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The Interaction between Reverberation and Hearing Aid Processing for Speech Perception in Noise

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ABSTRACT

The Interaction between Reverberation and Hearing Aid Processing for Speech Perception in Noise.

Inability to understand speech in the presence of background noise is one of the most common complaints of individuals with hearing impairment. Two hearing aid processing strategies specifically recommended for improved speech perception in noise include: digital noise reduction (DNR) and cognition-based wide dynamic range compression (WDRC) speed. While previous studies have found that these processing recommendations provide benefit under laboratory conditions, that work did not account for reverberation which is present to varying extents in most everyday listening situations, and which disrupts the transmission of important acoustic cues involved in hearing aid signal processing.

The first set of experiments examined acoustic and perceptual effects of DNR under a range of reverberant conditions in older adults with hearing impairment. Participants listened and responded to speech-in-noise processed with simulated reverberation and DNR processing. While DNR processing had minimal effect on speech intelligibility without reverberation, speech intelligibility was substantially lower with DNR processing in reverberant conditions. Unlike in the anechoic condition, listeners reported no subjective benefit with DNR processing in reverberant conditions. These findings were consistent with acoustic analyses showing that acoustic benefit of DNR processing decreased with increasing degrees of reverberation.

The second set of experiments examined acoustic effects of WDRC under a range of reverberant conditions in older adults with hearing impairment and whether listeners benefitted

from different WDRC speeds based on working memory (i.e., cognition-based WDRC speed). Consistent with previous findings, in the absence of reverberation individuals with high working memory performed better with fast-acting WDRC, whereas individuals with low working memory performed better with slow-acting WDRC. However, this effect was diminished in mildly reverberant conditions and eliminated at greater amounts of reverberation. These findings were consistent with acoustic analyses which showed that the acoustic difference between fast-acting and slow-acting WDRC decreases with increasing degrees of reverberation.

These experiments provide evidence that reverberation alters hearing aid signal processing in such a way that may diminish the perceptual benefits of some processing strategies (cognition-based WDRC speed) or even produce adverse perceptual effects (DNR). Overall, these findings suggest that the benefit experienced by listeners in the real world may vary based on the reverberant characteristics of a given listening situation.

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LIST OF ABBREVIATIONS

AAA: American Academy of Audiology

ANOVA: analysis of variance

DCT: Discourse Comprehension Test

DNR: digital noise reduction

IEEE: Institute of Electrical and Electronics Engineers

PTA: pure tone average

RAU: rationalized arcsine units

RST: Reading Span Test

SNR: signal-to-noise ratio

SPL: sound pressure level

T60: reverberation time

WDRC: wide dynamic range compression

DEDICATION

This work is dedicated to the millions of individuals with hearing loss

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CHAPTER 1: INTRODUCTION

There are over 31 million individuals with hearing impairment in the U.S., and hearing impairment is the third most common chronic condition reported by older adults (Cruickshanks et al., 1998; Lethbridge-Çejku & Vickerie, 2004; Kochkin, 2005). Age-related sensorineural hearing impairment (the most prevalent cause of hearing impairment) is typically accompanied by auditory processing deficits, such as broadened auditory filters (Festen & Plomp, 1983, Hopkins & Moore, 2011; Leek & Summers, 1993; Peters & Moore, 1992; Tyler et al., 1984) and reduced temporal resolution (Fitzgibbons & Wightman, 1982; Florentine & Buus, 1984; Hopkins & Moore; 2007, 2011; Lorenzi et al., 2006; Nelson & Freyman, 1987). These processing deficits result in a distorted or incomplete neural transduction of acoustic input signals. Behaviorally, this causes individuals with hearing impairment to suffer impaired speech perception, especially in complex listening environments containing background noise and reverberation (Duquesnoy & Plomp, 1980; Gordon-Salant & Fitzgibbons, 1993, 1995, 1999; Humes & Christopherson, 1991; Nábělek, 1998).

Inability to function in complex listening environments is not only an inconvenience faced by individuals with hearing impairment. It has a cascading impact on their overall health and well-being. The term ‘complex listening environments’ may refer to such real-world environments as restaurants, theaters, family events, and other social occasions in which communication plays a critical role. Thus, the inability to communicate due to difficulty understanding speech may critically undermine the enjoyment of those environments. A recent study conducted in our laboratory surveyed 42 older listeners with hearing impairment to more fully understand how this population communicates in and interacts with these complex listening environments. Results highlight that the overwhelming majority of people surveyed regarded their ability to communicate in complex listening environments, such as the ones listed above, as important or very important (Figure 1; Reinhart & Souza, 2015). Older individuals with hearing impairment are at a disadvantage when tasked with communicating in situations containing background noise and reverberation compared to their normal-hearing peers. Inability to function normally in many social situations can result in a person with hearing impairment feeling disconnected from friends, loved ones, and other systems of social support. Survey results further indicate that individuals who report greater difficulty communicating in complex listening environments are more likely to avoid those important environments (Reinhart & Souza, 2015). In light of these data, it is no surprise that adults with untreated, uncorrected hearing impairment who have more problems communicating are more likely to avoid social situations resulting in higher rates of social isolation (Gopinath et al., 2012; Savikko et al., 2005). Furthermore, this social isolation and deterioration of social wellbeing may lead to higher rates of depression (Huang et al., 2010), and greater risk for dementia (Huang et al., 2010; Lin et al., 2011, 2013) in the hearing-impaired population.

1.1 Hearing Aids

For individuals seeking help with communication in complex listening environments, hearing aids are the most widely distributed rehabilitation device. However, fewer than 23% of the Americans who have difficulty hearing actually use any form of hearing instrument (Kochkin, 1992). One of the most common reasons driving this disparity is that many users do not perceive their hearing aids as performing well in complex listening situations (Kochkin, 2000, 2010, 2011a, 2011b). Furthermore, dissatisfaction with hearing aid performance in complex listening environments is one of the primary reasons reported for discontinuing hearing aid use (Kochkin, 2000; McCormack & Fortnum, 2013). While hearing aids are unlikely to fully compensate for the auditory processing deficits experienced by individuals with hearing impairment, there is ample opportunity to improve hearing aid processing for complex environments which contain noise and reverberation.

Background noise and reverberation both negatively impact speech perception; however, they are acoustically distinct forms of acoustic degradation and produce unique perceptual errors (Nábělek et al, 1989). How are hearing aids optimized for speech-in-noise alone (i.e., absent reverberation)? Currently there are hearing aid prescription recommendations endorsed by the American Academy of Audiology (the principal professional body for audiologists; AAA) to improve listener communication in noise. These prescription recommendations include directional microphones, digital noise reduction, and cognition-based wide dynamic range compression speed which are supported by current research for the purpose of improving listener communication in noise (Valente et al., 2006). The two prescription recommendations for improved speech-in-noise performance of focus for the current study are **digital noise reduction** and **cognition-based wide dynamic range compression speed**.

1.1.1 Digital noise reduction

Digital noise reduction (DNR) is a processing algorithm designed to reduce the adverse effects of background noise on speech. Briefly, DNR operates on the principle that noise and speech are distinguishable on the basis of unique acoustic properties. For example, speech information is dominated by a modulation frequency of approximately 2-4 Hz which corresponds to the physiological limitations of speech articulator movement. That is, speech is produced at an approximate rate of 2-4 phonemes per second. On the other hand, background noise is transmitted at higher modulation frequencies. DNR systems perform real-time acoustic analysis by periodically sampling the input acoustic signal. In each sampling time window the DNR algorithm estimates the presence (or absence) of noise based on some acoustic classification criterion (e.g., modulation spectrum). Then the algorithm estimates the relative level between speech and noise across a number of frequency bands. If noise is determined to be the dominant signal within a frequency band for a given time window, then the DNR processor reduces gain for that frequency region. Using this method, the purpose of DNR is to primarily amplify the speech components while only minimally amplifying the noise components within a combined speech and noise signal. As a caveat, this time-window sampling analysis relies on estimations of the speech and noise power spectra and invariably leads to some misclassification and the introduction of acoustic artifacts (Boll, 1979; Preuss, 1979).

Using DNR processing has been shown to improve the signal-to-noise ratio of a speech-in-noise input (Gustafson et al., 2014). This has the effect of improving speech recognition using automatic speech recognition systems (Hirsch & Ehrlicher, 1995). Typically, an increase in SNR is associated with an improvement in listener intelligibility. However, because DNR processing additionally introduces artifacts by misclassifying and subsequently removing parts of the speech, the improvement in SNR associated with DNR processing is not equivalent to a raw improvement in SNR achieved by adjusting the level of the speech or noise at their sources. As a result of this, DNR has not been demonstrated to improve the intelligibility of speech in noise for listeners and may even cause a slight decrease in intelligibility (Bentler et al., 2008; Brons et al., 2014; Desjardins & Doherty, 2014; Mueller et al., 2006; Ng et al., 2013; Ricketts & Hornsby, 2005; Sarampalis et al., 2009). While DNR processing does not improve listener intelligibility, it has been found to improve speech in noise perception in other ways.

There are a number of subjective benefits associated with DNR processing. Brons et al. (2014) found that DNR reduces the annoyance of background noise in listeners. Mueller et al. (2006) found that listeners were more tolerant of higher levels of background noise measured using the Acceptable Noise Levels test when hearing aid DNR was active. Boymans and Dreschler (2000) found that listeners reported lower aversion to noise in different situations during extended hearing aid trials when using hearing aids with DNR processing compared to without DNR processing. Moreover, DNR processing reduces effortful listening, reducing the cognitive strain of listening under degraded conditions (Desjardins & Doherty, 2014; Gustafson et al., 2014; Ng et al., 2013; Sarampalis et al., 2009). Because of all these benefits, individuals with hearing impairment indicate a strong preference for listening to noisy speech with DNR processing (Brons et al., 2014; Jamieson et al., 1995; Ricketts & Hornsby, 2005).

For all these reasons, it is recommended by AAA to prescribe DNR processing for patients as a means of improving communication in noise (Valente et al., 2006). However, the efficacy of DNR has not been validated under reverberant conditions, such as those that would occur in the real world. Subsequent sections discuss how reverberation affects speech acoustics and consider how that might modulate the functioning and efficacy of DNR processing.

1.1.2 Cognition-based wide dynamic range compression speed

Wide dynamic range compression (WDRC) is the core amplification strategy in modern digital hearing aids. The purpose of WDRC is to balance an individual’s need for signal amplification with the limitations of a reduced dynamic range. This is achieved by applying different amounts of gain based on the level of input, such that low-intensity inputs will receive more gain than high-intensity inputs. The speed at which the processor reacts to changes in input level depends on the attack time and release time parameters. The attack time dictates at what rate gain is decreased in response to increases in input sound level. The release time defines the rate at which gain is increased in response to decreases in input sound level. In consumer hearing aids, attack time values are typically short in order to prevent loudness discomfort by activating rapid decreases in gain following loud impulse noises (e.g., a door slamming); however, there is little consensus on how to set release time values to optimize hearing aid performance.

In clinical hearing aids, WDRC release times range widely, from a few milliseconds to several seconds. In general, short release times provide improved audibility because the compressor rapidly increases gain to lower-intensity sounds. This quick gain restoration places more of the speech cues in an audible range for individuals with hearing impairment (Souza, 2002; Jenstad & Souza, 2005; Henning & Bentler, 2008). However, the rapid gain alterations that occur with short WDRC release times can also take a toll on the temporal characteristics of the signal. The amplitude differences among phonemes in the speech signal (i.e., the temporal envelope) contain important linguistic information regarding manner of articulation, voicing, and prosody (Rosen, 1992). As a result, shorter release times have also been shown to degrade speech recognition compared to longer release times (Stone & Moore, 2004, 2007; Jenstad & Souza, 2005; Reinhart et al., 2016). Additionally, multiple studies have shown that listeners report speech processed with longer release times as more clear (e.g., Hansen, 2002; Reinhart &

Souza, 2016). Given the audibility vs. distortion tradeoff between short and long WDRC release times, there is little consensus on how they should be set in clinical hearing aids.

Over the past decade there has been a proliferation of information regarding the interaction between WDRC speed and listener cognition. That is, a prescribed WDRC speed which may be appropriate for one individual may not be optimal for another due to differences in cognitive function. Many studies have demonstrated that individuals with higher cognitive function perform better in background noise with fast-acting WDRC, whereas individuals with lower cognitive function perform better in background noise with slow-acting WDRC (Gatehouse et al., 2003; Lunner & Sundewall-Thorén, 2007; Lunner et al., 2009; Ohlenforst et al., 2016; Rudner et al., 2009; Souza & Sirow, 2014). The relationship between cognition and WDRC speed has been demonstrated across a range of signal modulation characteristics (Ohlenforst et al., 2016), and has even been demonstrated in a clinical environment with wearable hearing aids (Souza & Sirow, 2014).

The theoretical underpinning of this relationship between cognitive function and WDRC speed can be better understood in the framework of the Ease of Language Understanding Model (Rönnberg et al., 2008; 2013). According to this model, successful language understanding involves both bottom-up auditory processing and top-down cognitive processing. In the initial stages of processing incoming auditory information is processed in the periphery and relayed to higher-order structures (i.e., bottom-up processing). This relayed information is subsequently compared to phonological representation in the linguistic long-term memory. If there is a mismatch between phonological information extracted from the acoustic signal and the phonological representation in the long-term memory, the system invokes explicit cognitive processing resources. The explicit cognitive processing is a cognitive compensatory mechanism which uses semantic context, phonotactic probability, and lexical knowledge (when available) to resolve the mismatch (i.e., top-down processing). However, this cognitive compensatory mechanism is part of a limited-capacity system in which individuals vary in the amount of cognitive resources they have available for this compensatory processing. In summary, both bottom-up and top-down processing occur rapidly and simultaneously when processing a sequence of auditory information to achieve language understanding.

Individuals with hearing impairment are faced with greater task demands when processing speech due to their deficits in bottom-up processing (i.e., effects of hearing impairment). This in turn means that they are more reliant on top-down processing to achieve speech understanding (Pichora-Fuller & Singh, 2006). Individuals with higher cognitive function are considered to have a larger pool of cognitive resources than individuals with lower cognitive function. In the context of the relationship between individual cognitive function and WDRC processing this means that individuals with higher cognitive function have greater cognitive resources to recruit to achieve a lexical match under degraded conditions. Recall that fast-acting WDRC improves signal audibility but places an additional distortion on the temporal envelope of the signal. This allows individuals with higher cognitive function to benefit from the greater signal audibility of speech processed by fast-acting WDRC, despite the associated acoustic degradation (i.e., because they have sufficient cognitive resources to compensate for the signal distortion). On the other hand, individuals with lower cognitive function do not have sufficient cognitive resources to overcome the distortion caused by fast-acting WDRC and thus perform better with slow-acting WDRC.

For these reasons it is recommended that fast-acting WDRC may be beneficial for patients with high levels of cognitive functioning and slow-acting WDRC may be optimal for individuals with lower cognitive function (Valente et al., 2006). However, this relationship among cognitive function, WDRC speed, and benefit in noise has not yet been validated under reverberant conditions, such as those that would occur in the real world. The following sections highlight how the presence of reverberation is an important consideration that may substantially alter this paradigm.

1.2 Effects of Reverberation

Despite the prevalence of these processing recommendations for improved speech perception in noise, poor device performance in noise remains one of the most frequently cited reasons by hearing aid users for discontinuing device use (Bertoli et al., 2009; Kochkin, 2000; McCormack & Fortnum, 2013). One possibility is that listeners may not receive as much benefit as anticipated from these hearing aid processing strategies when employed under realistic listening conditions. Further research is necessary to verify the benefits of hearing aid processing strategies for speech perception in noise under a wide range of realistic listening conditions. While the benefits of DNR and cognition-based WDRC speed have been well described, these benefits have thus far been constrained to laboratory conditions. Previous studies have overwhelmingly neglected to acknowledge the effects of real room acoustics, and thus their real-world generalizability is limited.

