-specific gain reduction can attenuate low-frequency noise without affecting the speech intelligibility zone in the mid-frequencies.

Finally, the *temporal domain* of digital noise reduction determines the differences in signal envelopes for the competing signals (desired input versus noise). The envelope of a signal is the boundary by which the signal is contained, following its fluctuations in frequency and amplitude. These variations in frequency or amplitude differ depending on the type of signal. For instance, while mechanical or steady-state noise has a low modulation depth and high modulation frequency, speech

signals typically have a high modulation depth and low modulation frequency. When the signal is getting processed, the shape of the signal envelope can be key in whether the signal gets attenuated or amplified. This is the most popular technique in digital noise reduction strategies.

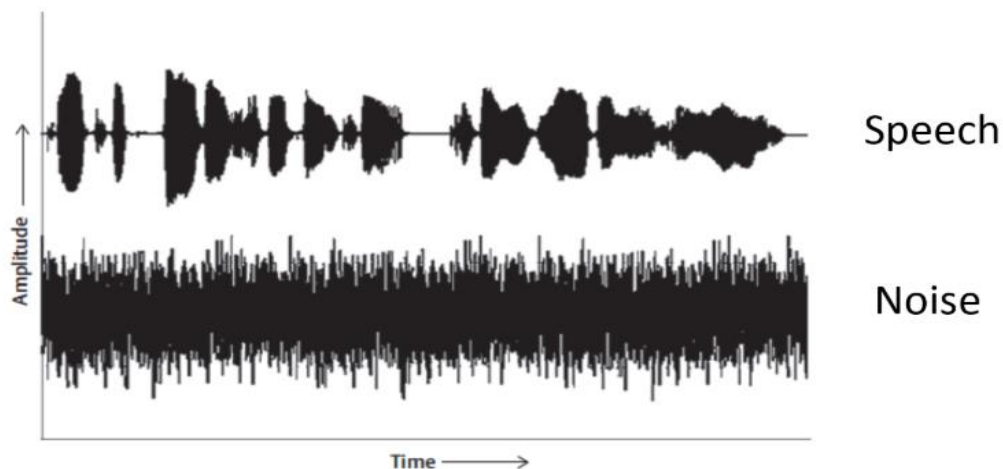


Figure 2. Waveforms of speech (top) and computer fan noise (bottom) are shown. The x-axis represents time and y-axis shows amplitude in arbitrary units. Speech has a slow modulation rate compared to that of noise, which fluctuates rapidly. The depth of modulation for speech is deep while the modulation depth for noise is shallow.

Directionality

As explained above, hearing aids can either use omnidirectional or directional microphones. Omnidirectional microphones are sensitive to sound originating from all directions (degrees of azimuth) relative to the person wearing the hearing aid. In a system of two omnidirectional microphones, an artificial delay between ports can be created.

On the other hand, directional microphones work by improving the SNR based on the spatial location of sounds of interest, relative to the location of the unwanted sounds. Directional microphones use two inlet ports, spaced a certain distance apart. Depending on the location of a sound, the wave will

arrive at the two ports at two different moments in time. Sounds arriving from the side should arrive at the two ports at the same time, canceling out most of that energy and giving more amplification to sounds arriving from the front (assuming that is the direction of interest) because sounds arriving from the front or back will reach the two inlet ports at slightly different times. This principle works for hearing aid users because “noise” (any undesirable signal that interferes with the message) will likely be coming from behind them rather than only from the sides - think of a busy restaurant, with noise going on all around the hearing aid user. With a single mic that has two ports, the filter in the middle will delay the arrival of sound to the diaphragm. Depending on the length of the delay, the directional pattern and direction of focus on the incoming signal will be different.

Listening Effort in Noise

The ability to process and understand speech requires attention as well as substantial cognitive resources. The cognitive load created by processing speech in a variety of environments is called listening effort. This load, and thereby the required listening effort, may be increased with a degraded signal, interference such as background noise, or in limitations of the listener, such as hearing loss (McGarrigle et al., 2014). Note that listening effort may still be required for a normal hearing listener; however, as discussed above, normal hearing listeners have an improved ability to carry out “backstage operations” - that is, selective processing of a particular sound while simultaneously filtering out irrelevant information. This is similar to the phenomenon of “listening in the dips” described by Moore (1996). This ability is also called a selective gain mechanism. This mechanism is degraded somewhat in hearing-impaired listeners. The process of maintaining ongoing “backstage operations” is much more taxing, particularly in challenging environments. Speech perception is not the only type of auditory processing that requires effort if hearing loss is present; as discussed above, localization ability is often degraded with high-frequency hearing loss (Gardner & Gardner, 1973). Based on this, it can be assumed

that additional mental effort would be required for speech understanding in the presence of background noise. Ohlenforst et al. (2017) suggest that the degree of processing required to keep up with simultaneous ongoing auditory streams (for instance, understanding speech in background noise) increases the cognitive load imposed by the listening task.

When a listener has hearing loss, this cognitive load is even larger. Indeed, Larsby et al. (2005) found that hearing-impaired listeners struggle more with speech-recognition tasks compared to normal-hearing listeners. Additionally, it was found that hearing-impaired listeners require a higher listening effort in order to achieve speech-recognition results comparable to those of normal-hearing listeners.

The effects of increased listening effort in subjects with hearing loss have been well documented. Fatigue, stress, and tension were all found to be consequences of increased listening effort (Hornsby, 2013; Stephens & Hetu, 1991; Kramer et al., 1997, 2006). These effects can be attributed to increased strain on an auditory system already experiencing deficits due to hearing loss. These consequences can lead to increased social isolation, due to the amount of energy and effort required to follow speech, particularly in noisy environments (Weinstein & Ventry, 1982; Demorst & Erdman, 1986; Strawbridge et al., 2000).

It has been well documented that modern hearing aid technology can assist with improving speech understanding in noise due to advancements made in compression, directional microphones, and noise reduction algorithms (Ohlenforst et al., 2017; Dillon, 2001). Zekveld et al. (2011) found that subjective reports of listening effort decreased slightly when speech-in-noise intelligibility was increased for young and middle-aged listeners with hearing loss, in contrast to age-matched normal-hearing listeners. Klink et al. (2012) further demonstrated that this phenomenon should be attributed to hearing loss, rather than age, as a factor of intelligibility. Due to the compensation for the hearing loss provided at least partially by amplification, hearing aids were found to result in a reduction in listening effort (Hornsby, 2013; Picou, Ricketts, & Hornsby, 2013).

Several studies have investigated objective and subjective measures of listening effort in both quiet and noisy environments, for both normal hearing and hearing-impaired listeners. Gatehouse & Noble (2004) created the Speech, Spatial & Qualities of Hearing Scale (SSQ) in an effort to quantify listening effort in various real-world situations, with participants giving answers on a scale of 0-10. The Amsterdam Checklist for Hearing & Work (Kramer et al., 2006) was another attempt to quantify listening effort and fatigue in noisy situations. These subjective methods, along with other questionnaires or self-rating scales, provided either immediate or retrospective data regarding the amount of effort perceived by the subject during a task (Ohlenforst et al., 2017).

Objective measures have also been used to quantify listening effort. In the case of Larsby et al. (2005), listening effort was measured using reaction times on speech-recognition tasks, as well as subjective ratings. Zekveld and Kramer (2014) investigated objective measures of listening effort by observing subjects' pupil dilations during a word-recognition task, using subjective ratings of listening effort as a cross-check method. Larger pupil dilations were found to be associated with a larger cognitive load, while subjects were asked to rate themselves 0-10 on questions such as "How much effort did you put into this task?" "How many sentences do you think you perceived correctly?" and "How often did you give up trying to understand the sentences?"

Listening effort is an emerging area of research as hearing aid technology continues to improve. Both objective and subjective measures of listening effort should continue to be investigated as a potential means of exploring cognitive load and hearing aid technology's effects on speech perception in difficult listening environments.

Rationale for the Current Study

Based on the existing understanding of directional microphones and their role in digital speech processing and noise reduction, the current study seeks to examine the performance of one particular

processing algorithm created by the manufacturer Unitron. This processing algorithm, called SpeechZone2, combines three approaches for detecting speech in noise (Rule et al., 2017).

Speech Locator, its spatial component, uses multiple microphones and beamforming techniques to locate different sound sources and uses binaural technology to determine the location of speech.

Speech Focus, its directional component, applies adaptive directionality principles to focus on the determined direction of speech. For instance, if speech is determined to be coming from in front of the listener, the algorithm uses the maximum setting of its adaptive directionality system to focus in on the speech and works to reduce noise sources at the sides or behind the listener. On the other hand, for speech sources determined to be coming from the side, the algorithm will use a “binaurally coordinated asymmetric response” (Rule et al., 2017) and will focus on that side with more intensity than it will for the other side, front, or back. For speech sources determined to be coming from the back, beamforming techniques will still be used, but the beam will focus on the back while also maintaining some front audibility, although it reduces that by approximately 10 dB relative to the target speech at the back.

Dynamic Spatial Awareness, its third component, restores the natural localization cues that suffer when the hearing aid uses the aggressive directional responses of Speech Locator and Speech Focus. When speech is located to be either in the front or back of the listener, both ears will receive the same frequency response modification, making it sound “normal.” However, when the signal is found to be coming from the side, DSA will attenuate the contralateral side, thereby enhancing the head shadow effect and making speech sound more natural (Rule et al., 2017).

SpeechZone 2 also uses binaural spatial processing to determine the exact location of speech, depending on the location of the speech source (front, back, left, or right). Based on the determination of the speech source location, the hearing aid pair automatically selects a symmetric or asymmetric microphone strategy to ensure the best speech understanding in noisy environments. This is all done

automatically when classification within Unitron's SoundNav automatic program is either the "Conversation in a crowd" or "Conversation in noise" listening environments. When speech is determined to be arriving from the front of the hearing aid, the microphone response is a symmetric multi-band adaptive directionality. If the speech is from the side, the microphone response becomes asymmetric, applying omnidirectional with Pinna Effect on the side from the speech side and adaptive directionality on the side with noise. When speech is located from the back of the hearing aid, the microphone response becomes symmetric, with both hearing aids in omnidirectional mode. It is important to note that in order for the SpeechZone2 algorithm to work efficiently, the hearing aid user needs to wear a pair of hearing aids that are programmed to synchronize and wirelessly communicate with each other (Howard, 2014).

The goal of this study is to evaluate the capabilities of the SpeechZone2 processing algorithm, combined with the benefits of adaptive directionality, compared to a traditional omnidirectional microphone setting with no additional processing algorithm. In addition, the current study will evaluate the claim that SpeechZone 2 can reduce listening effort in difficult listening environments.

CHAPTER II

METHODS

Participants

Seven adults (2 male, 5 female) with previously diagnosed, mild to severe sloping symmetrical sensorineural hearing loss served as participants in the present study. Mean hearing thresholds with standard deviations of the participants are shown in Figure 3. Data for the left and right ears at octave frequencies between 250 to 8000 Hz are depicted using “X” and “O” symbols, respectively. Symmetry between the two ears was defined as a difference between hearing thresholds of no more than 20 dB at any octave frequency between 500 Hz to 4000 Hz (Pittman & Stelmachowicz, 2003). Air conduction hearing thresholds were tested as a part of the research protocol, and normal middle ear functioning was assessed by verifying the presence of a type ‘A’ tympanogram. Participants ranged in age from 25 to 81 years (mean age = 64.6 years).

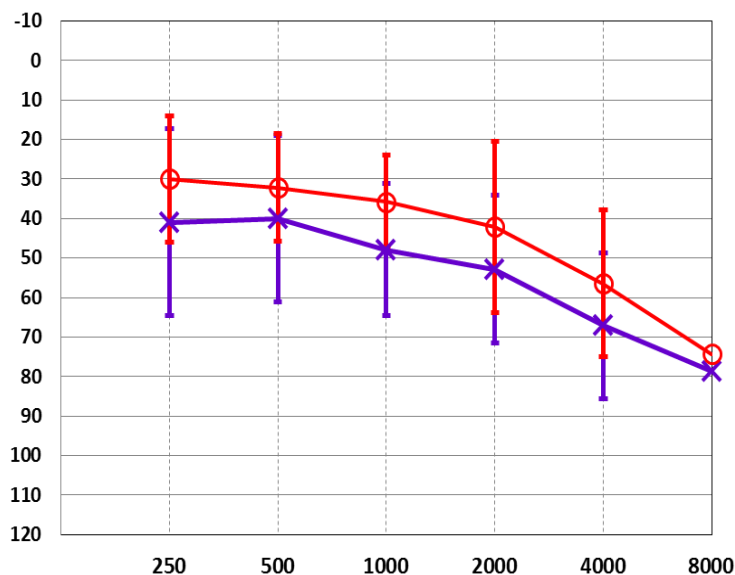


Figure 3. Mean audiometric thresholds at octave frequencies for the right (O) and left ear (X). Error bars represent ± 1 standard deviation from mean.

The participants' experience with hearing aids ranged from 2 years to 45 years. Subjects were recruited via fliers posted around the university and at the university audiology clinic, as well as at local retirement facilities. A mass email was also sent out to JMU faculty with the eligibility information.

Preparation

The test duration was approximately two hours, including short breaks. If necessary, the test was divided into two sessions to accommodate subjects' needs or schedules. Subjects were compensated \$10 per hour of their time in this study.

All subjects underwent air-conduction threshold audiometry to confirm the degree of their hearing loss. Tympanometry was performed on all patients to ensure there was no conductive component to the hearing loss. A brief case history was also taken, in which the subject was asked to detail how long they had had a hearing loss and how long they had been wearing hearing aids.

After the audiogram was completed, a pair of Unitron Flex digital hearing aids with closed domes were programmed to the participant's hearing loss. In the hearing aid software, two separate programs were created. Program 1 was the "omnidirectional" mode, in which SpeechZone 2 was turned off. Program 2 was the "adaptive" mode, in which SpeechZone 2 was turned on. In-situ verification of the hearing aids' output was performed before proceeding to the Hearing-In-Noise Test (HINT). Starting condition (either Program 1 or Program 2) was counterbalanced across participants to minimize any learning effect.

Stimuli

Pre-recorded sentences from Hearing in Noise Test (HINT), an adaptive procedure that tests speech understanding in the presence of background noise, was used to evaluate the effect of SpeechZone 2. This study utilized background babble rather than speech-shaped noise used in the

original HINT. Subjects were required to repeat sentences of varying intensity in the presence of background babble, which was kept at a constant intensity level of 70 dBA.

The first sentence the subject repeated was increased in intensity until the subject was able to repeat the entire sentence correctly in the babble. The next four sentences were described as “practice.” If the subject repeated the sentence correctly, the intensity level of the speech was decreased by 4 dB HL. If the subject repeated the sentence incorrectly, the intensity was increased by 4 dB HL.

After the four “practice” sentences were completed, the subject repeated six more sentences following the same procedure, but with the step size reduced to 2 dB. This created a total of ten sentences repeated by the subject. This comprised one “set” of HINT sentences.

Test Setup

All testing took place in the Hearing Aid Research Laboratory. There were five speakers arranged in a circle, 90 degrees apart from each other. Background babble was presented from four speakers (0, 90, 180, 270 degrees) while the speech was delivered from a separate speaker located at 0 degrees azimuth. The subject began the test seated in a chair that had the ability to swivel 360 degrees. Hearing aids were randomly programmed either for Program 1 or Program 2 for the first condition. The subject began facing forward, so that speech was delivered at 0 degrees azimuth, as seen in Fig. 4 shown below.

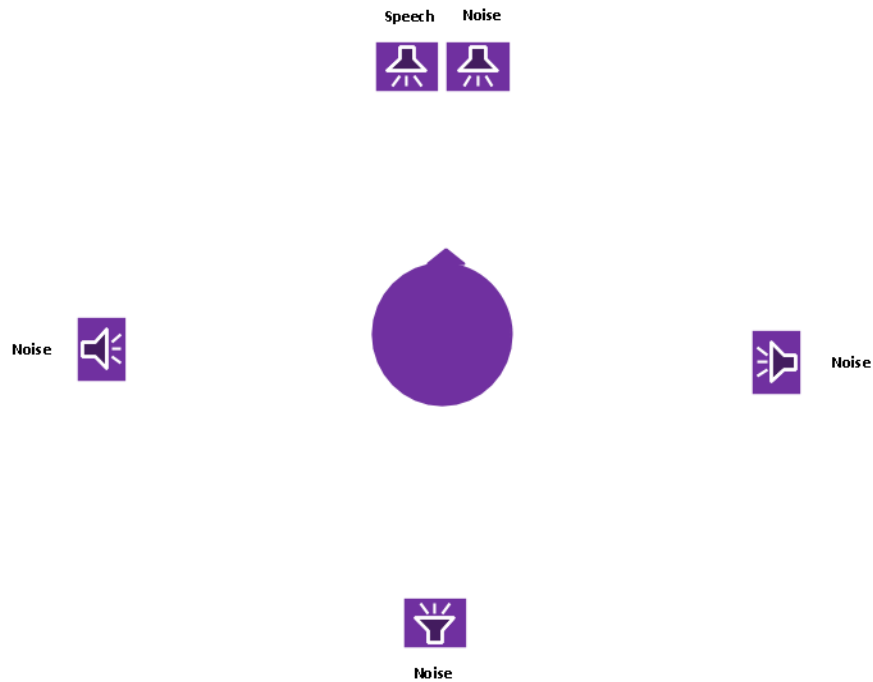


Figure 4. Schematic representation of speech and noise sources in the study. Separate loudspeakers were used for speech and noise at 0 degree azimuth to simulate a real world situation (speech originating from a separate location).