1.2.1 Acoustic effects of reverberation

Real rooms unavoidably contain some amount of reverberation which additionally distorts the signal acoustics (Table 1). Reverberation refers to the reflection of acoustic energy off of features in an environment which causes acoustic energy to arrive to listeners' ears substantially delayed in time relative to the direct sound. A common way to quantify reverberation is to measure the time required, in seconds, for the reflected acoustic energy to decrease by 60 dB after the offset of the source signal (T60). Nábělek et al. (1989) describe the effects of reverberation on ongoing signals, such as speech, as a function of two main components: self- and overlap-masking. Self-masking refers to the degradation occurring within each phoneme by reverberation. Overlap-masking refers to the degradation that occurs when acoustic information from previous phonemes spills over into the subsequent speech components. Both self- and overlap-masking combine to degrade the spectral and temporal cues important for speech perception. See Figure 2 for a visualization of the acoustic effects of reverberation in both the spectral and temporal domains for the nonsense syllable /ata/. Note in the "reverberation time = 0.0 seconds" conditions the absence of energy during the closure of the stop consonant /t/; however, as reverberation is introduced to the signal, energy from the previous vowel phoneme begins to carry over into the stop closure. Overall, reverberation has been found to degrade the signal acoustics by flattening the temporal envelope and attenuating the high frequency modulation energy (Hawkins & Yacullo, 1984; Reinhart et al., 2016). These acoustic effects have a detrimental effect on speech perception, especially for individuals with hearing impairment.

1.2.2 Perceptual effects of reverberation in the hearing-impaired auditory system

Reverberation alone significantly hinders speech understanding in older listeners with hearing impairment (Duquesnoy & Plomp, 1980; Gordon-Salant & Fitzgibbons, 1993, 1995, 1999; Humes & Christopherson, 1991; Nábělek, 1998). A study by Reinhart and Souza (in press) investigated the listener factors of individuals with hearing impairment associated with speech intelligibility in reverberant conditions. The factors investigated were degree of hearing loss, age, temporal envelope sensitivity, and working memory capacity. These factors were investigated in relation to listeners' speech intelligibility at three levels of reverberation: no reverberation (reverberation time = 0.0 second), moderate reverberation (reverberation time = 1.0 second), and severe reverberation (reverberation time = 4.0 seconds) in thirty-three participants. The results were that in the no reverberation condition, temporal envelope sensitivity was the only significant listener factor associated with speech intelligibility. When speech was degraded by moderate reverberation, both age and degree of hearing impairment were significantly associated with speech intelligibility. Lastly, listener working memory capacity and age were both significantly associated with speech intelligibility in the severe reverberation condition. Overall, these results suggest that individuals with hearing impairment are substantially more susceptible to the effects of reverberation than normal hearing listeners due to suprathreshold processing deficits. In fact, it is suggested that reverberation disrupts speech understanding even at relatively short reverberation times of 0.4 seconds (Gordon-Salant & Fitzgibbons, 1999). Thus, reverberation will be of real-world significance across a range of everyday noisy listening situations (Table 1).

1.3 Reverberation and Hearing Aid Signal Processing

It is important to understand that the effects of reverberation are not merely additive when it comes to considering how hearing aids may process noisy speech in reverberant environments. Instead, reverberation may diminish the efficacy of prescription recommendations for improved speech-in-noise performance. For example, it is known that the presence of reverberation decreases the benefit of adaptive directional microphone technology (Ricketts, 2000; Ricketts & Henry, 2002). It is hypothesized that this occurs because reverberation results in the spatial diffusion of sound energy from the source location due to the acoustic reflections. Thus, reverberation decreases the ability of the adaptive processor to spatially identify the sound source. Due to the acoustic effects of reverberation on noisy speech acoustics, we hypothesize that the presence of reverberation may reduce or negate the benefit of other prescription recommendations for improved speech-in-noise performance, namely DNR and cognition-based WDRC speed. The next sections more fully describe these hypotheses.

1.3.1 DNR and reverberation hypothesis

Recall that DNR operates by completing a time-window analysis of the acoustics of an incoming signal to accurately identify speech and noise components. Because reverberation causes a delay of acoustic energy, this will cause the spectral and temporal content of both the speech and noise to ‘bleed over’ into subsequent sampling time windows (Figure 2). This spread of energy will likely decrease the ability of the processor to make valid assessments of the speech vs. noise balance across frequency bands. This in turn would lead to a decrease in noise suppression and increase in artifact rate. For these reasons it is predicted that the presence of reverberation will reduce the benefits of DNR for noisy speech perception. Moreover, we predict that the amount of reverberation will interact with DNR, such that more reverberation will lead to a decreased benefit of DNR due to greater acoustic 'bleed over' from time window to time window.

1.3.2 Cognition-based WDRC speed and reverberation hypothesis

The cognition-based WDRC speed recommendation states that individuals with higher cognitive function will likely have higher speech intelligibility in noise with fast-acting WDRC, whereas individuals with lower cognitive function will perform better with slow-acting WDRC. However, previous research has indicated that this interaction is contingent upon the modulation of the noise masker. That is, when the noise is unmodulated the cognition-based benefit of different WDRC processing is absent (Lunner et al., 2009; Rudner et al., 2009). Because reverberation 'smears' the signal acoustics to reduce the modulation depth of a signal (Reinhart et al., 2016), modulated noise in reverberation becomes more like unmodulated noise (Figure 2). For these reasons it is predicted that the presence of reverberation will reduce the benefits of this cognition-based WDRC speed recommendation for noisy speech perception. Similar to the DNR and reverberation prediction, it is predicted that larger amounts of reverberation will lead to a greater reduction in the cognition-based WDRC speed benefit. This is because greater amounts of reverberation will likely decrease the modulations of the noise masker which are necessary for this cognition-based WDRC speed interaction to occur.

1.4 Current Studies

These potential relationships between reverberation and hearing aid processing recommendations (i.e., either DNR or cognition-based WDRC speed) may substantially impact how clinicians use these features in treating patients. Clinicians currently adopt these features expecting their patients to benefit and even tell the patient that they will help (e.g., reduced listening effort, increased listening comfort, and intelligibility). However, these strategies may not be providing the full benefit expected by clinicians due to the effects of reverberation. This discrepancy between expectation and reality may lead to patient dissatisfaction with their devices' performance in noise and even discontinuation of hearing aid use. Moreover, an understanding of these issues is necessary to stimulate development of algorithms that can cope with noise in the presence of reverberation.

The current study addresses the interactions between the two hearing aid prescription recommendations for improved speech-in-noise perception (DNR and cognition-based WDRC speed) with reverberation. The approach utilizes acoustical and behavioral analyses to comprehensively evaluate the underlying mechanics and clinical implications of the interaction between reverberation and hearing aid processing. The overarching goal of the current study was to investigate the efficacy of these prescription recommendations for improved speech-in-noise performance under realistic listening situations with reverberation. In doing so, the knowledge gained intended to fill a critical gap in the literature that has thus far neglected the likely role of reverberation and room acoustics on hearing aid processing. If clinicians have a better understanding of how the benefits of DNR and cognition-based WDRC speed may (or may not) uphold under the acoustic effects of reverberation, then they will be better able to understand the benefits of these features in the real world and counsel patients accordingly to increase satisfaction.

CHAPTER 2: REVERBERATION AND DIGITAL NOISE REDUCTION: ACOUSTIC AND BEHAVIORAL EFFECTS

2.1 Abstract

Digital noise reduction (DNR) processing is used in hearing aids to enhance perception in noise by classifying and suppressing the noise acoustics. However, the efficacy of DNR processing is not known under reverberant conditions where the speech-in-noise acoustics are further degraded by reverberation. The purpose of this study was to investigate acoustic and perceptual effects of DNR processing across a range of reverberant conditions for individuals with hearing impairment.

Twenty-six listeners with mild-to-moderate sensorineural hearing impairment were tested under varying degrees of reverberation with and without DNR processing. Speech stimuli were combined with unmodulated broadband noise at several signal-to-noise ratios (SNRs). A range of reverberant conditions with realistic parameters were simulated, as well as an anechoic control condition without reverberation. Reverberant speech-in-noise signals were processed using a spectral subtraction DNR simulation. Improvement in SNR as a result of DNR processing was quantified using a phase inversion technique. Sentence intelligibility and subjective ratings of listening effort, speech naturalness, and background noise comfort were examined with and without DNR processing across the conditions.

Improvement in SNR was greatest in the anechoic control condition and decreased as the ratio of direct to reverberant energy decreased. There was no significant effect of DNR processing on speech intelligibility in the anechoic control condition, but there was a significant decrease in speech intelligibility with DNR processing in all of the reverberant conditions. Subjectively, listeners reported greater listening effort and lower speech naturalness with DNR

processing in some of the reverberant conditions. Listeners reported higher background noise comfort with DNR processing only in the anechoic control condition.

Results suggest that reverberation affects DNR processing in such a way that decreases the ability of DNR to reduce noise without distorting the speech acoustics. Overall, DNR processing may be most beneficial in environments with little reverberation, and that the use of DNR processing in highly reverberant environments may produce adverse perceptual effects.

2.2 Introduction

Hearing aid performance in situations with background noise remains one of the most common complaints among hearing aid users. Many listeners report that their hearing aids merely amplify the background noises, potentially causing loudness discomfort (Kochkin, 2000). In response to this, digital noise reduction (DNR) algorithms have been developed to improve the amplification of speech-in-noise signals by hearing aids. Briefly, DNR operates on the principle that while hearing aids receive a combined speech-in-noise input, speech and noise are acoustically distinct. Therefore, it is possible to estimate the speech and noise signals from the combined input using a time-variant sampling and classification of the input signal. Combined speech-in-noise inputs are decomposed across a number of frequency channels, and if the estimated power of the noise is greater than that of the speech, then gain is reduced within that channel (Levitt, 2001; Bentler & Chiou, 2006). Using this process DNR improves the long-term signal-to-noise ratio (SNR) of a speech-in-noise input (Gustafson et al., 2014).

While these algorithms have become increasingly advanced, the reliance on estimations of the speech and noise invariably leads to some misclassification of the signals (i.e., speech misclassified as noise). This misclassification introduces acoustic artifact which degrades the speech information (Boll, 1979) and occurs to a greater extent when the speech and noise are acoustically similar (Arehart et al. 2003). As such, behavioral research has overwhelmingly demonstrated either no change in intelligibility or even a slight decrease in intelligibility with DNR processing (Alcántara et al. 2003; Ricketts & Hornsby, 2005; Mueller et al. 2006; Bentler et al. 2008; Sarampalis et al. 2009; Ng et al. 2013; Desjardins & Doherty, 2014). Thus, the improvement in SNR observed with DNR processing is not equivalent to a direct improvement in SNR by adjusting the speech or noise levels at their sources.

While DNR processing is no longer expected to improve speech intelligibility in noise, research has identified other perceptual benefits associated with DNR processing. Previous studies have found that listeners experience a higher tolerance for noise and report increased comfort in noisy listening situations when listening with DNR processing (Boymans & Dreschler, 2000; Mueller et al. 2006; Bentler et al. 2008). By suppressing the amplification of noise, DNR addresses one of the core complaints of hearing aid wearers: that background noise is often amplified to uncomfortably loud levels. As a result, research has shown that listeners prefer listening in noise with DNR processing due to increased background noise comfort (Jamieson et al. 1995; Ricketts & Hornsby, 2005; Brons et al. 2014). While background noise comfort can be maximized with aggressive processing parameters which suppress the noise, this may also lead to increased misclassification and distortion of the speech which would have a substantial impact on speech understanding and perceived naturalness. Previous work has identified that listeners prefer DNR when it provides optimal background noise comfort without substantially decreasing speech understanding and perceived speech naturalness (Brons et al. 2013, 2014).

DNR processing is also associated with decreased listening effort in noise. Listening effort is broadly defined as the cognitive resources required for speech recognition, with greater resources being expended during more effortful listening (Hicks & Tharpe, 2002). Listening effort is higher in listening situations with competing auditory signals (e.g., background noise), because listeners are required to utilize top-down processing to inhibit or suppress the non-desired, noise signals and focus on the speech (Rönnberg et al. 2008, 2013; Stenfelt & Rönnberg, 2009). When DNR processing is active, the hearing aid assists in suppressing the noise before the combined signal is relayed to the auditory system. In doing so, DNR may make available cognitive resources that would otherwise be dedicated towards effortful noise suppression. In support of this hypothesis, previous studies have demonstrated that listeners experience enhanced dual-task performance (Sarampalis et al. 2009; Desjardins & Doherty, 2014), more rapid decoding of the speech signal (Gustafson et al., 2014), and improved consolidation of information for later recall (Ng et al. 2013) when listening under highly unfavorable conditions with DNR processing. In summary, so long as DNR is able to suppress the noise while minimizing acoustic artifact and speech distortion, then DNR processing may decrease listening effort when perceiving noisy speech.

Given these perceptual benefits, DNR is a ubiquitous feature available in commercial hearing aids (Bentler & Chiou, 2006). Despite the prevalence of DNR, poor hearing aid performance in noisy situations remains one of the most frequently cited reasons for discontinuing device use and hearing aid non-adoption (Kochkin, 2007; Bertoli et al. 2009; McCormack & Fortnum, 2013). While the benefits of DNR have been relatively well researched and validated under laboratory conditions, listeners are still often unhappy with device performance in realistic listening situations.

A potential reason for this disparity may be that realistic listening situations frequently contain reverberation which may interact with DNR processing. Reverberation refers to the reflection of acoustic energy off features in an environment which causes a portion of energy to arrive at listeners substantially delayed in time relative to the direct energy. This late-arriving energy causes the spectral and temporal contents of the signal received by the hearing aid microphone to be smeared across time (Nábělek et al. 1989; Reinhart et al. 2016). Recall that DNR must be able to accurately identify the speech and noise signals within a channel using time-varying estimations of the input signal. Any factor that disrupts the ability of DNR processing to validly distinguish the speech from noise will likely increase the speech distortion caused by DNR. Because reverberation causes a smearing of acoustic energy, this spread of energy will potentially decrease the ability of the DNR algorithm to accurately distinguish the time-varying speech and noise signals. As a result, reverberation may cause decreased noise suppression and increased artifact rates relative to processing of anechoic signals. For these reasons, we hypothesize that reverberation reduces the benefits of DNR for noisy speech perception. Because hearing aids perform DNR processing in environments with varying reverberation, the interaction between DNR processing and reverberation is an important consideration for generalizing the effects of DNR to the real world.

The purpose of this experiment was to evaluate the effects of DNR processing on speech-in-noise across a range of simulated, reverberant environments. To explore the potential interaction between DNR processing and reverberation we examined the effects of DNR on signal acoustics, speech intelligibility, and subjective ratings including perceived listening effort, speech naturalness, and background noise comfort. Based on the extant literature, we predicted that without reverberation DNR would improve the SNR. Furthermore, in the absence of reverberation we expected either no difference or a small decrease in intelligibility and subjective speech naturalness with DNR processing, as well as subjective improvements in listening effort and background noise comfort. Lastly, we predicted that these benefits of DNR would decrease or be eliminated under varying amounts of reverberation.

2.3 Methods

2.3.1 Participants

Twenty-six older adults with sensorineural hearing impairment participated in the study (mean age = 73.0 years, range 60 to 85 years; 16 males, 10 females). Air conduction thresholds were measured at 250-8000 Hz octave frequencies and inter-octaves at 3000 and 6000 Hz. Bone conduction testing was performed at octave frequencies 250-4000 Hz. Participants presented with no more than a single air-bone gap ≥15 dB. Participants had symmetrical hearing loss defined as no more than a 10 dB difference in pure-tone average (thresholds 500, 1000, 2000 Hz) between ears. Mean participant audiogram for both ears are depicted in Figure 3. The Northwestern University Institutional Review Board approved all study procedures. Participants completed an informed consent process prior to participation, and they were compensated for their time.

2.3.2 Speech stimuli

Sentences: Sentence stimuli from the IEEE corpus (IEEE, 1969) were used in the current study. Overall, these sentences are low-context (e.g., “The rope will bind the seven books at once”). Sentences spoken by a single male talker from the Inland North dialect area were locally-recorded in order to control for differences in regional dialect (McCloy et al. 2015). Sentence stimuli were selected to measure speech intelligibility because they are more indicative of real-world listening than isolated speech segments, such as nonsense syllables.

Story passages: Story passages from the Discourse Comprehension Test (DCT; Brookshire & Nicholas, 1993) were also used in the current study. Passages from the DCT corpus were used because they are age-appropriate for adults and passages are controlled for length, grammatical complexity, listening difficulty, and information content. Story passages were selected to measure subjective ratings because they are longer stimuli that have the advantage of providing the listener a longer auditory sample with which to form subjective judgments. Additionally, the length of the story passages would more fully capture the dynamic, time-varying effects of the DNR processor than shorter stimuli. This would better reflect the cumulative effects of DNR processing on listener perception in the real world.