The subject repeated ten sentences with speech at 0 degrees azimuth, to complete one “set”. The subject then rotated to face the speaker at 90 degrees, so that speech was presented from the left side (270 degrees azimuth), and completed another set of ten sentences. Two more sets were completed, with the subject rotating in 90 degree steps. These four sets in total comprised one test block.

After the first test block was complete, the subject was given a short questionnaire with items adapted from several other questionnaires. These included the Speech, Spatial and Qualities of Hearing Scale (Gatehouse & Noble, 2004), the Amsterdam Checklist for Hearing and Work (Kramer et al., 2006), and studies by Hornsby (2013), Koelewijn et al. (2011), and Zekveld & Kramer (2014). The short questionnaire used in the current study included five questions about listening effort, required

concentration, perceived speech understanding, giving up, and a final question about listening fatigue. The subject was asked to answer the questions based on the test block they had just completed.

After the subject had completed all four sets with either Program 1 or Program 2, the subject was turned to face 0 degrees azimuth again. The hearing aids were changed to the other program and the second test block was completed in the same way. Afterward, a second copy of the questionnaire was administered, with the subject asked to base their answers to the questions on the second test block. The participant was blinded to the hearing aid programming.

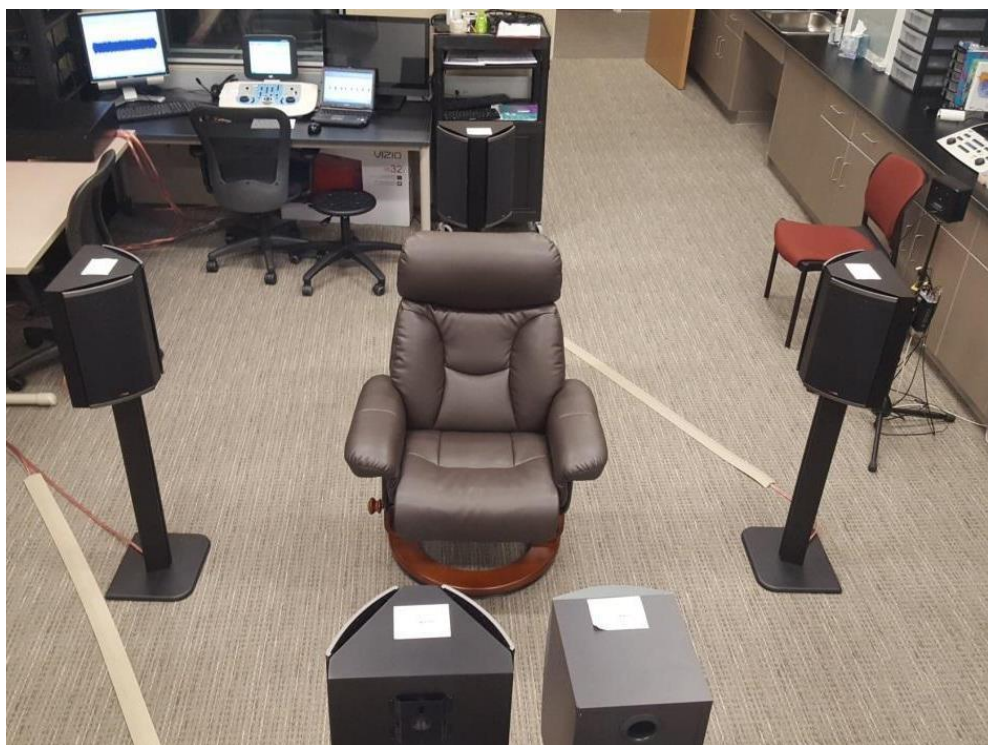


Figure 5. Image of actual test setup.

Data Analysis

A repeated-measures Analysis of Variance (ANOVA) was conducted to examine the differences in HINT scores for subjects ($n=7$) between the conditions of SpeechZone2 OFF and ON. SpeechZone 2 (two levels) and speech source azimuth (four levels) were used as within-subject variables.

Planned pairwise comparisons with Bonferroni correction were performed for degrees of azimuth between conditions (omnidirectional vs. SpeechZone 2 for 0 degrees of azimuth, 90 degrees, etc.).

CHAPTER III

RESULTS

Speech reception thresholds (HINT score) are shown as a function of speech source location for omnidirectional and adaptive directionality + SpeechZone2 in Figure 6. The error bars represent ± 1 standard error of mean. Lower HINT scores indicate better performance. There was large intersubjective variability observed in this small sample size ($n=7$). The effect of adaptive directionality and SpeechZone2 processing was examined by a repeated measure ANOVA ($\alpha=0.05$). Results indicated that there was a trend of improved speech understanding with these features turned on. However, statistical significance was not observed ($F_{(1,6)}=1.723$; $p = 0.237$).

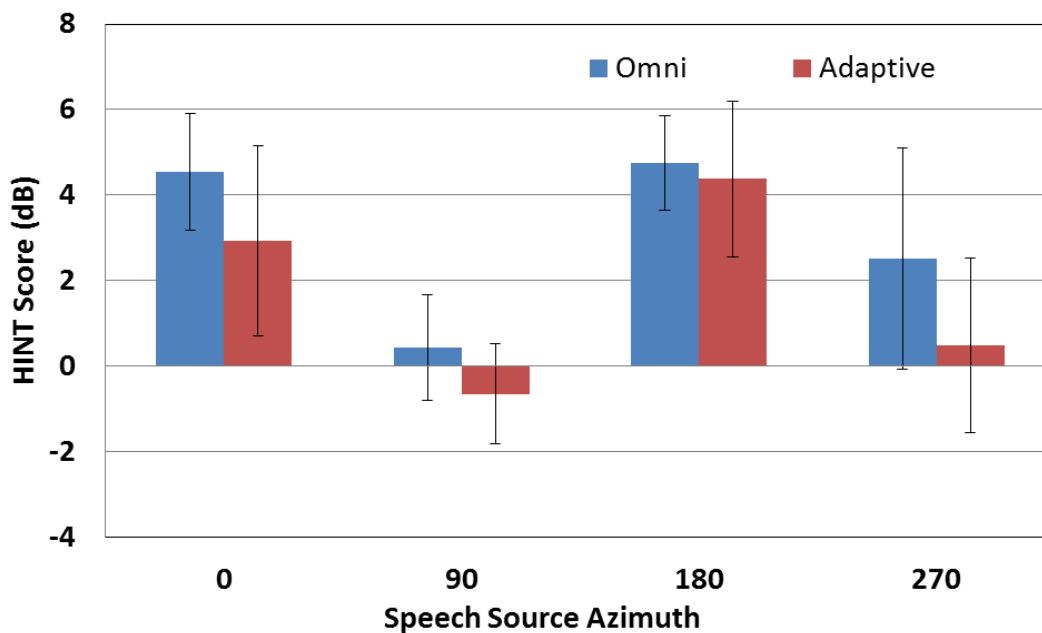


Figure 6. Mean HINT scores for Omnidirectional (blue bars) and Adaptive directional with SpeechZone2 (red bars) for different speech source locations (0, 90, 180, and 270 degree azimuth). Error bars indicate ± 1 SE of mean. A lower HINT score indicates better performance.

Pairwise comparison with Bonferroni corrections ($\alpha=0.0125$) indicated that there was a significant improvement only when speech was presented from 270 degree azimuth ($p=0.002$). The ANOVA also revealed a significant effect of the speech source location ($F_{(3,18)}= 5.62$; $p=0.02$). Regardless of the directionality and speech processing, the participants performed better when speech was presented from the sides (90° and 270°).

Individual Performance

As evidenced in the standard error bars in Figure 6, there were large differences across individual performances. Each participant's speech reception threshold (HINT score) is shown in individual panels in Figure 7 below. A lower HINT score indicates better performance. The performance of participant 2 was different from the rest of the group. First they needed much more favorable signal to noise ratio (SNR) to understand speech, and second, their HINT score increased (worsening performance) with the adaptive directionality+SpeechZone2 condition. Since there was large variation in the HINT scores across the seven participants, the difference in HINT scores between the omnidirectional and the adaptive programs were calculated from individual participants. Figure 8 shows the difference (omnidirectional – adaptive) for each participant. A positive difference in Figure 8 indicates better performance with adaptive directionality + SpeechZone2 program.

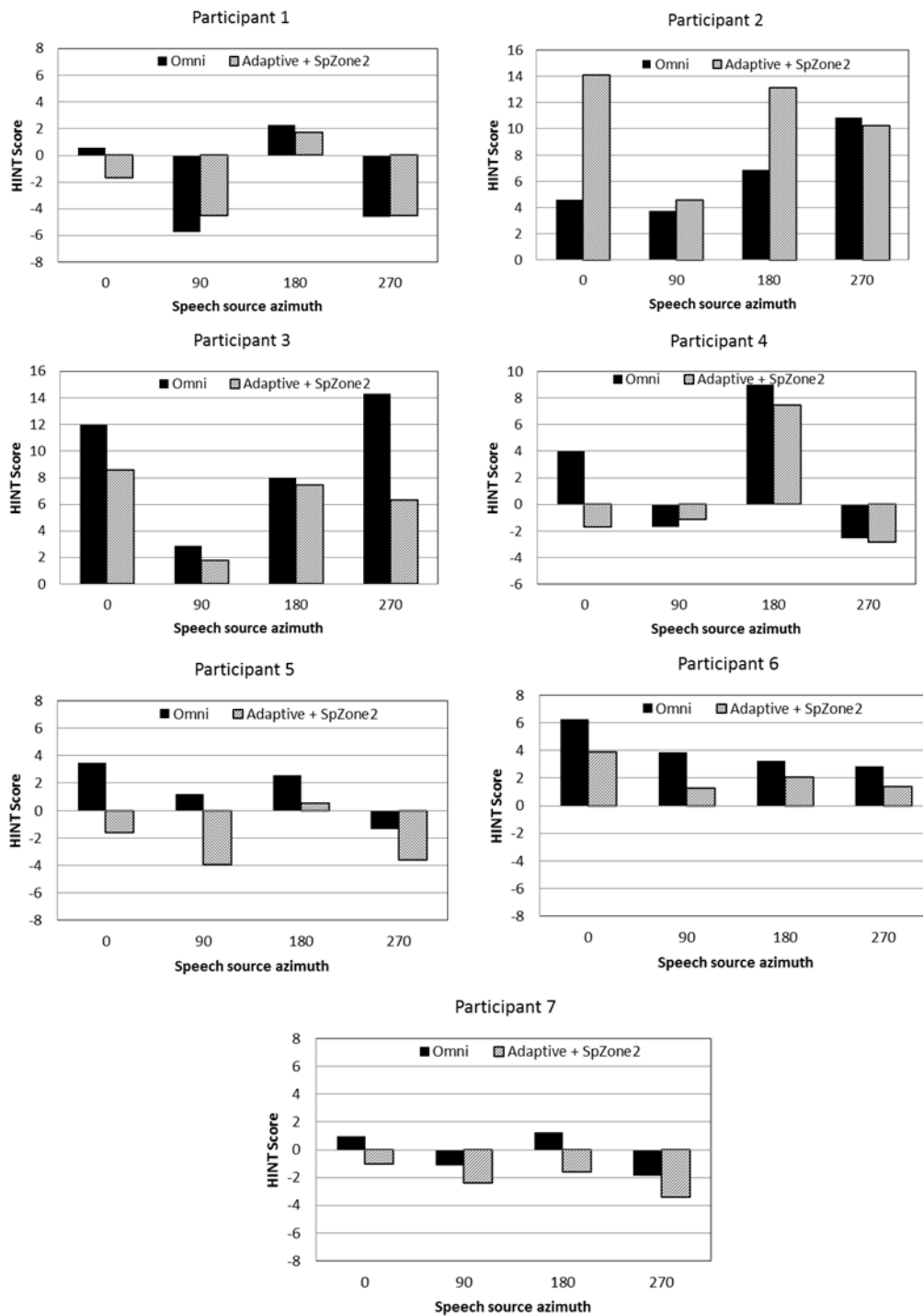


Figure 7. Individual HINT scores for the two test conditions – omnidirectional (black) and adaptive directionality+SpeechZone2 (gray). The range on the y-axis is shifted to fit the wide range of individual scores. A lower HINT score indicates better performance.

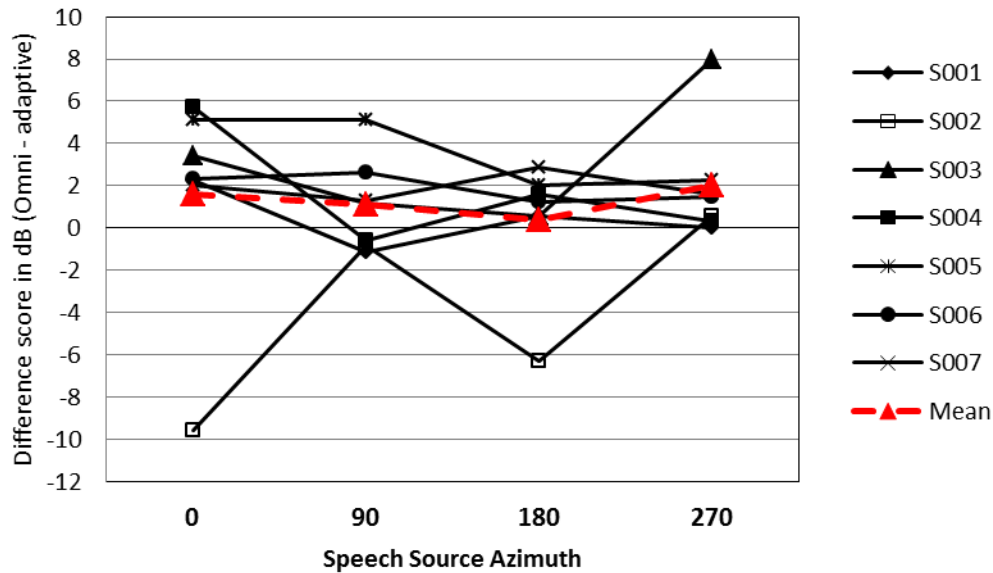


Figure 8. Individual differences in performance between the omnidirectional and the adaptive directional + SpeechZone2 conditions. Positive values indicate a better speech understanding with the adaptive program. Mean scores are shown in the dashed line.

Listening Effort

Mean self-reported scores on a 10-point Likert scale for five different questions are shown in Figure 9. A higher rating indicates greater effort or concentration for the first two sets of bars. The third question explored how much the participants “gave up” during the test. A lower score indicates that the participants reported not giving up while responding to the speech in noise. The fourth question on speech understanding evaluated the overall feeling of how well a participant perceived to have performed on the speech understanding task. A higher score indicates that the participants self-reported that they understood the sentences in the presence of the background noise. The final question was on the overall fatigue a participant experiences at the end of the day. This question was included as a measure of internal consistency for each subject as there should not be any significant

difference in the self-reported overall fatigue at the end of the day irrespective of the hearing aid programming they listened to in the current session. A Wilcoxon signed-rank nonparametric test showed that there was no significant difference in the self-reported scores in the listening effort questionnaire ($Z = 0.637$, $p = 0.39$).

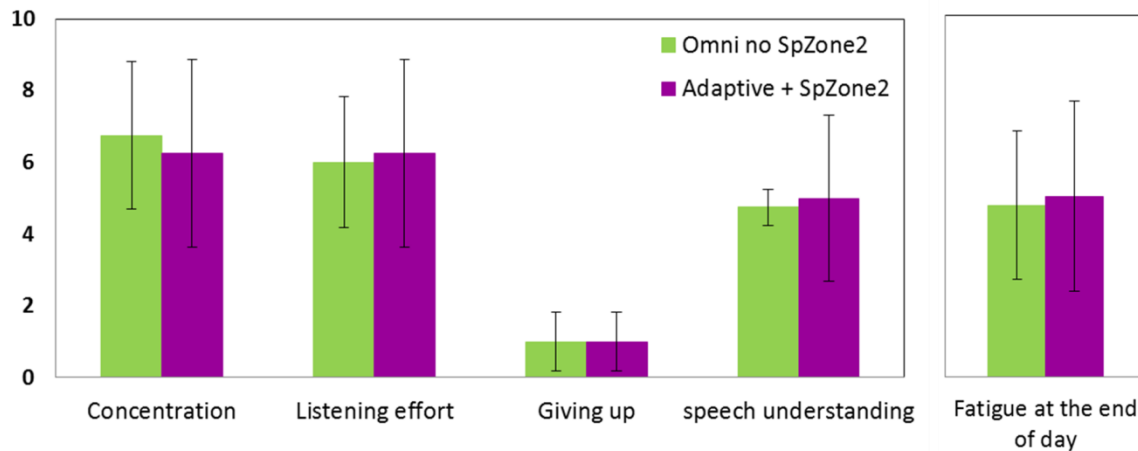


Figure 9. Mean subjective rating of perceived concentration, listening effort, speech understanding, and quitting between the two conditions. Error bars indicate ± 1 SD from mean. The right panel shows average ratings provided by the participants to one item on the questionnaire (how fatigued do you usually feel at the end of the day because of your listening demands?). A rating of 10 on the y-axis indicates greater self-reported difficulty in each area.