2.3.3 Stimulus processing

Sentence and story passage stimuli were processed in three stages: (1) combined with noise, (2) convolved with simulated reverberation (3) processed with simulated hearing aid DNR.

Noise: Both sentence and story passage stimuli were first combined with unmodulated broadband noise. The broadband noise had a flat spectrum from 0-20,000 Hz. This noise was used because DNR may not be engaged by more speech-like noises due to the acoustic similarities between the speech and noise (Bentler et al. 2008). The noise preceded the onset of the speech stimuli by 2.0 seconds to provide the DNR processor with an initial noise estimate. On the basis of pilot testing to avoid floor and ceiling effects when combined with reverberation, the background noise was added at 3 and 8 dB SNR for the sentence stimuli and at 8 dB SNR for the story passages. In addition to avoiding floor and ceiling effects, these SNRs represent the range of typical SNRs in everyday listening environments (Hodgson et al., 2007). The speech level was fixed at 65 dB SPL and the noise level varied to yield the final SNR levels.

Reverberation simulation: The second stage of signal processing was to process the speech-in-noise signals using a reverberation simulation. Reverberation was simulated using a Matlab-based program developed and validated by Zahorik (2009) which produced binaural room response simulations. Briefly, the simulation used a three-dimensional image model (Allen & Berkley, 1979) to simulate early specular reflections within a hypothetical room. The simulated reflections were a computation of the directions, delays, and attenuations of different sound energy reflections across the frequency spectrum. The direct-path and 500 early reflections were spatially rendered using non-individualized head-related transfer functions. Late response modeling was constructed for each simulated ear using independent Gaussian noise samples. Separate decay functions were applied to six octave-bands ranging from 125-4000 Hz derived from the Sabine equation (Equation 1.1; Sabine, 1922) to estimate the time required, in seconds, for the reflected acoustic energy to decrease by 60 dB after the offset of the source signal (T60):

$$T_{60} = 0.163 \frac{V}{S\bar{\alpha}_i}, \qquad \textit{Equation (1.1)}$$

This equation calculated this estimation from the parameters of room volume (*V*), total surface area of the reflecting surfaces (*S*), and the average absorption coefficient for all surfaces (α_i). The simulation method allowed for the manipulation of these factors to vary the simulated reverberation conditions. Overall, this method provided experimental control and flexibility while producing reverberant signals that are accurate physical and perceptual approximations of those measured in real rooms (Zahorik, 2009).

In the current simulation, source-listener distance was fixed, while the room size and absorptive properties of the reflective surfaces were varied to produce a range of reverberation conditions. Source-listener distance was fixed at 1.4 m which is a typical conversational distance. Both the speech and noise signals had the same source position. In the real world, larger rooms tend to produce longer reverberation times. To reflect this, room size was increased incrementally with increasing reverberation time conditions. In total, four room conditions were simulated exploring a range of reverberation conditions: one anechoic control condition (without any reverberation) and three experimental conditions (containing various amounts of reverberation). See Table 2 for a summary of these conditions and the reverberation parameters.

Two metrics were used to quantify the degree of reverberant degradation: reverberation time and clarity index. Reverberation time was calculated as the time required for the impulse response to decay by 60 dB relative to its initial level averaged across octave bands from 125-4000 Hz. Longer reverberation times are typically indicative of greater reverberant degradation because reverberant energy persists for a longer duration. Clarity index was calculated as the logarithmic ratio of direct sound and early reflections arriving within the first 80 ms to the late reflections arriving after 80 ms (e.g., Martellotta, 2010). A higher clarity index is expected to yield higher perceived clarity because there is less reflected energy to mask out the direct energy from subsequent speech portions. Both duration (reverberation time) and density (clarity index) quantifications may be important measures when it comes to how the reverberation will affect listener perception and interact with hearing aid processing.

DNR simulation: The third and final stage of signal processing was to process the reverberant speech-in-noise signals through a DNR amplification simulation. DNR was simulated using a Matlab-based hearing aid simulation developed by Kates (2008). The method uses a spectral subtraction DNR algorithm (Boll, 1979) which is a family of related algorithms widely implemented in commercial hearing aids. Both channels of the reverberant speech-in-noise signal were separately processed by the DNR simulation, similar to a bilateral hearing aid fit independently applying signal processing. Briefly, the algorithm used an adaptive procedure described by Arslan et al. (1995) which operates in two stages: (1) estimate the noise spectrum across frequency bands and then (2) use the estimated SNR in frequency bands to control a time-varying gain to subtract or attenuate the estimated noise spectrum from noisy speech.

The initial noise spectrum estimate was calculated in a 100 ms time window at the beginning of the digital file. The noise spectrum estimate was then continuously updated in 50 ms time windows in which the incoming signal was windowed and the short-time Fast Fourier transform was computed from the windowed data sequence. To prevent the adaptive noise estimate from fluctuating too rapidly, updated noise estimates could not exceed 1.006 times the previous estimate or be smaller than .978 times the previous estimate. Next, the speech envelope was estimated using a peak detector with an attack time of 3 ms and release time of 50 ms. The instantaneous SNR was calculated across frequencies using the current noise estimate and the speech estimate obtained from the peak detector. If the speech envelope was greater than the noise estimate in a frequency channel, then the noise estimate was incremented at a rate of 10 dB/sec in that frequency region. Conversely, if the speech envelope was estimated to be less than the noise in a frequency region, then the noise estimated was decremented at a rate of -25 dB/sec at that frequency region. In the current simulation the maximum allowed speech attenuation was set to 10 dB which is within the range of attenuation values typically used in commercial hearing aids (Kates, 2008). Lastly, stimuli were bandpass filtered from 250-6000 Hz to represent the typical hearing aid receiver bandwidth.

2.3.4 Acoustic measure

The changes to the reverberant speech-in-noise signals as a result of DNR processing were measured as the change in SNR (i.e., the difference between output SNR following DNR amplification and input SNR). Change in SNR was quantified using the Inversion Method (Hagerman & Olofsson, 2004) which has previously been used to quantify the acoustic effects of DNR in hearing aids (Gustafson et al., 2014). This method used signal phase cancellation to isolate speech and noise signals following processing in order to compute and compare the relative root mean square values. In order to isolate speech and noise signals, three versions of the speech-in-noise signals were processed by the DNR simulation: (1) a phase-normal speech and noise combination ($S_N N_N$); (2) a phase-normal speech and a phase-inverted noise signal ($S_N N_I$); (3) a phase-inverted speech and a phase-normal noise ($S_I N_N$) signal (Equation 1.2) where S and N are the digital speech and noise signals, respectively, after setting the initial SNR levels and being processed with any reverberation.

$$\mathbf{S_N N_N = S + N}$$
$$\mathbf{S_N N_I = S + (N \times -1)} \qquad \textit{Equation (1.2)}$$
$$\mathbf{S_I N_N = (S \times -1) + N}$$

After these signals were processed with DNR, different combinations (Equation 1.3) were used to extract either the processed speech (S_P) or processed noise (N_P) from the combined-processed stimulus.

$$\mathbf{S_P} = \frac{\mathbf{S_N N_I + S_N N_N}}{\mathbf{2}} \qquad \textit{Equation (1.3)}$$
$$\mathbf{N_P} = \frac{\mathbf{S_I N_N + S_N N_N}}{\mathbf{2}}$$

After isolating the processed speech and noise signals, the root mean square values were calculated only in the sampling window where speech was present (i.e., ignoring the effects of DNR on noise preceding and following the speech). Isolated speech and noise signals were then divided by 2 to correct for the doubling of the phase-normal signal. The processed SNR (SNR_P) in dB was then calculated comparing the log ratio of the processed speech (S_P) to the processed noise (N_P) (Equation 1.4).

$$\mathbf{SNR_P} = \mathbf{20log_{10}}\left(\frac{\mathbf{S_P}}{\mathbf{N_P}}\right) \qquad \textit{Equation (1.4)}$$

Lastly, change in SNR was calculated as the difference between the initial SNR of the reverberant speech-in-noise signal (either 3 or 8 dB) and the output SNR of the signal following DNR processing.

2.3.5 Behavioral measures

Speech intelligibility: Speech intelligibility scores were obtained in each of the 16 test conditions (two SNRs, four reverberation conditions, and either with or without DNR processing) using 160 sentence stimuli (10 per condition). Speech intelligibility testing was conducted in a double-walled sound booth. Digital signals were converted to analog by Tucker-Davis Technologies equipment and played through Etymotic-ER2 insert phones. Sentences were presented binaurally. The speech presentation was fixed at 65 dB SPL to represent a conversational level of speech. With the addition of noise, final presentation levels were calibrated to either 66.8 dB SPL (for the 3 dB SNR condition) or 65.6 dB SPL (for the 8 dB SNR condition). Listeners received individual NAL-NL1 shaping for each ear to mimic the individualized frequency shaping provided by wearable hearing aids (Byrne et al., 2001). Sentences were presented and scored using custom-developed Matlab code. Each sentence contained five key words for scoring (e.g., "The rope will bind the seven books at once"). Final speech intelligibility was recorded as the percent of words correctly repeated within a given condition (50 key words per condition).

Subjective ratings: Subjective ratings were obtained in eight test conditions (four reverberation conditions, each with/without DNR) using nine passages from the DCT (1 per condition + a practice story). Due to the limited number of story passages, we only examined the effects of DNR at 8 dB SNR. Story listening was performed in a double-walled sound booth. Digital signals were converted to analog by M-Audio sound card and played through Sennheiser HD25 supra-aural headphones. Stories were presented binaurally. Similar to the sentence task, speech presentation was fixed at 65 dB SPL with the noise added for a final presentation level of 65.6 dB SPL for the 8 dB SNR condition plus NAL-NL1 frequency shaping based on individual audiograms for each ear (Byrne et al., 2001).

Following presentation, listeners were asked to rate three subjective aspects of the story listening experience: (1) how much overall listening effort it took for them to understand the story, (2) how natural the speech of the story was, (3) and how comfortable the background noise was while listening to the story. The ratings were obtained on 7-point scales (see Appendix for scales) similar to those which have been previously used in the hearing literature (e.g., Johnson et al., 2015). Comprehension scores were recorded using the original questions from the DCT. Stories were presented and listeners made their ratings using custom-made Matlab code. Listeners completed one practice story in quiet to acclimate them to the test procedure/interface and storytelling narrative structure. Subjective listening effort, speech naturalness, and background noise comfort in noise measures were recorded as the rating (1-7 scale) assigned after listening to a story by a participant for a given condition.

2.4 Results

2.4.1 Acoustic results

See Figure 4 to view the change in SNR as a result of DNR processing for the different reverberation room conditions and different input SNRs. A two-way analysis of variance (ANOVA) test was conducted with Input SNR and Room Condition as fixed, independent variables and change in SNR as the dependent variable. There was a statistically significant main effect of Input SNR [$F(1,392)=280.013$, $p<0.001$, partial $\eta^2=0.417$] in which the change in SNR as a result of DNR processing was greater at the lower input SNR. There was also a statistically significant main effect of room condition [$F(3,392)=64.622$, $p<0.001$, partial $\eta^2=0.331$]. The interaction between room condition and input SNR was not statistically significant [$F(3,392)=0.634$, $p=0.593$, partial $\eta^2= 0.005$].

To further explore the main effect of room condition, pairwise comparisons with a Bonferroni correction were conducted. Change in SNR was highest in the control room condition compared to any of the reverberant room conditions ($p<0.001$). Additionally, the change in SNR in room condition 2 was significantly lower than in either room condition 1 or room condition 3 ($p<0.001$). There was no significant difference between room condition 1 and room condition 3 ($p=0.228$).

2.4.2 Speech intelligibility results

Raw intelligibility scores were first transformed to rationalized arcsine units (RAU) to normalize the variance near floor and ceiling (Studebaker, 1985). See Figure 5 to view the transformed speech intelligibility scores for all the conditions. Data were analyzed using a three-way repeated-measures ANOVA with Room Condition (Control, 1, 2, 3), Input SNR (3, 8), and DNR (unprocessed, DNR) as within-subject variables. All assumptions of the model were met. The three-way interaction was not significant [$F(3,72) = 0.559$, $p=0.644$, partial $\eta^2 = 0.022$].

There was a significant Room Condition × SNR interaction [$F(3,72) = 2.772$, $p=0.047$, partial $\eta^2 = 0.100$]. To further examine this interaction, separate one-way repeated-measure ANOVAs were conducted with Room Condition as within-subjects variable at each Input SNR condition. Main effects of Room Condition were significant for both 3 dB [$F(3,72)=131.060$, $p<.001$, partial $\eta^2 = 0.784$] and 8 dB [$F(3,72)=90.812$, $p<.001$, partial $\eta^2 = 0.840$]. Post-hoc comparisons with a Bonferroni correction for both input SNRs revealed that intelligibility was highest in the control condition (all $p<.05$) and poorest in room condition 2 (all $p<.05$). There was no significant difference between room conditions 1 and 3 in either input SNR condition (both $p>.05$). Overall, the effects of room condition on speech intelligibility were similar in the 3 and 8 dB input SNRs; however, there was a slightly larger effect of reverberation time in the 8 dB SNR condition than in the 3 dB SNR condition.

There was also a significant Room Condition × DNR interaction [$F(3,72)=2.781$, $p=.046$, partial $\eta^2= 0.101$]. To further examine this interaction, data were transformed to examine the difference in speech intelligibility for unprocessed vs. DNR conditions for each room condition. Transformed data are depicted in Figure 6. One sample t-tests were conducted for each room condition using a Bonferroni correction to examine whether there was a significant decrement in speech intelligibility with DNR processing. The difference in speech intelligibility for unprocessed vs. DNR conditions was not significantly different from 0 in the control room condition [$t(25)=-1.620$, $p=.118$]. This difference was significant for all other room conditions (all $p<.01$).

2.4.3 Subjective ratings results

Listening comprehension scores from the DCT can be seen in Figure 7. As expected given the Yes-No answer format of the questions and that only one story was used per participant per condition, the results were highly variable. The comprehension questions were included in the test protocol to keep listeners engaged with the listening task.

Subjective listener ratings of listening effort, speech naturalness, and background noise comfort during story listening in noise across different room conditions both with and without DNR processing can be seen in Figure 8. All of the *y*-axes are oriented such that larger numbers reflect better ratings (i.e., less listening effort, higher speech naturalness, greater background noise comfort). There was a significant effect of room condition on listening effort ratings, as indicated by a significant Friedman test (*p*<.001). *Post hoc* Wilcoxon signed-rank tests with a Bonferroni correction showed a significant difference between the control condition and room condition 2 (*p*=.010). To analyze whether there were any benefits of DNR on subjective listening effort, Wilcoxon signed-rank tests with a Bonferroni correction were conducted between DNR and unprocessed conditions at each level of room condition. There was a trend toward improved listening effort with DNR in the control condition, but this relationship failed to reach significance (*p*=.161). Listening effort was significantly poorer with DNR processing in room condition 2 (*p*=.035).

A Friedman test showed no effect of room condition on speech naturalness (*p*=.316). To analyze whether there were any effects of DNR on speech naturalness, Wilcoxon signed-rank tests with a Bonferroni correction were conducted between DNR and unprocessed conditions at each level of room condition. Speech naturalness was significantly poorer with DNR processing in room condition 1 (*p*=.005) and room condition 2 (*p*=.039). There was no effect of DNR processing on speech naturalness in the control condition or room condition 3 (both *p*>.05).

For background noise comfort ratings, a Friedman test showed no effect of room condition (*p*=.078). To analyze whether there were any benefits of DNR on background noise comfort, Wilcoxon signed-rank tests with a Bonferroni correction were conducted between DNR and unprocessed conditions at each level of room condition. Background noise comfort was significantly higher with DNR processing in the control condition (*p*=.019). There was no benefit of DNR processing on background noise comfort in any of the experimental room conditions (all *p*>.05).