CHAPTER IV

DISCUSSION

The two claims that the study was aiming to assess were the advantage for speech understanding in noise provided by SpeechZone 2, as well as the reported reduction in listening effort required to understand speech in noise. These claims were put forth by the manufacturer, and several universities were involved in this pilot study to investigate. It was discovered that neither claim put forth by the manufacturer has significant support from the data collected in this study.

When investigating the reported improvement in speech understanding in the presence of background noise, the HINT test was used with SpeechZone 2 off and then on. HINT sentence scores were compiled to provide an assessment of subjects' speech understanding in noise with and without the processing algorithm, among four degrees of azimuth. While there was a general trend toward improved speech understanding when SpeechZone 2 was enabled, statistical analysis indicated that this trend was not significant overall. Additionally, a trend toward speech recognition was noted when speech was presented from 90 and 270 degrees of azimuth, for both conditions tested. It should be noted that this trend was also not statistically significant and should not be taken as indicative of any benefit provided by the hearing aid processing algorithm. Rather, this improvement in speech recognition from 90 and 270 degrees of azimuth is the demonstrated phenomenon of the head shadow effect. This advantage has been demonstrated in multiple studies (Shaw, 1974; Bronkhorst & Plomp, 1989).

Additionally, the SpeechZone 2 processing algorithm was not shown to have a significant impact on lessening listening effort, in contradiction to claims made by the manufacturer. A comparison of subjects' responses to the questionnaire created for this study showed no significant differences between the conditions for reported concentration, listening effort, speech understanding, or the need

to “give up” listening during the tasks. This indicates that when SpeechZone 2 was enabled, subjects did not perceive a significant difference in the amount of effort they had to expend during the HINT, compared to when SpeechZone 2 was turned off and the speech was processed by omnidirectional microphones alone. A trend toward increased concentration was noted when SpeechZone 2 was turned off, but again this is not a statistically significant finding.

Prior to this study, Unitron had reported some preliminary findings of SpeechZone 2 through white papers and field validation studies. These preliminary findings dealt principally with localization ability with SpeechZone 2, but as discussed in the introduction, localization ability can play a role in speech understanding in background noise. Rule et al. (2017) compared localization abilities of subjects wearing amplification with traditional directional microphones and amplification with Dynamic Spatial Awareness, the algorithm created by Unitron to focus in on the direction of the desired signal in the presence of background noise. Recall that Dynamic Spatial Awareness is one of three components of the SpeechZone 2 algorithm and is concerned in part with restoring natural localization cues that come from the head shadow effect (Figure 10).

Rule et al. (2017) used 30 subjects between 32-87 years old (mean age = 68). A mix of experienced and inexperienced hearing aid users were utilized; however, the definition of an “experienced” hearing aid user was not provided. All subjects had mild to moderate sensorineural hearing loss. Testing was performed in a room with 8 speakers arranged in a circle around the subject, 45 degrees apart. For this study, babble noise was presented at a fixed SNR of -3 dB, meaning that the noise was 3 dB louder than the speech. Subjects were required to perform a localization task, determining where among the 8 speakers the speech was presented. Subjects were not required to perform any word recognition tasks. Subjects were scored on their ability to correctly identify the direction of the speech source, from 0-100%. It was noted that subjects’ localization abilities improved when the speech signal originated from 90 and 270 degrees azimuth, compared to 0 or 180 (Figure 11).

It is known that localization ability can influence speech understanding in noise for individuals with hearing loss (Gardner & Gardner, 1973) but overall this study did not examine the effect of localization on speech recognition in background noise.

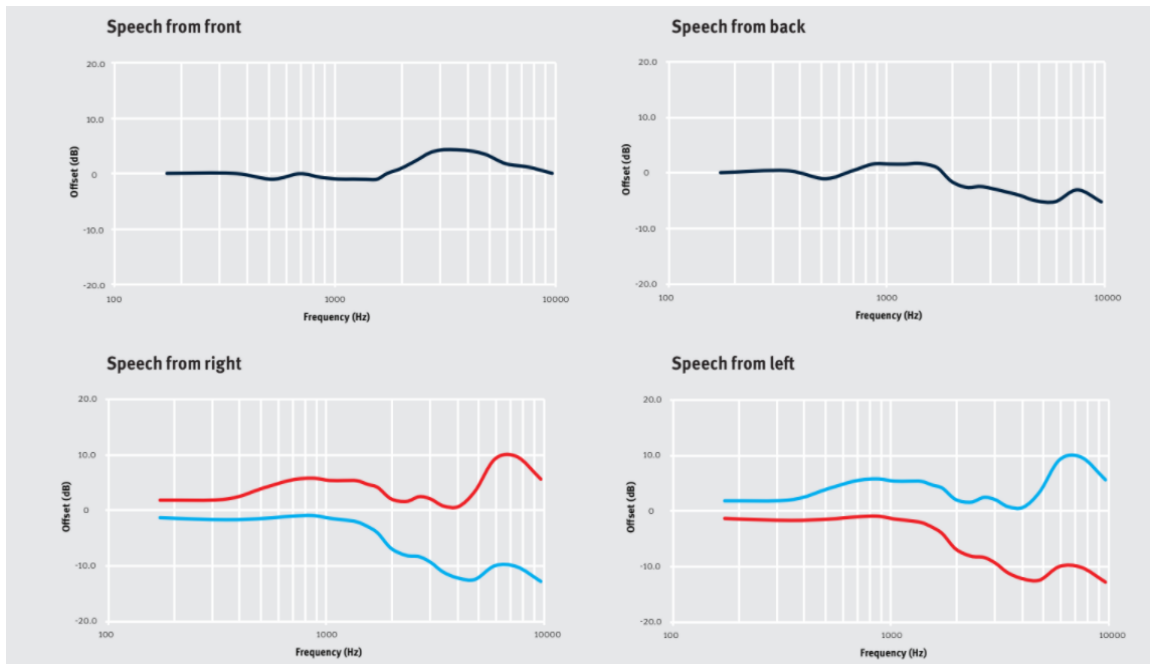


Figure 10. Frequency response from both ears is noted in dark blue, while frequency response from right and left ears individually are noted in red and light blue, respectively. Rule et al. (2017) demonstrates the ability of Dynamic Spatial Awareness to adaptively attenuate the frequency response of the ear contralateral to the source of the desired signal, depending on the direction of speech. Notice that when speech comes from either the front or the back, DSA does little to attenuate the frequency response. However, when speech comes from the right, note the reduction in the response from the left side, and vice versa.

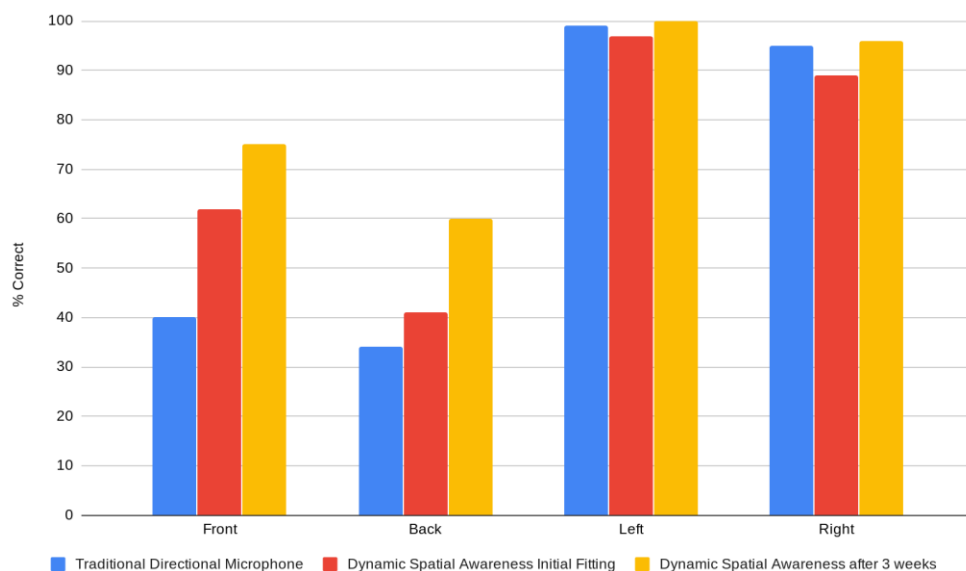


Figure 11. *Adapted from Rule et al. (2017)*. The difference in localization scores (0-100%) for traditional directional microphones vs. Dynamic Spatial Awareness (DSA), over a period of three weeks. Note that in both initial fitting and three weeks post, localization abilities trend toward improvement compared to traditional directional microphones. However, with the omission of error bars, findings cannot be taken as significant.