2.5 Discussion

The purpose of this the current study was to investigate whether there was an interaction between DNR processing and reverberation using both acoustic and behavioral outcome measures. There was evidence for an interaction in the acoustic analyses. Consistent with previous work (Gustafson et al. 2014), DNR processing improved the SNR of speech-in-noise signals (Figure 4). However, improvement in SNR decreased in the reverberant conditions and was smallest in room condition 2 which had the poorest ratio of direct to reverberant energy (i.e., clarity index). This suggests that the presence of reverberant energy interferes with the ability of the DNR processor to form valid estimates of the signals and estimate the momentary SNR. If the DNR processor does not have accurate estimates of the speech and noise then it will be more likely to misclassify parts of the speech as noise and distort those parts of the speech signal. While the improvement in SNR is not equivalent to adjusting the levels at the source, it is indicative of to what extent the noise is being selectively suppressed without also removing the speech. It is possible that the presence of reverberation increased the amount of speech distortion caused by DNR as a result of misclassifying speech as noise. This would result in an overall reduction in the overall output speech level. This increased acoustic artifact hypothesis would potentially explain why there was a smaller improvement in SNR in the reverberant conditions compared to the anechoic control condition.

The interaction between DNR processing and reverberation was also evident in the speech intelligibility data. Consistent with previous studies (e.g., Sarampalis et al. 2009), there was a slight but non-significant decline in speech intelligibility with DNR processing in the anechoic control condition, suggesting that DNR introduced some speech distortion but not enough to significantly hinder speech intelligibility. However, there was a significant decline in speech intelligibility with DNR processing (Figure 6) across all three experimental reverberant conditions, which all had smaller improvements in SNR with DNR processing. Furthermore, the average decline in intelligibility was poorest in room condition 2 and there was a trend such that the declines in speech intelligibility with DNR processing increased with poorer ratios of direct to reverberant energy. Overall, the speech intelligibility results are consistent with the hypothesis that a poorer ratio of direct to reverberant energy decreases the efficacy of DNR processing leading to greater artifact and subsequent speech distortion.

While it is generally accepted that DNR may lead to a slight decrease in speech intelligibility, it is believed that the perceptual benefits of DNR on listening effort and background noise comfort will outweigh that effect. Consistent with our prediction, listeners reported a significant increase in background noise comfort with DNR processing in the anechoic control condition. We also expected a benefit of DNR processing for listening effort; however, while we observed a trend in that direction in the anechoic control condition (Figure 8A), that trend failed to reach statistical significance. There could be several reasons for this. Previous studies have often only found significant effects of DNR on listening effort in a subset of the most difficult listening conditions (e.g., Sarampalis et al. 2009; Desjardins & Doherty, 2014). It is possible that the 8 dB SNR broadband noise used in the story listening task was not challenging enough to yield a measurable change in listening effort with DNR. Another possibility is that the subjective listening effort scale used in the current study was not sensitive to changes in listening effort with DNR processing. Previous research has suggested that objective measures of listening effort (e.g., dual-task) may be more sensitive to effects on listening effort than subjective measures that rely on listener report (Desjardins & Doherty, 2014; Johnson et al. 2015). Thus, it is possible we would have observed a significant improvement in listening effort using an alternative, objective measure. With the growing interest in listening effort as a hearing aid outcome measure, there remains a lack of a proper method to measure listening effort (Edwards, 2007; Brons et al. 2013). Moreover, properly defining and quantifying listening effort may be particularly difficult for evaluating DNR processing in which the potential listening effort benefit may occur concurrently with a slight decrease in speech intelligibility.

In the reverberant environments, the decline is in speech intelligibility as a result of DNR was greater than in the anechoic control condition. Because the primary purpose of hearing aid processing is to facilitate speech understanding, this decrease in speech intelligibility with DNR processing may outweigh any other potential benefits of DNR processing. In the subjective ratings listeners reported DNR processed speech as significantly less natural in room conditions 1 and 2 (Figure 8B). This finding suggests that listeners may have been aware of the poorer overall speech intelligibility with DNR processing in reverberant environments. Listeners also indicated greater listening effort in room condition 2, presumably because DNR processing led to significantly decreased intelligibility of the story. This increase in listening effort with DNR processing is consistent with models of degraded speech perception. With the increased misclassification and degradation of speech by DNR in reverberant environments, the fragmented bottom-up representations of speech received by listeners do not match the phonological representation in their mental lexicon; thus, it is necessary for listeners to recruit top-down cognitive resources to help compensate and resolve this phonological mismatch

(Rönnberg et al., 2008, 2013). This increased allocation of mental resources would lead to an increase in listening effort. Overall, these findings are consistent with the hypothesis that DNR introduces a greater amount of acoustic artifacts and speech distortion when processing reverberant speech-in-noise signals. Moreover, behavioral results suggest that this increased distortion may outweigh the potential benefits of DNR processing in reverberant environments.

Brons et al. (2013, 2014) suggested that speech naturalness and background noise comfort are determining factors for whether listeners prefer DNR processing compared to unprocessed. That is, listeners prefer DNR most when background noise comfort is maximized and speech naturalness is not degraded. Based on this, we may be able to infer whether listeners would prefer DNR processing in anechoic and reverberant environments. In the anechoic condition, there was no significant decrease in speech intelligibility or perceived speech naturalness with DNR processing; however, there was a significant increase in background noise comfort. Therefore, it is likely that listeners would prefer listening with DNR processing because DNR processing provided a significant benefit (background noise comfort) without any significant decrement (speech naturalness). However, in room condition 2 which had the poorest ratio of direct to reverberant energy and poorest speech intelligibility, listeners indicated no benefit in background noise comfort and a decrease in speech naturalness with DNR. In this situation, we may be able to infer that listeners would actually prefer listening without DNR processing because DNR processing caused a significant decrement (speech naturalness) without providing any significant benefit (background noise comfort).

We predicted that the benefits of DNR processing would be decreased or eliminated in reverberant environments. Interestingly, these results extend beyond our prediction and suggest there may even be adverse perceptual effects of using DNR processing in highly reverberant environments. While DNR is typically applied in most hearing aid programs designed for speech listening, this would suggest the use of DNR processing in the real world should be contingent on room acoustics. For listening in highly reverberant environments, listeners may be best served with a program with the DNR processing algorithm inactive.

When considering how DNR may affect listeners in realistic noisy conditions, it is also important to consider the role of noise source characteristics. In the current study, we used a broadband noise which was spectrally stationary (i.e., the spectrum did not vary over time). A broadband, stationary signal is an ideal noise from a DNR perspective because it does not vary over time, and it is spectrally dissimilar from speech. For these reasons, it is easier for DNR to estimate and suppress a stationary broadband noise compared to non-stationary, speech-like noises (Alcántara et al. 2003; Arehart et al. 2003; Bentler et al. 2008). It is possible that the effects of reverberation on DNR efficacy would be even greater with non-stationary noise sources, especially babble, because those noise signals are already more difficult for DNR to accurately estimate. Thus, reverberation may have an even greater effect on the efficacy of DNR processing when the speech and noise signals are inherently less distinguishable.

2.5.1 Conclusions

We conclude that reverberation affects the efficacy of DNR processing in such a way that likely increases the introduction of speech distortion caused by DNR processing. While DNR processing had minimal effect on speech intelligibility without reverberation, speech intelligibility was substantially lower with DNR processing in reverberant conditions. Listeners also indicated greater listening effort and rated speech as less natural with DNR processing in a subset of the reverberant conditions. Lastly, while participants reported listening in noise as more comfortable with DNR processing in the anechoic condition, there was no increase in comfort with DNR in any of the reverberant conditions. Overall, these results suggest that DNR processing may be most beneficial in environments with little reverberation, and that the use of DNR processing in highly reverberant environments may actually produce adverse perceptual effects.

CHAPTER 3: REVERBERATION AND WIDE DYNAMIC RANGE COMPRESSION: ACOUSTIC EFFECTS

3.1 Abstract

Wide dynamic range compression (WDRC) processing in hearing aids alters the signal-to-noise ratio (SNR) of a speech-in-noise signal. This effect depends on the modulations of the speech and noise, input SNR, and WDRC speed. The purpose of the present experiment was to examine the change in output SNR caused by the interaction between modulation characteristics and WDRC speed. Two modulation manipulations were examined: (1) reverberation and (2) variation in background talker number. Change in SNR was quantified using a phase inversion technique. Results indicated that fast-acting WDRC altered SNR more than slow-acting WDRC; however, reverberation reduced this difference. Additionally, less modulated maskers led to poorer output SNRs than modulated maskers.

3.2 Introduction

Modern hearing aids use nonlinear, wide dynamic range compression (WDRC) processing to provide signal audibility for individuals with hearing impairment. Using WDRC, the amount of gain added varies on the basis of the input level, such that low-intensity inputs receive greater gain than high-intensity inputs. The primary purpose of such processing is to provide signal audibility while accounting for the reduced dynamic range of individuals with sensorineural hearing impairment. However, evidence suggests that WDRC amplification has the incidental effect of altering the signal-to-noise ratio (SNR) of a speech-in-noise signal (Souza et al., 2006; Alexander & Masterson, 2015). This change in SNR by WDRC may affect listener perception (see Naylor & Johannesson, 2009 for discussion of perceptual relevance).

Previous researchers have suggested a signal modulation hypothesis to account for the manner by which WDRC processing alters the SNR. This hypothesis states that when there is an input of a speech-in-noise signal, the higher amplitude signal will primarily control the WDRC gain function. That is, at positive SNRs the speech drives the compressor, and at negative SNRs the noise drives the compressor. In response to low-intensity portions of the signal controlling the compressor, the gain will increase in reaction to the decreased overall input level. This increase in gain consequently 'boosts' the lower level signal which may be present during these low-intensity modulations (Figure 9). Over the course of the signal duration, this has the effect of altering the output SNR from the hearing aid receiver relative to the initial input SNR picked up at the hearing aid microphone by disproportionately increasing the lower level signal. At negative SNRs, this means an overall improvement in SNR; whereas, at positive SNRs this means the output will be at a poorer SNR because of WDRC processing.

The rate at which the WDRC system adjusts the gain further affects the extent to which the SNR of a speech-in-noise input is altered. WDRC using short release times--generally classified as fast-acting WDRC--is quicker to respond to decreases in the input level and increase the gain than slow-acting WDRC which uses longer release times. As a result, fast-acting WDRC increases the gain more than slow-acting WDRC during these low-intensity portions of the signal which causes greater alteration to the SNR (Naylor & Johannesson, 2009; Alexander & Masterson, 2015).

Given that the degree to which WDRC alters the SNR of a signal depends on both the signal modulation characteristics and the speed of the WDRC system, there is likely an interaction between these two factors. In a partial examination of this interaction, Alexander &

Masterson (2015) examined the output SNR for fast-acting and slow-acting WDRC in either two-talker modulated or unmodulated noise. They observed an interaction between WDRC speed and noise modulation, such that output SNR was altered more by fast-acting WDRC than slow-acting WDRC. The difference between the two processing schemes was greater in the unmodulated condition. These results suggest there is an interaction between signal modulation characteristics and WDRC speed. However, this interaction has only been examined in a modulated versus unmodulated dichotomy. In the real world, signal modulation characteristics vary widely based on intrinsic (e.g., background talker number) and extrinsic (e.g., reverberation) factors.

The purpose of the current study was to examine the interaction between signal modulation characteristics and WDRC speed. To do this, acoustic analyses were performed to calculate the change in SNR to speech-in-noise signals processed by different WDRC strategies. The effect of signal modulation characteristics was examined along a continuum using two different manipulations which frequently occur in the real world: (1) reverberation which affects the modulation of the speech and noise and (2) variation in the number of background talkers which maintains the modulation properties of the speech.

3.3 Methods

3.3.1 Speech stimuli

Test stimuli were a set of 180 sentences taken from the IEEE corpus (IEEE, 1969). Each sentence was spoken by a male talker. Sentences were then processed in three stages: (1) combined with noise stimuli; (2) reverberation simulation; (3) WDRC simulation.

3.3.2 Noise stimuli

Speech stimuli were combined with noise from the International Collegium for Rehabilitative Audiology (ICRA; Dreschler et al., 2001). The ICRA noise is an artificial signal with speech-like spectral properties and varying temporal modulations based on number of speakers. Speech stimuli were combined with ICRA noise at four levels of modulation: 1-talker modulated, 2-talker modulated, 6-talker modulated, and unmodulated. The noise preceded the onset of the speech stimuli by 2.0 seconds in order to engage the WDRC processor prior to sentence presentation. The initial signal SNRs were set to range from -10 to +10 dB SNR in 5-decibel increments. The speech level was fixed at 65 dB SPL and the noise level varied to yield the final SNR levels.

3.3.3 Reverberation simulation

Virtual acoustic techniques were used to simulate four reverberant environments with average broadband (125 to 4000 Hz) reverberation times of 0.00 (anechoic), 0.75, 1.50, and 3.00 seconds. This simulation method allowed for the control of simulated room size and absorption coefficients to yield the final reverberation time. Because larger rooms tend to yield higher reverberation times, the simulated room size incrementally increased for each level of reverberation. See Table 2 for a summary of the simulated room sizes for each reverberation time condition. Across all conditions the source-listener distance was fixed at 1.4 m.

Briefly, the simulation method used an image model to compute directions, delays, and attenuations of the early reflections, which were spatially rendered along with the direct-path using non-individualized head-related transfer functions (HRTFs). The HRTFs were referenced relative to the center of the head in anechoic space, with the head absent. The late reverberant energy was simulated using exponentially decaying Gaussian noise. Overall, this simplified method has been found to produce binaural room impulse responses that are reasonable physical and perceptual approximations of those measured in real rooms (Zahorik, 2009).

3.3.4 WDRC simulation

Reverberant speech-in-noise signals were processed using a six-channel Matlab based WDRC simulation to mimic hearing aid processing using either fast-acting or slow-acting WDRC processing. The fast-acting condition was processed using a release time of 12 milliseconds, and the slow-acting condition used a release time of 1500 milliseconds. Both simulation conditions used a compression threshold of 45 dB SPL, compression ratio of 2:1, and attack time of 10 ms. There was also a linear control condition simulated in which the compression threshold was set to 1:1. In this linear processing condition we did not expect any change in SNR because both speech and noise signals would receive the same amplification regardless of level. Thus, this condition was included to verify the analysis methods.

Briefly, the WDRC simulation consisted of two basic stages. The first stage included nonlinear processing and spectral equalization and the second stage included linear filtering. This simulation approach accurately reflected the processing in real hearing aids, in which the nonlinear processing occurred in the body of the instrument and the effects of the receiver, tubing, and ear-canal acoustics were introduced by linear filtering (Arehart et al., 2010).

3.3.5 Acoustic analyses

The changes to the speech-in-noise signals as a result of WDRC processing were quantified as the change in SNR. Change in SNR was quantified using the Inversion Method (Hagerman & Olofsson, 2004) which has previously been used to quantify the effects of WDRC processing (Souza et al., 2006; Naylor & Johannesson, 2009; Alexander & Masterson, 2015). This method used signal phase cancellation to isolate speech and noise signals following processing in order to compute and compare the relative root mean square values. In order to isolate speech and noise signals, three versions of the speech-in-noise signals were processed by the WDRC simulation: (1) a phase-normal speech and noise combination (S_NN_N); (2) a phase-normal speech and a phase-inverted noise signal (S_NN_I); (3) a phase-inverted speech and a phase-normal noise (S_IN_N) signal. These three versions of the same sentence were then identically processed by the WDRC simulation. Following processing, different signals were added together to cancel out either the speech or noise and isolate the other signal (e.g., processed speech = $S_NN_N + S_NN_I$). After isolating the processed speech and noise signals, the root mean square values were calculated only in the sampling window where speech was present (i.e., ignoring the effects of WDRC on noise preceding and following the speech). Change in SNR was calculated as the difference between the initial SNR of the speech-in-noise signal and the output SNR of the signal following WDRC processing for individual sentences.

The purpose of the current study was to examine the interaction between WDRC speed and signal modulation characteristics using two different modulation manipulations: reverberation and background talker number. Figure 10 shows the specific signal processing conditions used to investigate each of these manipulations.

3.4 Results and Discussion

3.4.1 Effects of reverberation

Data were analyzed using a three-way analysis of variance (ANOVA) with reverberation time, WDRC speed, and input SNR as independent variables and change in SNR as the dependent variable. All assumptions of the model were met. Data are depicted in Figure 11. Significant main effects were found for all three factors [reverberation time: $F(3,10737)=2.964$, $p=0.031$, partial $\eta^2=0.001$; WDRC speed: $F(2,10738)=10.149$, $p<0.001$, partial $\eta^2=0.002$; input SNR: $F(4,10736)=4759.260$, $p<0.001$, partial $\eta^2=0.639$]. As seen in Figure 11, change in SNR was greatest with the fast-acting WDRC, followed by slow-acting WDRC, and there was no change in SNR with the linear amplification (as expected). Change in SNR was also greatest at input SNRs further from 0 dB. These main effects are consistent with previous literature (Souza et al., 2006; Naylor & Johannesson, 2009; Alexander & Masterson, 2015). Moreover, change in SNR decreased at higher reverberation times.

Based on the research question the primary interaction of interest was reverberation time × WDRC speed, which was significant [$F(6,10734)=3.360$, $p=0.003$, partial $\eta^2=0.002$]. This interaction was such that as the reverberation time increased, the difference between fast-acting and slow-acting WDRC decreased. Recall that WDRC alters the output SNR by disproportionately amplifying the lower amplitude signal in the valleys of the higher amplitude signal. Because increasing reverberation decreases the amplitude modulation depth of the input signal (Reinhart et al., 2016), then the glimpsing opportunities necessary for the WDRC processing to alter the SNR are decreased or removed. If the WDRC does not have the glimpsing opportunities necessary to go in and alter the SNR, then the specific rate of gain adjustment (i.e., WDRC release time) does not have as much of an impact. That is, it does not matter whether the gain adjustment is fast-acting or slow-acting if there is minimal gain adjustment occurring in the first place. This effect likely varies based on input SNR as indicated by significant reverberation time × WDRC speed × input SNR interaction [$F(24,10716)=46.143$, $p<0.001$, partial $\eta^2=0.093$].

Overall, these results suggest the acoustic effects of fast-acting vs. slow-acting WDRC for speech in noise may be diminished or even eliminated in reverberant environments. Specifically, fast-acting WDRC is more affected by reverberation. Because most everyday listening environments contain some amount of reverberation, this could have important behavioral implications for different WDRC strategies for complex listening environments. For example, previous research has suggested that in noisy environments hearing aid users may benefit from having a WDRC speed tailored to their cognitive functioning (Lunner & Sundewall-Thorén, 2007; Souza & Sirow, 2014). However, these results suggest that in environments that are both noisy *and* reverberant it may not matter what release time hearing aid users are fit with because both fast-acting and slow-acting WDRC may have similar acoustic effects. While there are other effects of WDRC related to perception, similarity in change in SNR is likely indicative of similarity of the other acoustic measures of perceptual relevance.

3.4.2 Effects of background talker number

Data were analyzed using a 3-way ANOVA with background talker number, WDRC speed, and input SNR as independent variables and change in SNR as the dependent variable. All assumptions of the model were met. Data are depicted in Figure 12. Significant main effects were found for all three factors [background talker number: $F(3,10462)=4484.256$, $p<0.001$, partial $\eta^2=0.562$; WDRC speed: $F(2,10463)=7050.370$, $p<0.001$, partial $\eta^2=0.709$; input SNR:

$F(4,10461)=6360.349$, $p<0.001$, partial $\eta^2=0.709$]. Based on Figure 12, less modulated maskers (i.e., a greater number of talkers) led to more negative changes SNR particularly for fast-acting WDRC. At positive SNRs where the speech is driving the compressor this effect occurs because the noise is more likely to be present during the low-intensity portions of the speech. At negative SNRs where the noise is driving the compressor, this effect occurs because the noise does not provide the opportunities necessary for the compressor gain function to increase the speech. This main effect is consistent with previous literature (Naylor & Johannesson, 2009; Alexander & Masterson, 2015).

Based on the research question the primary interaction of interest was background talker number × WDRC speed which was significant [$F(6,10459)=1600.007$, $p<0.001$, partial $\eta^2=0.478$]. At the positive SNRs, the effect of the WDRC speed is greater with the less modulated maskers. That is, the higher the likelihood the noise is present during the low-intensity portions of the speech (i.e., with less modulated maskers) then the greater effect the WDRC speed has acoustically. Conversely, at negative SNRs the masker does not provide glimpsing opportunities; therefore, the rate of gain adjustment may not have as much of an effect. This effect likely varies based on input SNR as indicated by significant background talker number × WDRC speed × input SNR interaction [$F(24,10441)=35.012$, $p<0.001$, partial $\eta^2=0.074$].

While few real-world listening situations contain true unmodulated noise, these results suggest that in less modulated situations fast-acting WDRC will cause a poorer output SNR than slow-acting WDRC. It has been hypothesized that the decrease in SNR with fast-acting WDRC at positive SNRs may affect listener perception (Naylor & Johannesson, 2009). Due to the interaction between WDRC speed and background talker number on change in SNR, this effect may be magnified in situations with less modulated noise sources. Because most listening environments have positive SNRs this may have important behavioral implications for hearing aid wearers in the real world (Hodgson et al., 2007). That is, WDRC (particularly fast-acting WDRC) may decrease intelligibility by providing the listener with a poorer SNR, especially with less modulated noise sources.

3.4.3 Conclusions

Overall, results indicated that fast-acting WDRC altered SNR more than slow-acting WDRC; however, reverberation reduced or eliminated the differences between WDRC speeds. Additionally, less modulated maskers led to poorer output SNRs than modulated maskers, especially when processed with fast-acting WDRC at positive input SNRs. These findings suggest that the modulation characteristics of a speech-in-noise signal can either accentuate or eliminate the effect of hearing aid compression speed on altering the SNR of a signal.

CHAPTER 4: REVERBERATION AND WIDE DYNAMIC RANGE COMPRESSION: BEHAVIORAL EFFECTS

4.1 Abstract

Previous work has suggested that when listening in modulated noise individuals benefit from different wide dynamic range compression (WDRC) speeds depending on their working memory ability. Reverberation reduces the modulation depth of signals and may impact the relationship between WDRC speed and working memory. The purpose of this study was to examine this relationship across a range of reverberant conditions.

Twenty-eight listeners with mild-to-moderate sensorineural hearing impairment were recruited in the current study. Individual working memory was measured using a reading span test. Sentence intelligibility in noise was measured across several signal-to-noise ratios and a range of reverberation times.

There was a significant relationship between WDRC speed and working memory with minimal or no reverberation. Consistent with previous research, this relationship was such that high working memory individuals had higher speech intelligibility with fast-acting WDRC, and low working memory individuals performed better with slow-acting WDRC. However, at longer reverberation times there was no relationship between WDRC speed and working memory. Overall, results suggest that there may not be any benefit of tailoring the WDRC speed to individuals based on individual working memory in reverberant listening environments.

4.2 Introduction

Speech perception in noise remains one of the most challenging tasks for hearing aid users. As such, a great deal of research has gone into investigating optimal hearing aid processing strategies specifically intended for improving speech perception in noise. One strategy for improved speech perception in noise is to tailor hearing aid signal processing based on an individual's working memory. Working memory is a short-term processing mechanism used in the simultaneous processing and storage of incoming information during complex cognitive tasks, such as speech perception (Baddeley, 2003). It has been argued that working memory is engaged during speech perception, especially in situations in which the auditory signal is deficient due to any combination of internal (e.g., hearing loss) or external (e.g., background noise) sources of distortion (Rönnberg et al., 2008, 2013). Because working memory is a limited-capacity system, listeners with lower working memory may be at a disadvantage than those with higher working memory in certain challenging listening situations. Due to the modifications that hearing aid signal processing performs on an auditory signal, previous work has demonstrated a link between working memory and amplified speech perception (see Souza et al., 2015 for a review).

One instance where working memory has been found to affect amplified speech perception is in regard to speech that has been modified using wide dynamic range (WDRC) processing. WDRC amplification provides time-varying gain based on the momentary intensity level of the incoming signals within any number of channels. To prevent too-rapid gain fluctuations, the speed of the time-varying gain function is partially controlled by a release time parameter. The release time controls the rate of increase in gain as a result of momentary decreases in the input intensity level. Fast-acting WDRC (i.e., processed with short release times) provides rapid gain adjustments which amplifies more of the low-intensity portions of the signal to an audible level; however in doing so, fast-acting WDRC degrades the modulation characteristics of the signal (Jenstad & Souza, 2005; Davies-Venn et al., 2009; Reinhart et al., 2016). Conversely, slow-acting WDRC (i.e., processed with long release times) adjusts the gain more slowly, which contributes to lower overall audibility of the low-intensity speech segments but less modulation distortion. Previous studies have demonstrated that individuals with higher working memory perform better in background noise with fast-acting WDRC, whereas individuals with poorer working memory perform better in background noise with slow-acting WDRC (Gatehouse et al., 2003; Lunner & Sundewall-Thorén, 2007; Lunner et al., 2009; Souza & Sirow, 2014; Ohlenforst et al., 2016). It is hypothesized that the reason for this is because individuals with greater working memory are able to benefit from the improved audibility of speech processed with fast-acting WDRC, while being able to compensate for the modulation distortion by using top-down processing (Rönnberg et al., 2008). In contrast, individuals with poorer working memory do not have sufficient cognitive resources to compensate for the modulation distortion caused by fast-acting WDRC and subsequently perform better with slow-acting WDRC.

However, the relationship between working memory and WDRC speed for speech perception in noise is limited to cases in which the background noise is modulated. Lunner and Sundewall-Thorén (2007) examined the relationship between working memory and speech intelligibility with fast-acting vs. slow-acting WDRC in both modulated and unmodulated noise conditions. They reported a significant relationship between working memory and WDRC speed for speech intelligibility only in the modulated noise condition. While the exact reason for this is not fully known, it is likely related to how modulated noises provide glimpses of speech information during low amplitude portions of the noise. During these glimpses, listeners receive a segment of the speech signal. Fast-acting WDRC will provide greater audibility of this underlying speech segment than would be provided by slow-acting WDRC. A listener with greater working memory resources will be better able to reconstruct the speech signal from these glimpsed, disjointed components of the signal than a listener with fewer working memory resources. While the results of Lunner and Sundewall-Thoren (2007) suggest that signal modulation characteristics impact the relationship between working memory and WDRC speed, the extent to which variation modulation characteristics in the real world will impact this relationship is not known.

In partial examination of this issue, Ohlenforst et al. (2016) examined the role of varying modulation in number of background noise talkers using 1-talker, 2-talker, and 6-talker modulated noises. They found that the relationship between working memory and speech intelligibility was present even in the least modulated noise condition (i.e., 6-talker). However, they also found that while the effect was present, it was also significantly smaller with the 6-talker noise (less modulated) condition than the 1-talker noise (more modulated) condition. This suggests that the degree of signal modulation may impact the benefit of fitting WDRC speed based on individual working memory. Nevertheless, the authors concluded that truly steady-state, unmodulated maskers are rarely present in the real world and thus this effect is still likely generalizable to the real world.

One critical factor previously unaccounted for is that most everyday listening situations contain reverberation. Reverberation occurs when acoustic energy reflections off of features in an environment persist even after the original sound source has ceased. As a consequence of this persistence of acoustic energy, reverberation reduces the modulation depth of signals (Houtgast

& Steeneken, 1985; Reinhart et al., 2016). In other words, reverberation causes modulated signals (including background noise) to become less modulated. This will likely decrease or even eliminate the glimpsing opportunities necessary for the relationship between working memory and speech intelligibility with fast-acting vs. slow-acting WDRC. For this reason we hypothesize that the presence of reverberation modifies the benefit of the cognition-based WDRC recommendation for speech intelligibility in noise.

In support of this hypothesis, Reinhart et al. (2017) performed acoustic analyses examining the relationship between reverberation and WDRC speed on speech-in-noise signals. They quantified the change in signal-to-noise ratio (SNR) as a result of fast-acting WDRC vs. slow-acting WDRC across a range of reverberant conditions. Consistent with previous work, they found a greater change in SNR with fast-acting WDRC than slow-acting WDRC in anechoic conditions (Naylor & Johannesson, 2009; Alexander & Masterson, 2015). However, as the amount of reverberation in the speech-in-noise increased, the acoustic difference between fast-acting WDRC and slow-acting WDRC decreased. This suggests that reverberation reduces the modulations of the signals which causes signals processed by varying WDRC speeds to become more acoustically similar than they otherwise would be in anechoic conditions. Thus, differences in performance with varying WDRC speeds may be reduced or even eliminated in more reverberant conditions. However, change in SNR is only one acoustic effect of WDRC processing of speech-in-noise signals. It is not known from pure acoustic analyses what the net effect of reverberation will be on listener perception.

The purpose of the present experiment was to examine the relationship between working memory and WDRC compression speed on speech across a range of reverberant conditions. We predicted that under anechoic conditions (i.e., without reverberation) higher working memory individuals will benefit more from fast-acting WDRC (compared to slow-acting WDRC), whereas lower working memory individuals will benefit more from slow-acting WDRC (compared to fast-acting WDRC). However, this relationship will potentially decrease or even be eliminated with increasing reverberation due to the signals becoming less modulated.

4.3 Methods

4.3.1 Participants

Twenty-eight older adults with sensorineural hearing impairment participated in the study (mean age = 73.3 years, range 60 to 85 years; 17 males, 11 females). Air conduction thresholds were measured at 250-8000 Hz octave frequencies and inter-octaves at 3000 and 6000 Hz. Bone conduction testing was performed at octave frequencies 250-4000 Hz. Participants presented with no more than a single air-bone gap ≥15 dB. Participants had symmetrical hearing loss defined as no more than a 10 dB difference in pure-tone average (thresholds 500, 1000, 2000 Hz) between ears. The Northwestern University Institutional Review Board approved all study procedures. Participants completed an informed consent process prior to participation, and they were compensated for their time.

4.3.2 Reading span test

Working memory was assessed using an English-language version of the Reading Span Test (RST) originally developed by Rönnberg et al. (1989). The RST taxed working memory resources by requiring individuals to simultaneously process and store sequential, incoming information. The test materials consisted of 54 sentences. Half of the sentences made semantic sense (e.g., “the captain saw his boat”) and half of the sentences did not make semantic sense (e.g., “the spider biked home”). Sentences were displayed on a 26-inch computer monitor in three clusters at a rate of 0.8 seconds. Sentences were presented in sets ranging from three to six sentences per set. Participants were required to complete three tasks during the RST: (1) to read the words aloud as they flashed across the screen, (2) at the end of each sentence to make a semantic judgment of whether the particular sentence made semantic sense, (3) at the end of each sentence set, to recall the first or last word of each sentence within that set of sentences. The participant was not told prior to seeing the set of sentences to whether the first or last word would be prompted. Whether the experimenter asked for the first word or last word of each sentence within a set was pseudo-randomized for each participant, such that first-word and last-word recall conditions occurred an equal number of times over the course of the test. The final RST score was the percentage of first or last words correctly recalled by the test participants out of the 54 sentences.

For a portion of the analyses listeners were split into high and low working memory groups on the basis of their performance on the RST. Based on previous studies in a similar sample population, a cut-off criterion of 41% was used in the present study (Arehart et al., 2013; Ohlenforst et al., 2016). Thirteen individuals were classified as high working memory (mean RST = 48.2%, SD = 4.9), and fifteen individuals were classified as low working memory (mean RST = 30.4%, SD = 3.8). Mean participant audiograms for both the left and right ears for both groups can be seen in Figure 13. There were no significant differences observed between high and low working memory groups in age [$t(26) = .93$, $p = .36$] or degree of hearing loss, as quantified by 4-frequency pure-tone average (mean of thresholds for both ears at 250-4000 Hz octaves) [$t(26) = .04$, $p = .97$].

4.3.3 Stimuli and processing

Sentence stimuli were taken from the IEEE corpus (IEEE, 1969). Low-context IEEE sentences (e.g., "The ripe taste of cheese improves with age") were used because the lengths of the sentence stimuli were likely to tax working memory, as well as to ensure that amplitude modulations in the signals would capture the fluctuations of the WDRC processor simulation. Sentences were locally-recorded from a single male talker from the Greater Chicago area in order to control for differences in regional dialect (McCloy et al., 2015). Stimuli were processed in three stages to yield the final set: (1) combined with noise, (2) reverberation simulation, (3) WDRC simulation.