In contrast to the Rule et al. (2017) study, the current study assessed word recognition in background noise for DSA, as well as listening effort. 7 subjects were used, ranging in age from 25-81 years old (mean = 65). All subjects were experienced hearing aid users, with “experienced” being defined as having worn amplification for at least 4 weeks prior to participating in the study. All subjects had diagnosed sensorineural hearing loss, but the range of losses was quite a bit broader than in the Rule et al. (2017) study, with configurations ranging from mild to moderately-severe or severe.

Four speakers were used to present background noise, rather than eight. This choice was made in part due to the fact that localization tasks were not a part of the study’s rationale; therefore, speech presentation from more specific degrees of azimuth was not crucial to the study. Additionally, rather

than a fixed SNR of -3 dB, the current study used an adaptive SNR due to the nature of the HINT as the vehicle for assessing word recognition ability. A localization task was not used in the current study.

In general, a similar trend to the manufacturers' was observed, in that there was a head shadow advantage when speech was presented at 90 and 270 degrees of azimuth. However, this finding can only be attributed to speech awareness, rather than speech recognition, as the tasks differed between studies.

The current study's results differ from the manufacturer's for several reasons. As previously discussed, the paradigms used differed between studies. Rule et al. (2017) used a localization task to assess the advantage of SpeechZone 2 in focusing on speech signals in a constant level of background noise, while the current study used a word recognition task to assess the advantages of SpeechZone 2 in an adaptive level of background noise. As the current study is part of a pilot investigation to examine the abilities of SpeechZone 2 as a whole, the rationale used to design the studies differs as a consequence. Additionally, the use of four speakers in the current study as opposed to eight resulted in a broader physical separation between the sources of signal and noise in the current study. This may have influenced the results obtained in the current study; however, due to the difference in design this is not currently possible to investigate further.

Based on an extensive search, limited literature exists regarding the effects of SpeechZone 2. References are made to "SpeechPro" in both field validation studies (Sinclair, 2018) and trials with normal hearing individuals (Hayes, 2014); however, it is unclear whether this research is focused on earlier versions of this processing algorithm. Therefore, these findings were not included in the literature review.

Based on the results of the current study, this algorithm does not appear to improve speech understanding in noise or lessen listening effort, both of which were claims by the manufacturer. While no significant results were observed, this study maintains clinical relevance as it demonstrates that

these hearing aids should not be recommended solely for the purpose of attempting to reduce listening effort. While amplification has been shown to assist with lightening the load of concentrating on speech, particularly in noisy environments, it is not advisable at this time to recommend this type of amplification on the strength of the manufacturer's claim that listening effort will be significantly reduced.

While the current study did not find any significant benefit of SpeechZone 2 for speech recognition in the presence of background noise, there are several limitations that may have contributed. First, the subject pool for the current study ($n=7$) is much smaller than for the Rule et al. (2017) study. The power analysis performed with the preliminary data for the current study indicated that a minimum sample size of twelve adults was needed. A larger sample size may have provided more information and produced different results. Additionally, without a localization task implemented in the current study, there is no way to directly compare the results of the Rule et al. (2017) study with the current investigation. Thus, it is difficult to cross-check the manufacturer results and determine some commonality of data.

It is not known whether the use of more speakers, providing a wider variation of degrees of azimuth, would have influenced the results of the study. Additionally, due to the small pool of subject data, degree of hearing loss could not be assessed as a variable in its own right. It is unknown the effects of SpeechZone 2 on word recognition for a certain degree of hearing loss.

Conclusion

1. There was no improvement in the speech reception threshold (HINT score) with the adaptive SpeechZone 2 algorithm in the commercially available Unitron hearing aids. These results are not in agreement with the manufacturer advertisements and product brochures.

2. There was no difference between the omnidirectional program and the adaptive SpeechZone 2 algorithm as measured by the self-reported listening effort and four other related questions (listening related concentration, giving up while listening to speech, perceived speech understanding performance, and overall fatigue).
3. These results are based on a small sample of seven experienced hearing aid users. The findings of this study should be interpreted with caution due to the small sample size.

REFERENCES

- Aniansson, G. (1974). Methods for assessing high frequency hearing loss in every-day listening situations. *Acta Otolaryngologica Supplementum*, 320, 1-50.
<https://doi.org/10.1080/16512251.1974.11675753>
- Baer, T., Moore, B.C.J., & Kluk, K. (2002). Effects of low pass filtering on the intelligibility of speech in noise for people with and without dead regions at high frequencies. *The Journal of the Acoustical Society of America*, 112(3), 1133-1144. <https://doi.org/10.1121/1.1498853>
- Besser, J., Festen, J.M., Goverts, S.T., Kramer, S.E., & Pichora-Fuller, M.K. (2015). Speech-in-speech listening on the LiSN-S test by older adults with good audiograms depends on cognition and hearing acuity at high frequencies. *Ear & Hearing*, 36(1), 24-41.
<https://doi.org/10.1097/AUD.000000000000096>
- Bronkhorst, A.W., & Plomp, R. (1989). Binaural speech intelligibility in noise for hearing-impaired listeners. *The Journal of the Acoustical Society of America*, 86(4), 1374-1383.
<https://doi.org/10.1121/1.398697>
- 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. <https://doi.org/10.1121/1.1408946>
- Demorst, M.E., & Erdman, S.A. (1986). Scale composition and item analysis of the communication profile for the hearing impaired. *Journal of Speech & Hearing Research*, 29(4), 515-535.
<https://doi.org/10.1044/jshr.2904.535>
- Dillon, H. (2001). *Hearing aids*. Sydney: Boomerang Press.

- Drullman, R., Festen, J.M., & Plomp, R. (1994). Effect of temporal envelope smearing on speech reception. *The Journal of the Acoustical Society of America*, 95(2), 1053-1064.
<https://doi.org/10.1121/1.408467>
- 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.
<https://doi.org/10.1121/1.428211>
- 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.
<https://doi.org/10.1121/1.1354984>
- Gardner, M.B., & Gardner, R.S. (1973). Problem of localization in the median plane: Effect of pinnae cavity occlusion. *The Journal of the Acoustical Society of America*, 53(2), 400-408.
<https://doi.org/10.1121/1.1913336>
- Gatehouse, S., & Noble, W. (2004). The speech, spatial, and qualities of hearing scale (SSQ). *International Journal of Audiology*, 43(2), 85-99. <https://doi.org/10.1080/14992020400050014>
- Glasberg, B.R., & Moore, B.C.J. (1989). Psychoacoustic abilities of subjects with unilateral and bilateral cochlear impairments and their relationship to the ability to understand speech. *Scandinavian Audiology Supplementum*, 32, 1-25.
- Hayes, D. (2014). How the LIVElab helped us evaluate the benefits of SpeechPro with speech awareness, accessed online from https://www.unitron.com/content/dam/unitron-2014/documents/english/Crossproduct/whitepaper/RS5372_027-6261-02_LIVElab_White_Paper_ENG.pdf, accessed on 21 March 2020.
- Hornsby, B.W. (2013). The effects of hearing aid use on listening effort and mental fatigue associated with sustained speech processing demands. *Ear & Hearing*, 34(5), 523-534.
<https://doi.org/10.1097/AUD.0b013e31828003d8>