Noise: Sentence stimuli were first combined with 1-talker modulated noise from the International Collegium for Rehabilitative Audiology (ICRA; Dreschler et al., 2001). ICRA noise was used due to its previous use in the literature exploring the relationship between working memory and WDRC speed (Gatehouse et al., 2003; Lunner & Sundewall-Thorén, 2007; Ohlenforst et al., 2016). The ICRA noise is an artificial signal with speech-like spectral and temporal properties but lacks informational content. The noise preceded the onset of the speech stimuli by 2.0 seconds in order to engage the WDRC processor prior to sentence presentation. On the basis of pilot testing to avoid floor and ceiling effects when combined with reverberation, the background noise was added at 2 and 5 dB SNR. Additionally, these SNRs represent the range of typical SNRs in everyday listening environments (Hodgson et al., 2007). The speech level was fixed at 65 dB SPL and the noise level varied to yield the final SNR levels. Multiple SNRs were used to increase generalizability of results.

Reverberation simulation: The second stage of signal processing was to process the speech-in-noise signals using a reverberation simulation. Reverberation was simulated using a Matlab-based simulation developed and validated by Zahorik (2009) which produced binaural room response simulations. Briefly, the simulation used an image-source model (Allen & Berkley, 1979) to simulate the direct sound and early specular reflections within a hypothetical room. The direction and delay of late, reverberant responses was estimated based on the source-listener location and dimensions of the modeled room. The attenuation of these late reverberant components were modeled using independent Gaussian noise samples with separate decay functions based on octave band (125-4000 Hz) absorption coefficients of the hypothetical room. Lastly, direct, early, and late reverberant components were spatially-rendered using a non-individualized head-related transfer function.

In the present simulation, source-listener distance was fixed, while the room size and absorptive properties of the reflective surfaces were varied to produce a range of reverberation conditions. Source-listener distance was fixed at 1.4 m which is a typical conversational distance. In the real world, larger rooms tend to produce longer reverberation times. To reflect this, room size was increased incrementally with increasing reverberation. In total, four reverberation conditions were simulated. See Table 2 for a summary of the simulated room sizes for each reverberation time.

WDRC simulation: The third and final stage of stimulus processing was to simulate the effects of varying WDRC speed processing on the reverberant speech-in-noise signals. This was achieved using a modified Matlab-based simulation developed by Kates (2008). Briefly, the program implemented a six-channel filter bank followed by a peak detector that reacted to within-band fluctuations in signal level. Increases in signal level were followed using an attack time which was set to 10 ms. Decreases in signal level were followed using a release time parameter. Two WDRC speed conditions were simulated using a release time of either 12 ms (fast-acting WDRC) or 1500 ms (slow-acting WDRC). In both conditions, the compression threshold was set at 45 dB SPL, and a compression ratio of 2:1 was used. Stimuli levels into the compressor were set such that the speech level was fixed at 65 dB SPL.

4.3.3 Speech task and procedure

Speech intelligibility was the outcome variable used in the study. Speech intelligibility testing was conducted with the participant seated in a double-walled sound booth. Signals were presented binaurally via Etymotic ER-2 insert phones (Elk Grove Village, IL). Signal playout level was calibrated such that the speech level was presented at 65 dB SPL, and NAL shaping (Byrne & Dillon, 1986) was subsequently applied according to the participant's hearing thresholds. Sentences were blocked by WDRC speed, and block order was randomized for each participant. One second prior to each sentence presentation there was a 1000 Hz pure tone of 250 ms duration in order to alert participants of the imminent sentence presentation. Following sentence presentation participants were asked to repeat the sentence back as best as they could. They were encouraged to guess even if they were not certain of what they heard or only got a part of the sentence. Participant response time was unlimited. Responses were recorded on the basis of five keywords per sentence (e.g., "The ripe taste of cheese improves with age") by a single experimenter to ensure consistent scoring. Listeners did not receive feedback during testing. Stimuli presentation and response recording were controlled by a custom-made Matlab program. There were a total of 10 sentences per condition for a total of 160 sentences (2 WDRC speeds * 4 reverberation times * 2 SNRs * 10 repetitions). Breaks were given in 20-minute intervals to minimize participant fatigue.

4.4 Results

Raw data can be seen in Figure 14. Mean intelligibility scores for each condition ranged from 37% to 81% which suggests no substantial floor or ceiling effects in the intelligibility data. Intelligibility was generally poorer in the 2 dB SNR condition than the 5 dB SNR condition. Intelligibility was also generally poorer in the reverberant conditions than the anechoic (0.00s reverberation time) condition. For statistical analyses, several transformations were performed on the data. First, intelligibility data were first transformed to rationalized arcsine units (RAU) in order to stabilize variance across the performance scale (Studebaker, 1985). Next, data were combined between the two SNR conditions because the role of SNR was not a central research question and preliminary analyses suggested no interaction with SNR. Lastly, similar to previous studies (Lunner & Sundewall-Thorén, 2007; Ohlenforst et al., 2016), data were reduced by subtracting the scores for the slow-acting WDRC speed from the scores for the fast-acting WDRC speed within-subjects.

4.4.1 ANOVA analyses

Data were initially analyzed using a group-split approach similar to those previously used to examine the relationship between working memory and performance with varying WDRC speeds (Lunner & Sundewall-Thorén, 2007; Souza & Sirow, 2014; Reinhart & Souza, 2016). The resulting RAU difference scores are shown in Figure 15. Positive values indicated that listener performance was better with fast-acting WDRC. Conversely, negative values indicated that listener performance was better with slow-acting WDRC.

The transformed speech intelligibility scores were analyzed using a 2-way repeated measures analysis of variance with one within-subjects variable (reverberation time) and one between-subjects variable (working memory group). All assumptions of the model were met. The main effect of reverberation time was not significant [$F(3,78)=.396$, $p=.756$, partial $\eta^2 =.015$]. There was a significant main effect of working memory group [$F(1,26)=6.202$, $p=.019$ partial $\eta^2 =.193$] which suggests that individuals with different working memory performed significantly different with either fast-acting or slow-acting WDRC. However, there was also a significant working memory group x reverberation time interaction [$F(3,78)=3.809$, $p=.013$, partial $\eta^2 =.128$] so these main effects should be interpreted with qualification.

To further explore the working memory group x reverberation time interaction, *post hoc* independent-samples t-tests were conducted between the difference RAU scores for the High and Low working memory groups at each reverberation time condition with a Bonferroni correction. These results indicated that there was a significant difference in performance with either fast-acting or slow-acting WDRC in the 0.00 second [$t(26)=3.532$, $p=.002$] and 0.75 second [$t(26)=2.524$, $p=.018$] reverberation time conditions. There was no difference between the groups in either the 1.50 second or 3.00 second reverberation time conditions (both $p>.050$).

4.4.2 Regression analyses

To examine further the potential clinical utility of considering individual working memory when selecting a WDRC speed, regression analyses were performed to quantify the amount of variance explained by working memory. It would be more clinically significant to consider listener working memory for conditions in which working memory accounts for a higher proportion of the variance than in conditions in which working memory does not account for as much variance in performance.

To explore these relationships, multiple hierarchical regression models were calculated. The dependent variable in each model was the transformed speech intelligibility data. A separate analysis was conducted at each of the reverberation time conditions. The primary predictor of interest was working memory, while also considering the effects of hearing loss (quantified as average of thresholds at 500, 1000, 2000, and 4000 Hz across both ears) and age. The predictors were entered into the models sequentially, with working memory being entered as the last variable. See Table 3 for the results of these analyses.

For the 0.00 second reverberation time condition, neither participant age nor PTA were significant predictors. In that condition, working memory was a significant factor (p=.011) and explained 35% of the variance in performance. Similarly, in the 0.75 second reverberation time condition working memory was the only significant factor (p=.004). However, in this condition with slightly less modulation the amount of variance explained by working memory decreased to 31.5%. In both the 1.50 and 3.00 second reverberation time conditions, none of the listener factors were significant predictors (all p>.050).

4.5 Discussion

The purpose of the present experiment was to examine the relationship between working memory and WDRC compression speed on speech-in-noise perception across a range of reverberant conditions. Speech intelligibility was measured in older listeners with hearing impairment for sentences in noise with varying amounts of reverberation processed by fast-acting and slow-acting WDRC. Consistent with previous research, there was a significant relationship between working memory and WDRC speed in the anechoic condition (0.00s reverberation time), such that high working memory individuals had higher speech intelligibility performance with fast-acting WDRC, whereas low working memory individuals had higher speech intelligibility with slow-acting WDRC. This is consistent with the hypothesis that fast-acting WDRC amplifies brief speech segments during modulations of the noise signal. While this distorts the speech signal, individuals with a higher working memory are able to allocate cognitive resources to construct the speech message from these disjointed speech segments using lexical and contextual information. In contrast, individuals with less working memory resources do not have sufficient cognitive resources to perform this compensatory decoding and instead perform better with slow-acting WDRC which causes less distortion.

The magnitude of this effect without any reverberation was slightly larger in the current study compared to previous studies that used a similar design with low context sentence stimuli (Gatehouse et al., 2003; Lunner et al., 2007; Ohlenforst et al., 2016). This is potentially due to differences in the release time values used between the fast-acting and slow-acting WDRC conditions. Previous studies have used release times of 40 and 640 ms for their fast-acting and slow-acting WDRC, respectively. In contrast, the current study used more extreme release time values of 12 and 1500 ms. It is likely that using more disparate release times would exacerbate the acoustic differences between fast-acting and slow-acting conditions. Thus, the perceptual implications for individuals with varying working memory resources would reflect this when comparing between more disparate release times.

4.5.1 Effects of reverberation

The presence of reverberation significantly affected the relationship between working memory and performance with fast-acting vs. slow-acting WDRC. In reverberant conditions, the benefit of tailoring WDRC speed based on working memory was reduced (in the 0.75 second reverberation time condition) or eliminated (in the 1.50 and 3.00 second reverberation time conditions). This effect of reverberation on the relationship between working memory and WDRC speed was consistent with our hypothesis. Reverberation reduces the modulations of the noise signal which prevents the WDRC gain function from glimpsing the underlying speech components through the noise.

The degree to which reverberation disrupts the relationship between working memory and WDRC speed is dependent on the amount of reverberation. At the mild reverberation time (0.75 seconds) there may have still been some modulations of the noise remaining which would provide glimpsing opportunities for the WDRC gain function to amplify speech segments. Thus, there was still some benefit of tailoring WDRC speed based on working memory (Figure 15). However, due to the overall reduction of signal modulation the benefit and amount of variance in performance explained by examining working memory was reduced compared to the anechoic condition which maintained a greater magnitude of signal modulation (Table 3). Greater amounts of reverberation were likely to completely fill in the noise modulations. This would prevent the WDRC gain function from increasing during the noise modulations to amplify segments of the speech. This meant that in the higher reverberation time conditions, there was no significant benefit of tailoring WDRC speed based on working memory.

Overall, these effects of reverberation on the relationship between working memory and WDRC speed are consistent with previous acoustic analyses. Reinhart et al. (2016b) concluded that fast-acting and slow-acting WDRC had similar effects on the SNR of a speech-in-noise signal as a function of increasing reverberation. It stands to reason that if slow-acting and fast-acting WDRC are acoustically similar in reverberant conditions, then listener perception between those processing conditions should also be similar.

While the current study suggests there may be a benefit of tailoring WDRC speed based on working memory at mild reverberation times, there may also be an interaction with noise characteristics. The current study used a 1-talker modulated noise which was highly modulated and provided maximal glimpsing opportunities. It is possible that with a less modulated noise source (e.g., 6-talker modulated noise) where the relationship between working memory and WDRC speed is already reduced (Ohlenforst, 2016), then even a mild amount of reverberation would reduce the few available modulations enough to eliminate any benefit of tailoring WDRC speed based on working memory.

4.5.2 Clinical implications and future directions

Overall these results suggest that when considering how to improve listener communication in noise in specific situations it may be important to consider the room acoustics of the noisy situations. Tailoring the WDRC speed based on individual working memory may provide real world benefit in environments without much reverberation (i.e., certain restaurants). However, this strategy may not be as beneficial for listeners in more reverberant environments (i.e., theaters or places of worship). In these moderately reverberant environments, a listener may experience better speech-in-noise benefit using a different rehabilitation strategy. For example, remote microphones would not only preserve the signal modulations by reducing the amplification of reverberant energy but would more importantly improve the SNR of the signal (Boothroyd, 2004).

There are several commercially available dereverberation algorithms designed to improve listener communication in reverberation (e.g., Fabry & Tehorz, 2005); however, the acoustic and behavioral benefits have not been explored in the scientific literature to our knowledge. It is not known whether the parallel use of a dereverberation algorithm would restore the modulations of the speech and noise signals in such a way that the relationship between working memory and WDRC speed would re-emerge. The potential benefits of dereverberation algorithms and how they might interact with other hearing aid signal processing algorithms (e.g., WDRC, digital noise reduction, directional microphones) requires further research.

4.5.3 Conclusions

Consistent with previous studies, these results suggest the potential for benefit of tailoring WDRC speed based on individual cognitive ability but only for certain environments. In anechoic conditions, listeners with high working memory performed better with fast-acting WDRC, whereas listeners with low working memory performed better with slow-acting WDRC. However, this effect was diminished in mildly reverberant conditions and eliminated at higher reverberation times. Overall, there should be caution when attempting to generalize expectation of the benefits of tailoring WDRC speed based on individual cognition to the real world where reverberation varies.

CHAPTER 5: DISCUSSION

5.1 Summary of Results

The results of the experiments described in this dissertation support the hypothesis that reverberation affects the efficacy of processing recommendations designed to improve speech perception in noise. This issue was explored using an approach including reverberation and hearing aid algorithm simulations implemented with realistic parameters. As such, the results represent a proof of concept of the potential interaction between reverberation and hearing aid processing for clinical applications.

Chapter 2 demonstrated that DNR processing is disrupted by reverberation. Acoustically, reverberation decreased the improvement in SNR achieved with DNR processing. Behaviorally, speech intelligibility was poorer with DNR processing only in the reverberant conditions. Moreover, subjective ratings suggested that listeners were aware of this because they rated speech as less natural and listening as more effortful with DNR processing in reverberant conditions. Listeners only indicated higher background noise comfort with DNR processing in the anechoic control condition.

Chapter 3 demonstrated that signal modulation characteristics (i.e., varying reverberation and background talker number) affect WDRC processing. The acoustic effects of WDRC processing (quantified as change in SNR from input to output) using fast-acting and slow-acting WDRC varied based on manipulation of the signal modulation characteristics. In general, fast-acting WDRC altered SNR more than slow-acting WDRC; however, reverberation reduced the difference between fast-acting and slow-acting WDRC. Additionally, less modulated maskers led to poorer output SNRs than modulated maskers.

Chapter 4 behaviorally examined the effects of varying WDRC speed under reverberant conditions. With minimal or no reverberation, individuals with high working memory had higher speech intelligibility with fast-acting WDRC, and individuals with low working memory had higher speech intelligibility with slow-acting WDRC. However, in more reverberant conditions the benefits of fitting WDRC speed based on individual working memory was eliminated. This is consistent with acoustic results of Chapter 3 that differences in perception between fast-acting and slow-acting WDRC would diminish as they become more acoustically similar under reverberant conditions. Overall, Chapters 3 and 4 demonstrate that the cognition-based WDRC speed recommendation for improved speech perception in noise is disrupted by reverberation.

5.2 Implications

Given the real-world prevalence of reverberation and its apparent interaction with hearing aid processing, there may be several implications of these findings.

Findings suggest that hearing aid processing recommendations designed to improve speech perception in noise may not provide as much benefit to listeners in realistic situations (i.e., those including reverberation) as was previously suggested by laboratory studies. This incongruity between expected benefit and reality could contribute to the perception of poor device performance in noise which remains one of the most frequently cited reasons for discontinued device use (Bertoli et al., 2009; Kochkin, 2000; McCormack & Fortnum, 2013). Most clinical fittings and validations are performed under idealized conditions which contain minimal reverberation. These environments may not represent the different listening environments in which hearing aid users will primarily be listening through their device(s). Given the apparent interaction between hearing aid processing and reverberation, how listeners perform with their devices in reverberation is a critical consideration. To address this, it might be beneficial to provide additional validation of device performance in more realistic conditions (i.e., with reverberation) than is currently applied for functional testing. This suggests that testing, both clinical and research, should incorporate some reverberation into hearing aid processing and speech perception testing in order to better approximate real-world listening and improve external validity.

Overall, a more comprehensive understanding of the benefits of various signal processing strategies in reverberant environments could refine the way these strategies are administered in the clinic. That is, if a patient seeks help for noisy situations with relatively little reverberation (e.g., small restaurant) then DNR and cognition-based WDRC strategies are more likely to provide perceptual benefit. However, if a patient seeks help for noisy situations which occur in more reverberant rooms (e.g., a lecture hall) then that patient may be better served by an alternative treatment approach. For example, the use of a remote microphone closer to the sound source would not only preserve the signal modulations of the speech by reducing the amplification of reverberant energy but would more importantly improve the SNR of the signal (Boothroyd, 2004).

In addition to informing expectations of current signal processing performance under reverberant conditions, results may also encourage development and modification of hearing aid processing algorithms to be more resistant to the effects of reverberation. In partial response to this, several hearing aid manufacturers have introduced dereverberation algorithms which are designed to mitigate the effects of reverberation on listener perception (e.g., Phonak "EchoBlock"; Siemens "EchoShield"). However, these dereverberation algorithms are typically only available in the highest technology levels which are more expensive than devices without these algorithms. While being marketed as a premium feature, the efficacy of this class of hearing aid algorithms have never been objectively evaluated in independent, peer-reviewed research.

Many hearing aid wearers and clinicians assume that more expensive technology and more hearing aid features will yield greater perceptual benefit. However, previous research has challenged this assumption (Cox et al. 2014, 2016; Johnson et al., 2016). In these studies, researchers examined whether advanced technology level hearing aids provided better objective or patient-reported outcomes than basic technology level hearing aids. They found that while both basic and advanced hearing aids provided substantial benefit in extended hearing aid trials, there was no evidence to suggest that the advanced hearing aids yielded better outcomes than the basic hearing aids. While it cannot be directly inferred that the dereverberation algorithms in advanced hearing aids do not improve outcomes, these results emphasize that advanced hearing aid features (including dereverberation algorithms) should be validated as providing significant benefit to justify their application in clinical settings. That is, whether dereverberation algorithms provide enough benefit in reverberant environments to merit the additional expense is not known. Furthermore, whether dereverberation algorithms interact with other algorithms (e.g., DNR and WDRC) is not known and requires further research. It is possible that the parallel use of a dereverberation algorithm will ‘correct’ the input signal which could remove the interaction between specific processing algorithms (e.g., DNR and WDRC) and reverberation presently observed.

5.3 Properties of Reverberation

In the present study we took a realistic approach in the simulation of reverberation in which both the size of the room and the absorption coefficients of the simulated room surfaces were manipulated to yield the final reverberation conditions. This led to several characteristics of the reverberation varying across the conditions. Reverberation is a multi-faceted acoustic phenomenon defined by several characteristics (e.g., reverberation time and direct-to-reverberant ratio). Varying either of these characteristics will potentially affect hearing aid processing. While reverberation time and direct-to-reverberant ratio are closely related, it is possible, as in the present experiment, that one room will have a worse direct-to-reverberant ratio whereas another will have a worse reverberation time (Table 2). As such, the exact mechanism by which reverberation affects hearing aid signal processing cannot be directly inferred from the present study. It is possible that both direct-to-reverberant ratio and reverberation time will disrupt hearing aid processing, albeit in different ways.

The direct-to-reverberant ratio refers to the density of the reverberant energy which is typically greater in less voluminous rooms (Gelfand & Silman, 1979). It is likely that increasing density of the detrimental reverberant energy (i.e., lower direct-to-reverberant ratios) disrupts hearing aid processing by decreasing the modulation depth of the signal causing the signal to become more steady-state. Both DNR and WDRC rely on following the modulations of the signal as part of their processing. WDRC applies a time-varying gain function which varies based on the changes in the instantaneous input level. A steady-state signal will receive a more constant gain across the duration of the signal compared to a signal with full modulation depth. This may decrease the overall audibility of a reverberant signal compared to an anechoic signal. Low-intensity phonemes are made audible when the gain is increased in response to a drop in the overall input level. In a signal where reverberant energy is increasing the momentary input level during low-intensity portions of the speech, the gain function may fail to increase and make those portions of the speech audible. Similarly, decreased signal modulation will also affect DNR by disrupting the ability of the algorithm to accurately calculate the instantaneous SNR within a channel. An accurate calculation of the SNR is essential for adjusting the channel gain to minimize noise amplification while not incidentally suppressing the speech.

Reverberation time refers to the duration that the reverberant energy is present which may also disrupt hearing aid processing. At short reverberation times (re: long reverberation times) the reverberant energy decays more rapidly, and at any point there may only be reverberant spectral components from the immediately preceding phoneme. In contrast, at longer reverberation times, there may be spectral components present from several preceding phonemes. Because hearing aid processing is frequency channel-specific this increased duration of channel activation may affect hearing aid processing. For example, consider a word with two higher-frequency phonemes with a string of lower-frequency phonemes in between (e.g. /slops/). At a longer reverberation time the reverberant reflections from the initial /s/ phoneme will remain present in the high frequency channels between the /s/ phonemes where there otherwise should not be any relevant high frequency information. At a lower reverberation time the high frequency reverberant energy from the initial /s/ would have dissipated, thus the high frequency channels would be relatively unperturbed by the time the final /s/ is presented. This increased duration of channel activation may disrupt DNR processing because the speech and noise estimates may not accurately update within a given channel. That is, in the previous example, the speech estimate may mistakenly classify that there is relevant speech information to amplify during the intermediate low-frequency portion of the signal when there really is not. For WDRC this increased duration of channel activation will disrupt the gain control from fluctuating properly. While the exact mechanism requires further research, it is evident that reverberation alters the input signal acoustics in such a way that will affect DNR and WDRC hearing aid processing.

5.4 Simulation

In the present study we used DNR and WDRC simulations implemented with realistic parameters that were representative of similar algorithms implemented in commercial hearing aids (Kates, 2008). A simulation approach has the advantage of providing greater experimental control than wearable, commercial hearing aids. Many properties of the hearing aid algorithms in commercial hearing aids lack transparency as they are proprietary in nature. Related to this lack of transparency, it can also be difficult to isolate processing features (e.g., DNR and WDRC) in commercial hearing aids because manipulation of some features is intertwined with other features. Moreover, these interrelationships of processing features are not always apparent from the user interface. The current approach optimizes the tradeoff between experimenter control and external validity. Nevertheless, to increase generalizability of results to more clinical situations, future work should consider effects of WDRC and DNR processing in wearable hearing aids in real reverberant environments.

As a note, currently there are advantages and disadvantages for both signal processing simulation and wearable hearing aid approaches to conducting hearing aid research. As part of an effort to integrate the benefits of transparent, well-controlled simulations and portable devices capable of real-time processing, the National Institutes of Health have supported the development of an open speech signal processing platform. Such a platform could change the way hearing aid research is done in the future.

5.5 Conclusions

In summary, the work presented in this manuscript provides new insights in the effect of reverberation on hearing aid signal processing of speech-in-noise signals. Due to the prevalence of reverberation in real world listening situations, understanding how hearing aid signal processing performs under reverberant conditions informs the benefits we can expect hearing aid wearers to receive across a wide range of reverberant conditions. Chapter 2 demonstrated that DNR processing is disrupted by reverberation in such a way that may yield adverse perceptual effects when used in reverberant environments. Chapters 3 and 4 showed that the benefits of fitting WDRC speed based on individual cognition may be diminished or eliminated in reverberant environments where WDRC processing with different parameters are more acoustically similar than in anechoic environments. Overall, these findings suggest that the benefit experienced by listeners in the real world may vary based on the reverberant characteristics of a given listening situation. Results will serve as the foundation for future work aimed at improving hearing aid signal processing for individuals with hearing impairment across a range of reverberant environments.

TABLES

Table 1. Chapter 1. Examples of real-world reverberant conditions

Example reverberation times representative of many noisy real-world listening situations.

Listening Environment	Reverberation Times
Restaurants	0.45-1.41 seconds (Hodgson et al., 1999)
Classrooms/Lecture Halls	0.50-1.80 seconds (Hodgson et al., 2007)
Concert Halls	1.00-2.30 seconds (Winckel, 1962)

Table 2. Chapters 2-5. Reverberation simulation parameters

Simulation condition details for each of the reverberation time conditions. Reverberation time and direct-to-reverberant energy ratio values were computed from the binaural room impulse responses used for the simulation. Critical distance was estimated from the room volume and broadband reverberation time, assuming an omni-directional source using methods described in Kuttruff (2000).

Room Size **(length x width x height)**	**Broadband Reverberation Time** (125 – 4000 Hz)	**Mean Octave-band Direct-to-Reverberant Ratio** (125 – 4000 Hz)	**Estimated Critical Distance**
Free field	0.00s	∞ dB	N/A
5.7m x 4.3m x 2.6m	0.75s	-5.10 dB	.52m
8.6m x 6.5m x 3.9m	1.50s	-3.56 dB	.67m
12.9m x 9.8m x 5.9m	3.00s	-2.75 dB	.89m

Table 3. Chapter 4. Regression analyses results

Results of regression analyses examining listener factors (pure-tone average, age, and working memory) associated with speech intelligibility difference score.

Reverberation Time (s)	**Variable**	**ΔR^2**	**F**	**p**
0.00	PTA	.005	.130	.721
	Age	.010	.188	.830
	Working memory	.350	4.601	.011
0.75	PTA	.079	2.219	.148
	Age	.022	1.397	.266
	Working memory	.315	5.696	.004
1.50	PTA	.012	.316	.579
	Age	.058	.935	.406
	Working memory	.041	.992	.413
3.00	PTA	.064	1.772	.195
	Age	.063	1.818	.183
	Working memory	.021	1.388	.271

FIGURES

Figure 1. Chapter 1. Importance of speech understanding in noise

Results of a survey regarding importance of communication in complex environments for listeners with hearing loss.

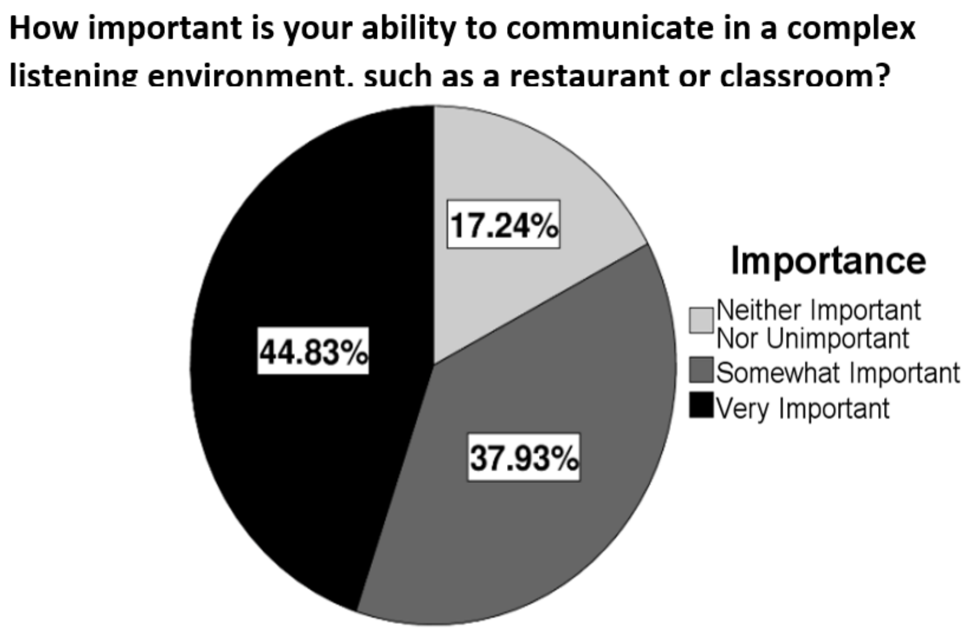


Figure 2. Chapter 1. Acoustic effects of reverberation

Effects of reverberation in both time (left panel) and spectral (right panel) domains. All panels depict the nonsense syllable /ata/ being affected by different amounts of reverberation.

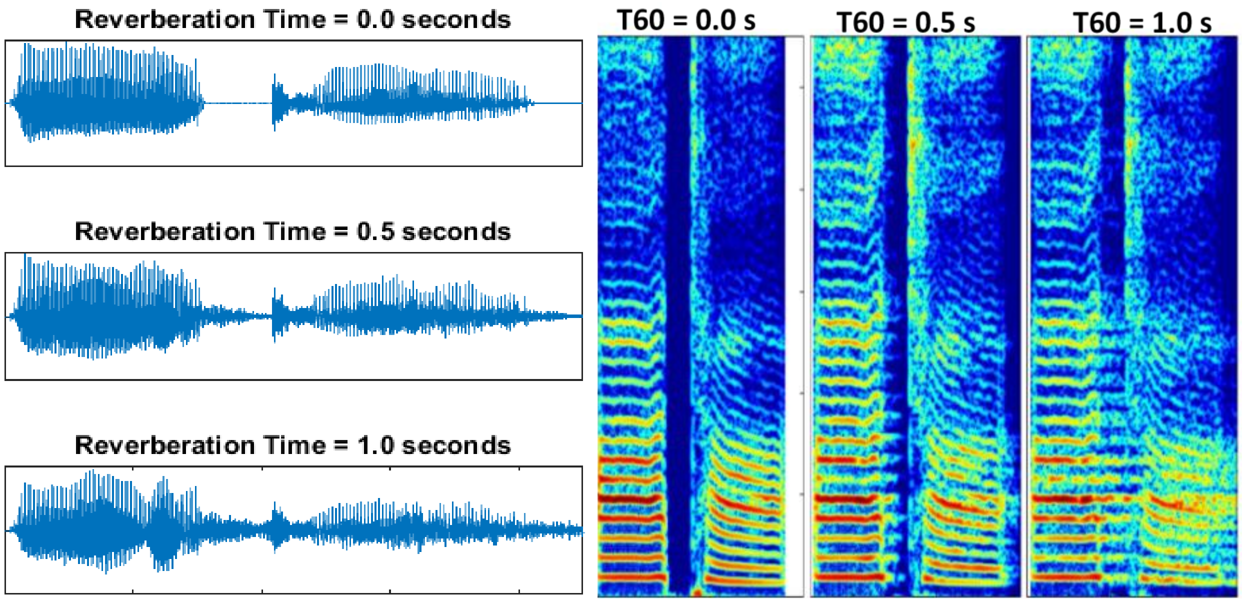


Figure 3. Chapter 2. Participant audiograms I

Mean air-conduction thresholds of participants (n=26). Error bars represent +/- 1 standard deviation.

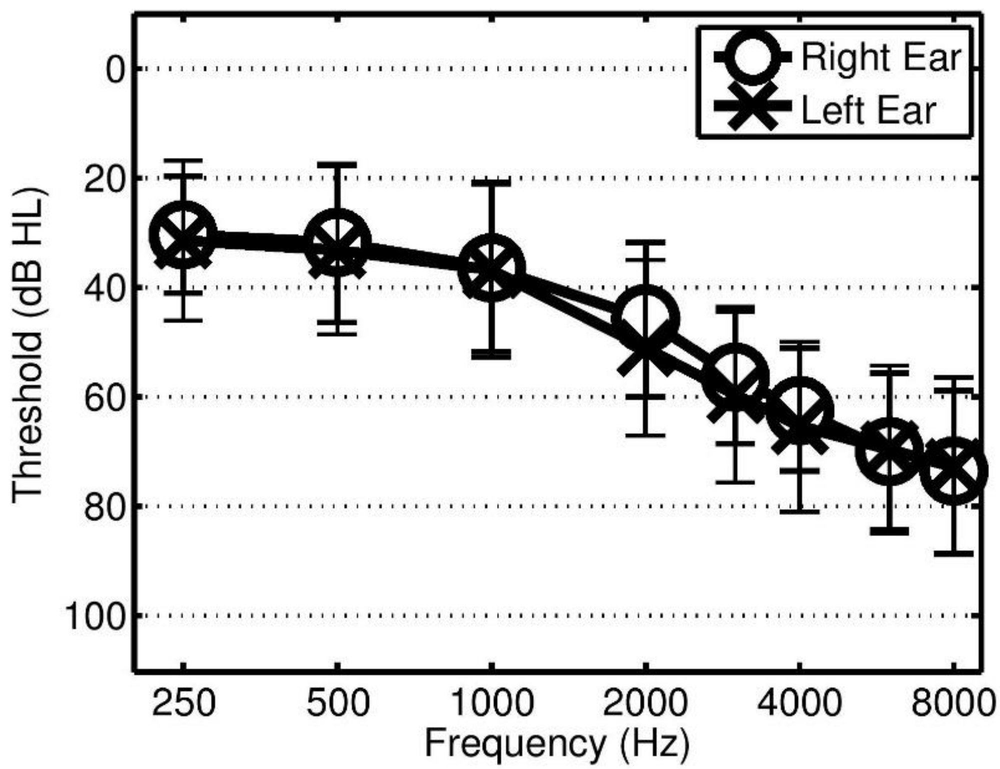


Figure 4. Chapter 2. Acoustic results with DNR

Summaries of change in SNR quantified using the Inversion Method as a result of DNR processing across the different reverberation room conditions. The left and right panels show acoustic effects at different Input SNRs (in dB). Data are displayed as boxplots, with outliers indicated by point symbols.