- Hornsby, B.W., & Ricketts, T.A. (2006). The effects of hearing loss on the contribution of high- and low-frequency speech information to speech understanding. II. Sloping hearing loss. *The Journal of the Acoustical Society of America*, 119(3), 1752-1763. <https://doi.org/10.1121/1.2161432>
- Howard, T. (2014) Putting the focus on conversations: Part 3: Speechzone 2, accessed online from https://www.unitron.com/content/dam/unitron-2014/documents/english/moxi-north/whitepaper/027-5894-02_SpeechZone2_Whitepaper_ENG.pdf, accessed on 11 November 2018.
- Klink, K.B., Schulte, M., & Meis, M. (2012). Measuring listening effort in the field of audiology: A literature review of methods, part 1. *Zeitschrift Fur Audiologie*, 51(2), 60-67.
- Koelewijn, T., Zekveld, A.A., Festen, J.M., & Kramer, S.E. (2011). Pupil dilation uncovers extra listening effort in the presence of a single-talker masker. *Ear & Hearing*, 33(2), 291-300. <https://doi.org/10.1097/AUD.0b013e3182310019>
- Kramer, S.E., Kapteyn, T.S., & Houtgast, T. (2006). Occupational performance: Comparing normally-hearing and hearing-impaired employees using the Amsterdam Checklist for Hearing and Work. *International Journal of Audiology*, 45(9), 503-512. <https://doi.org/10.1080/14992020600754583>
- Kramer, S.E., Kapteyn, T.S., Festen, J.M., & Kuik, D.J. (1997). Assessing aspects of auditory handicap by means of pupil dilation. *Audiology*, 36(3), 155-164. <https://doi.org/10.3109/00206099709071969>
- Larsby, B., Hallgren, M., Lyxell, B., Arlinger, S. (2005). Cognitive performance and perceived effort in speech processing tasks: Effects of different noise backgrounds in normal-hearing and hearing-impaired subjects. *International Journal of Audiology*, 44(3), 131-143. <https://doi.org/10.1080/14992020500057244>

Lorenzi, C., Gilbert, G., Carn, H., Garnier, S., Moore, B.C.J. (2006). Speech perception problems of the hearing impaired reflect inability to use temporal fine structure. *Proceedings of the National Academy of Sciences of the United States of America*, 103(49), 18866-18869.

<https://doi.org/10.1073/pnas.0607364103>

McGarrigle, R., Munro, K.J., Dawes, P., Stewart, A.J., Moore, D.R., Barry, J.G., & Amitay, S. (2014).

Listening effort and fatigue: What exactly are we measuring? A British society of audiology cognition in hearing special interest group white paper. *International Journal of Audiology*, 53(7), 433-445. <https://doi.org/10.3109/14992027.2014.890296>

Moore, B.C.J. (1996). Speech perception. In *Cochlear hearing loss: Physiological, psychological, and technical issues* (pp 201-232). Whurr Publishers Ltd.

Moore, B.C.J., Vickers, D.A., Glasberg, B.R., & Baer, T. (1997). Comparison of real and simulated hearing impairment in subjects with unilateral and bilateral cochlear hearing loss. *British Journal of Audiology*, 31(4), 227-245. <https://doi.org/10.3109/03005369709076796>

Moore, B.C.J. (2016). A review of the perceptual effects of hearing loss for frequencies above 3 kHz. *International Journal of Audiology*, 55(12), 707-714.

<https://doi.org/10.1080/14992027.2016.1204565>

Ohlenforst, B., Zekveld, A.A., Jansma, E.P., Wang, Y., Naylor, G., Lorens, A., Lunner, T., & Kramer, S.E.

(2017). Effects of hearing impairment and amplification on listening effort: A systematic review. *Ear & Hearing*, 38(3), 267-281. <https://doi.org/10.1097/AUD.0000000000000396>.

Picou, E.M., Ricketts, T.A., & Hornsby, B.W. (2013). How hearing aids, background noise, and visual cues influence objective listening effort. *Ear & Hearing*, 34(5), 52-64.

<https://doi.org/10.1097/AUD.0b013e31827f0431>.

- Pittman, A.L., & Stelmachowicz, P.G. (2003). Hearing loss in children and adults: Audiometric configuration, asymmetry, and progression. *Ear & Hearing*, 24(3), 198-205.
<https://doi.org/10.1097/01.AUD.0000069226.22983.80>.
- Rule, B., Galvis, C.R., Sinclair, J., & Cornelisse, L. (2017). Going beyond directionality in noisy conversations with dynamic spatial awareness, accessed online from
https://www.unitron.com/content/dam/unitron-2017/images/GovServ2018/PDFs/WhitePapers/RS4200_027-6073-02_Dynamic_Spatial_Awareness_White_Paper_ENG.pdf, accessed on 11 November 2018.
- Shaw, E.A.G. (1974). Transformation of the sound pressure level from the free field to the eardrum in the horizontal plane. *The Journal of the Acoustical Society of America*, 56(6), 1848-1861.
<https://doi.org/10.1121/1.1903522>
- Sinclair (2018). Field trial news from Unitron's audiology centre, accessed online from
https://www.unitron.com/content/dam/unitron-2014/documents/english/Crossproduct/whitepaper/027-6272-02_FieldTrialNews_Spatial%20Awareness_ENG.PDF, accessed on 21 March 2020.
- Skinner, M.W., Karstaedt, M.M., & Miller, J.D. (1982). Amplification bandwidth and speech intelligibility for two listeners with sensorineural hearing loss. *Audiology*, 21(3), 251-268.
<https://doi.org/10.3109/00206098209072743>
- Skinner, M.W., & Miller, J.D. (1983). Amplification bandwidth and intelligibility of speech in quiet and noise for listeners with sensorineural hearing loss. *Audiology*, 22(3), 253-279.
<https://doi.org/10.3109/00206098309072789>
- Stephens, D., & Hetu, R. (1991). Impairment, disability and handicap in audiology: Towards a consensus. *Audiology*, 30(4), 185-200. <https://doi.org/10.3109/00206099109072885>

- Strawbridge, W., Wallhagen, M.I., Shema, S.J., & Kaplan, G.A. (2000). Negative consequences of hearing impairment in old age: A longitudinal analysis. *Gerontologist*, 40(3), 320-326.
<https://doi.org/10.1093/geront/40.3.320>
- Thibodeau, L.M., & van Tasell, D.J. (1987). Tone detection and synthetic speech discrimination in band-reject noise by hearing-impaired listeners. *The Journal of the Acoustical Society of America*, 82(3), 864-873. <https://doi.org/10.1121/1.395285>
- Turner, C.W., & Henn, C.C. (1989). The relation between vowel recognition and measures of frequency resolution. *Journal of Speech and Hearing Research*, 32(1), 49-58.
<https://doi.org/10.1044/jshr.3201.49>
- Turner, C.W., Fabry, D.A., Barrett, S., Horwitz, A.R. (1992). Detection and recognition of stop consonants by normal-hearing and hearing-impaired listeners. *Journal of Speech and Hearing Research*, 35(4), 942-949. <https://doi.org/10.1044/jshr.3504.942>
- Villchur, E. (1977). Electronic models to simulate the effect of sensory distortions on speech perception by the deaf. *The Journal of the Acoustical Society of America*, 62(3), 665-674.
<https://doi.org/10.1121/1.381579>
- Weinstein, B.E., & Ventry, I.M. (1982). Hearing impairment and social isolation in the elderly. *Journal of Speech & Hearing Research*, 25(4), 593-599. <https://doi.org/10.1044/jshr.2504.593>
- Zekveld, A.A., & Kramer, S.E. (2014). Processing load across a wide range of listening conditions. *Psychophysiology*, 51(3), 277-284. <https://doi.org/10.1111/psyp.12151>
- Zekveld, A.A., Kramer, S.E., & Festen, J.M. (2011). Cognitive load during speech perception in noise: The influence of age, hearing loss, and cognition on the pupil response. *Ear & Hearing*, 32(4), 498-510. <https://doi.org/10.1097/AUD.0b013e31820512bb>

APPENDICES

Appendix A: IRB approvals

From: Morgan, Cindy - morgancs
Sent: Friday, February 15, 2019 11:51 AM
To: Compton, Abigail Elizabeth - comptoae (Dukes); Rout, Ayasakanta - routax
Cc: iis.finance
Subject: IRB Extension Approval

Dear Abigail,

I want to let you know that the extension request for your IRB protocol # 18-0329 entitled, "***Speech Understanding & Listening Effort in Noise with a New Speech Processing Algorithm***" has been approved for you to continue your study from 2/20/2019 to 2/19/2020. The signed action of the board form, approval memo, and close-out form will be sent to your advisor via campus mail.

Your Close-Out Form must be submitted within **30 days** of the project end date. If you wish to continue your study past the approved project end date, you must submit an Extension Request Form indicating a renewal, along with supporting information. Although the IRB office sends reminders, it is ultimately ***your responsibility*** to submit the continuing review report in a timely fashion to ensure there is no lapse in IRB approval.

Thank you again for working with us to get your protocol extension approved. We look forward to receiving your project close-out form upon completion of your study.

Best Wishes,
Cindy

Cindy Morgan
IRB Coordinator
Office of Research Integrity - James Madison University
Engineering/Geosciences Bldg., Room 3152
MSC 5738
Harrisonburg, VA 22807
morgancs@jmu.edu
(540) 568-7025

We have moved!! Office of Research Integrity is now located in the Engineering/Geosciences (EnGeo) Bldg., Room 3152.

From: Morgan, Cindy - morgancs
Sent: Tuesday, February 20, 2018 10:01 AM
To: Compton, Abigail Elizabeth - comptoae (Dukes)
Cc: Rout, Ayasakanta - routax; iis.finance
Subject: IRB Approval Notification

Dear Abigail,

I wanted to let you know that your IRB Protocol entitled, "*Speech Understanding and Listening Effort in Noise with a New Speech Processing Algorithm*," has been approved effective from 2/20/2018 through 2/19/2019. The signed action of the board form, approval memo, and close-out form will be sent to you via campus mail. Your protocol has been assigned No. 18-0329. Thank you again for working with us to get your protocol approved.

All research must be conducted in accordance with this approved submission, meaning that you will follow the research plan you have outlined in your protocol, use approved materials, and follow university policies.

Please take special note of the following important aspects of your approval:

- Participants in the study will be offered monetary compensation for their time.
- Any changes made to your study require approval *before* they can be implemented as part of your study. Contact the Office of Research Integrity at researchintegrity@jmu.edu with your questions and/or proposed modifications. An addendum request form can be located at the following URL: <http://www.jmu.edu/researchintegrity/irb/forms/irbaddendum.doc>.
- As a condition of the IRB approval, your protocol is subject to annual review. Therefore, you are required to complete a Close-Out form before your project end date. You *must* complete the close-out form unless you intend to continue the project for another year. An electronic copy of the close-out form can be found at the following URL: <http://www.jmu.edu/researchintegrity/irb/forms/irbcloseout.doc>.
- If you wish to continue your study past the approved project end date, you must submit an Extension Request Form indicating a renewal, along with supporting information. An electronic copy of the close-out form can be found at the following URL: <http://www.jmu.edu/researchintegrity/irb/forms/irbextensionrequest.doc>.
- If there are in an adverse event and/or any unanticipated problems during your study, you must notify the Office of Research Integrity within 24 hours of the event or problem. You must also complete adverse event form, which can be located at the following URL: <http://www.jmu.edu/researchintegrity/irb/forms/irbadverseevent.doc>.

Although the IRB office sends reminders, it is ultimately *your responsibility* to submit the continuing review report in a timely fashion to ensure there is no lapse in IRB approval.

Thank you again for working with us to get your protocol approved. If you have any questions, please do not hesitate to contact me.

Best Wishes,
Cindy

Appendix B: Consent form

Consent to Participate in Research

Identification of Investigators & Purpose of Study

You are being asked to participate in a research study conducted by Abigail Compton from James Madison University. The purpose of this study is to assess the performance of a new hearing aid algorithm when participants are listening to speech in background noise. This study will contribute to the researcher's completion of her Doctor of Audiology dissertation research.

Research Procedures

Should you decide to participate in this research study, you will be asked to sign this consent form once all your questions have been answered to your satisfaction. This study consists of several tests of speech understanding in noise, which be administered to individual participants in HBS 5008. In these tests, you will be asked to listen to and repeat sentences that are presented in various conditions. You will also be given two short questionnaires to rate your preference on a rating scale.

Time Required

Participation in this study will require approximately 2 hours of your time. The testing is broken up into two one-hour long sessions: in the first, your hearing status will be evaluated using standard clinical protocol. A pair of hearing aids will be programmed to fit your exact degree of hearing loss. The second session is comprised of a speech understanding test in four different conditions. You have the option of completing both session in the same day, or returning a different day for the second session.

Risks

The investigator does not perceive more than minimal risks from your involvement in this study (that is, no risks beyond the risks associated with everyday life). A potential risk may include fatigue from the effortful listening required during the speech recognition tasks, however this risk will be minimized by allowing you to take both restroom and water breaks between each of the four listening conditions.

Benefits

The results of this study could potentially be beneficial to the field of clinical audiology by informing us how directional microphones perform in real world situations. The participants will receive a free hearing test and get experience of evaluating a new hearing aid technology.

Incentives

The participant will receive \$10 per hour in financial compensation for participation in this study, leading to a maximum of \$20 in participation fees. The participant will also receive a free hearing test.

Confidentiality

The results of this research will be presented at a conference and at the researcher's doctoral dissertation defense. The results of this project will be coded in such a way that the respondent's identity will not be attached to the final form of this study. The researcher retains the right to use and publish non-identifiable data. While individual responses are confidential, aggregate data will be presented representing averages or generalizations about the responses as a whole. All data will be stored in a secure location in the Hearing Aid Research Laboratory accessible only to the researcher and her research advisor.

Upon completion of the study, all information that matches up individual participants with their answers, including the transcription of the repeated sentences, will be shredded.

Participation & Withdrawal

Your participation is entirely voluntary. You are free to choose not to participate. Should you choose to participate, you can withdraw at any time without consequences of any kind.

Questions about the Study

If you have questions or concerns during the time of your participation in this study, or after its completion or you would like to receive a copy of the final aggregate results of this study, please contact:

Abigail Compton
Communications Sciences and Disorders
James Madison University
comptoe@dukes.jmu.edu

Ayasakanta Rout
Communications Sciences and Disorders
James Madison University
Telephone: 540-568-3874
routax@jmu.edu

Questions about Your Rights as a Research Subject

Dr. David Cockley
Chair, Institutional Review Board
James Madison University
(540) 568-2834
cocklede@jmu.edu

Giving of Consent

I have read this consent form and I understand what is being requested of me as a participant in this study. I freely consent to participate. I have been given satisfactory answers to my questions. The investigator provided me with a copy of this form. I certify that I am at least 18 years of age.

Name of Participant (Printed)

Name of Participant (Signed)

Date

Name of Researcher (Signed)

Date

Appendix C: Mean HINT scores and data analysis excluding outlier

As seen in Figures 7 and 8, participant #2 performed different compared to the rest of the participants. A separate analysis was performed after excluding participant 2 from the raw data. Mean HINT scores for omnidirectional and adaptive with SpeechZone2 are shown in Figure 12 below. Error bars represent standard error. A repeated measure ANOVA showed significant improvement with SpeechZone2 processing compared to omnidirectional microphone mode without SpeechZone2 ($F_{(1,5)} = 20.84, p = 0.006$) when the data from Subject#2 were removed. There was also a significant main effect of speech source location ($F_{(3,15)} = 4.85, p = 0.015$). However, no interaction between processing strategy and speech source location was observed ($F_{(3,15)} = 1.42, p = 0.27$)

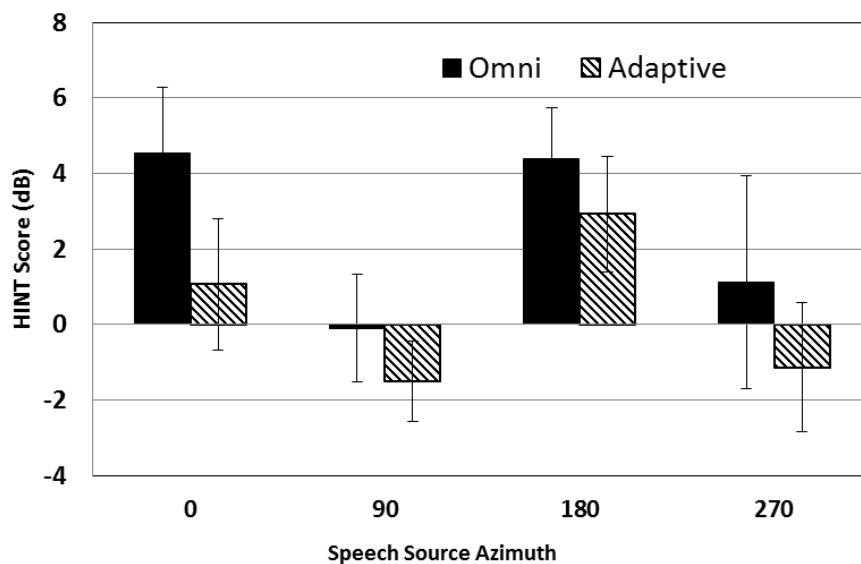


Figure 12. Mean HINT scores without participant 2 for Omnidirectional (solid bars) and Adaptive directional with SpeechZone2 (hatched bars) for different speech source locations (0, 90, 180, and 270 degree azimuth). Error bars indicate ± 1 SE of mean. A lower HINT score indicates better performance.