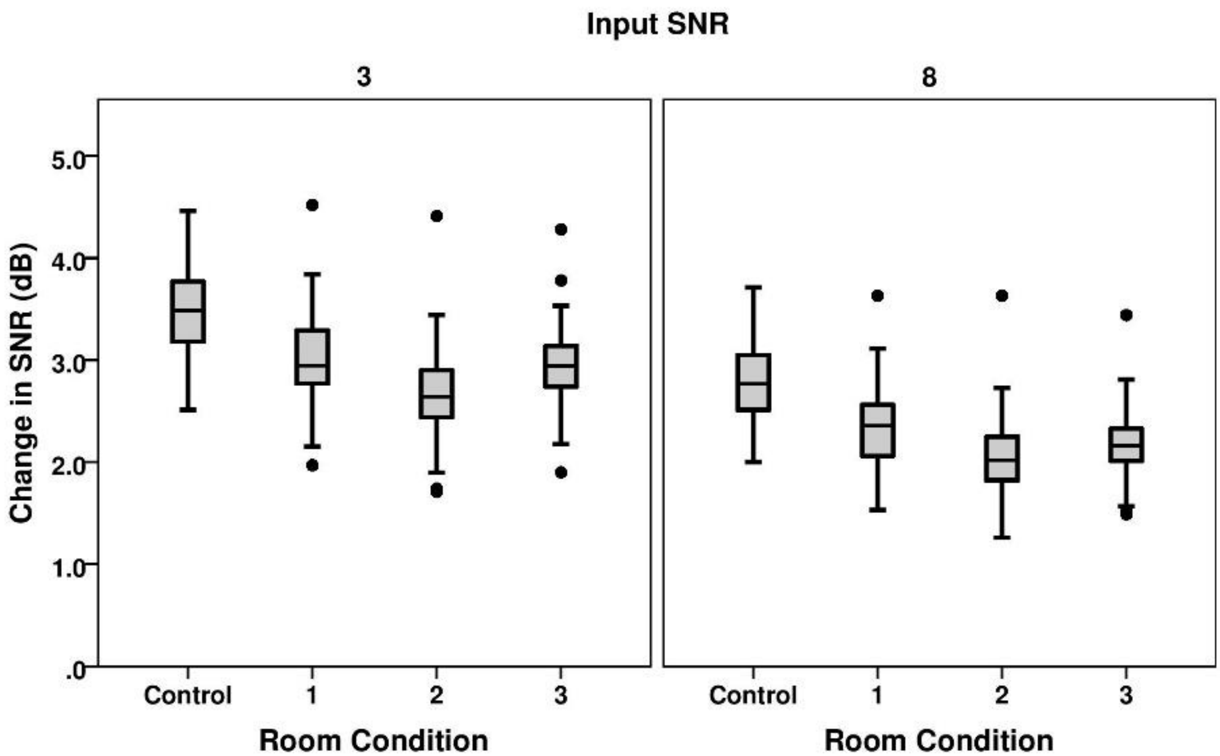


Figure 5. Chapter 2. Speech intelligibility results with DNR

Speech intelligibility scores in RAUs with and without DNR processing across the different room conditions. The left and right panels show performance at different Input SNRs (in dB). The error bars represent +/- 1 standard error of the mean.

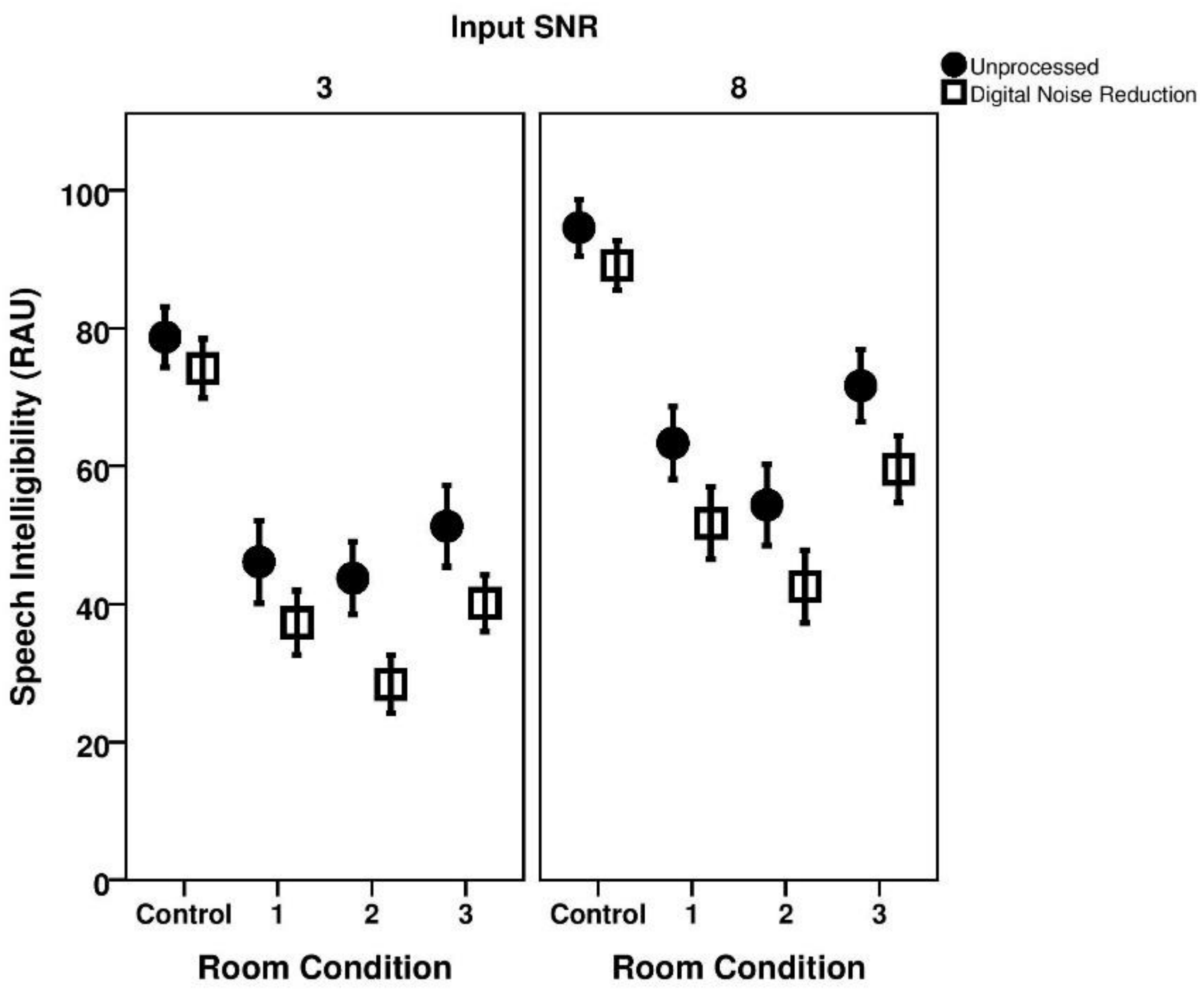


Figure 6. Chapter 2. Difference in intelligibility with DNR

Decrease in speech intelligibility (RAU-transformed unprocessed speech intelligibility – RAU-transformed DNR speech intelligibility) collapsed across input SNR for each of the room conditions. The error bars represent +/- 1 standard error of the mean.

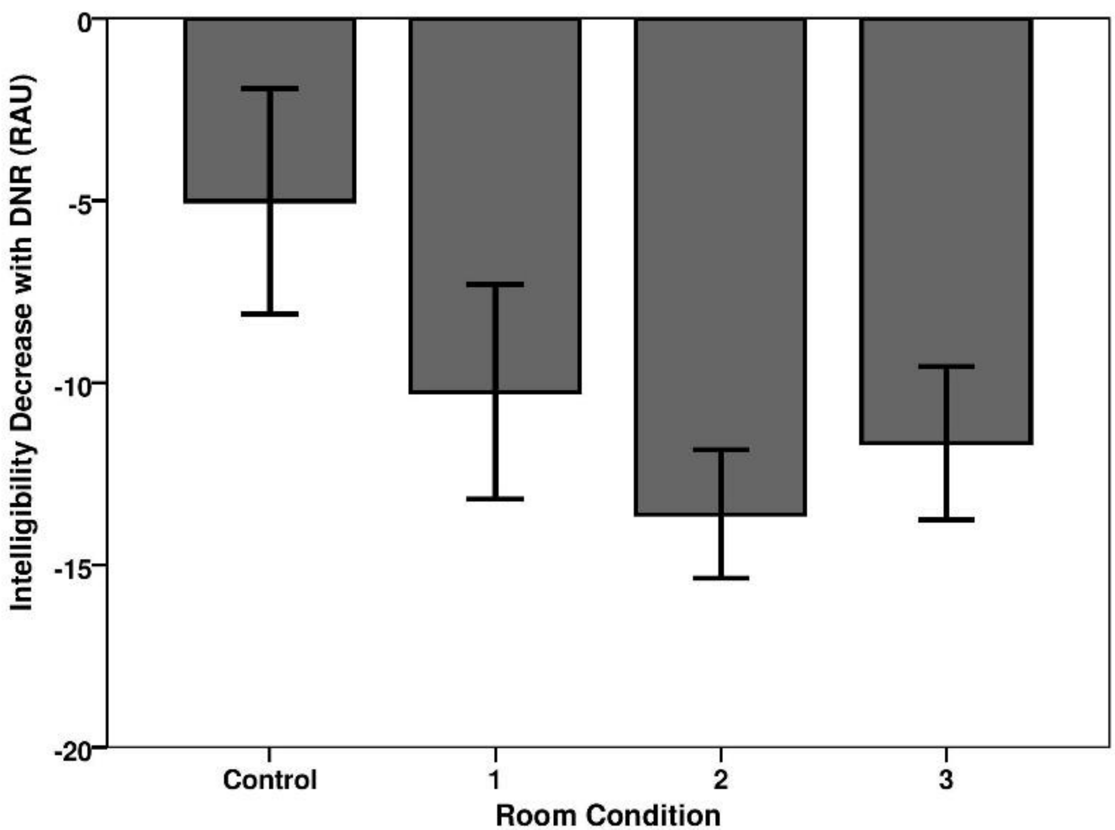


Figure 7. Chapter 2. Discourse Comprehension Test results

Listening comprehension scores on the Discourse Comprehension Test with and without DNR processing across the different room conditions. The noise level was set at 8 dB SNR. The error bars represent +/- 1 standard error of the mean.

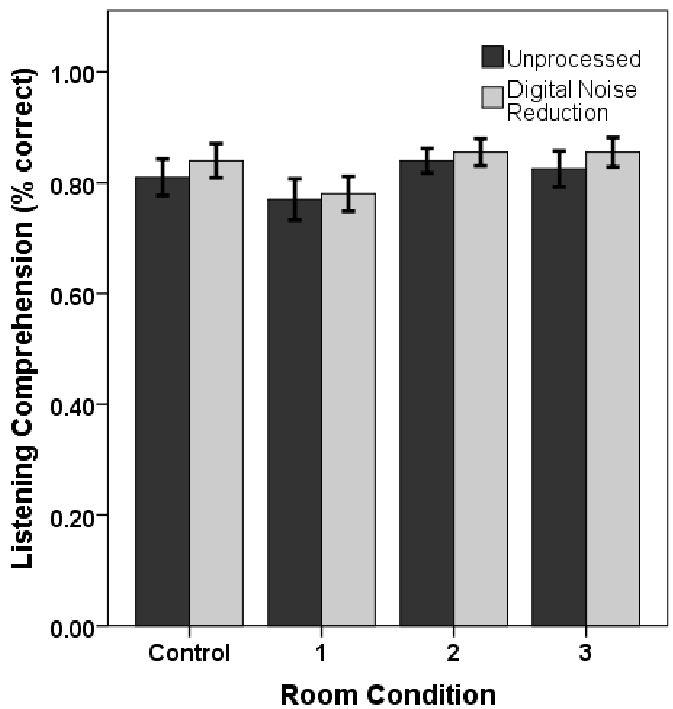


Figure 8. Chapter 2. Subjective ratings with DNR

Mean subjective listener ratings after the story listening task with and without DNR processing across the difference room conditions. Panel A depicts subjective listening effort, Panel B depicts speech naturalness, and Panel C depicts background noise comfort. Y-axes of all the panels are oriented such that higher ratings reflect an improved state for the given scale. The error bars represent +/- 1 standard error of the mean. * p<.05 after Bonferroni correction.

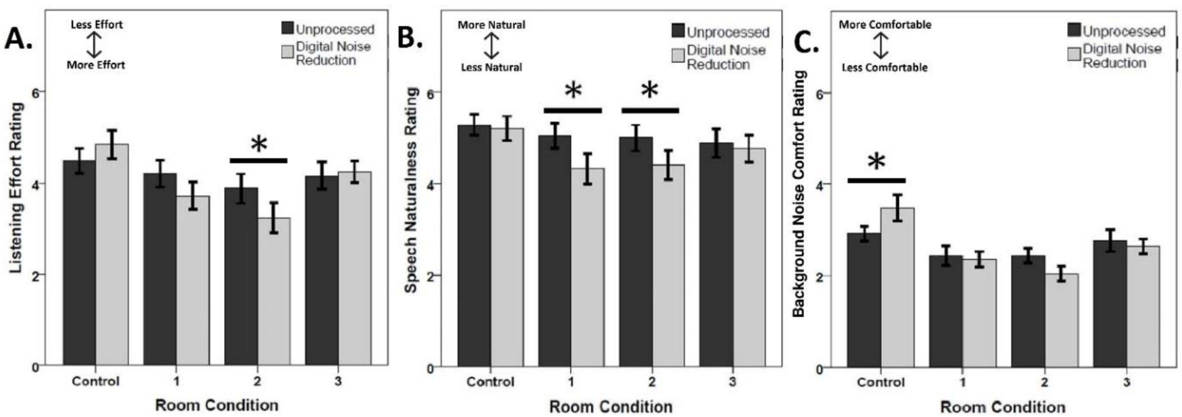


Figure 9. Chapter 3. Effects of WDRC on speech-in-noise signals

Waveforms of speech and noise envelopes at the input of a wide dynamic range compressor (left panel) and at the output of the compressor (right panel). Note that the noise envelope becomes modulated at the output as a result of receiving increased gain during low-intensity portions of the speech envelope.

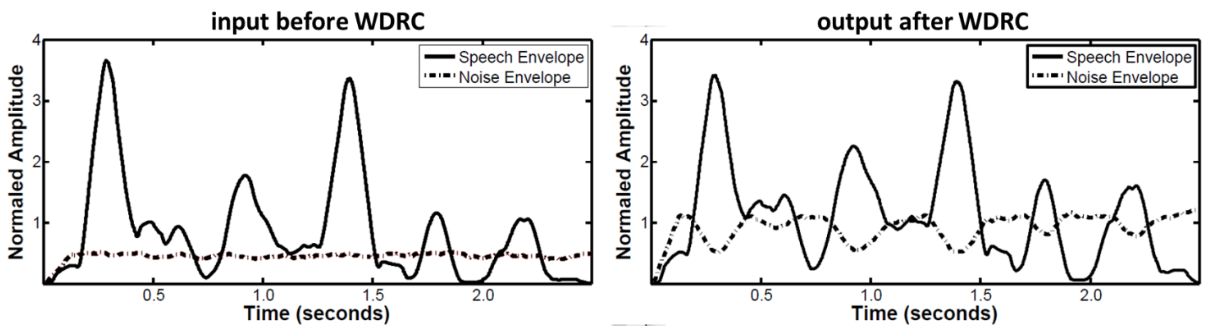


Figure 10. Chapter 3. Signal processing schematic

Signal processing schematic for each of the signal modulation manipulations.

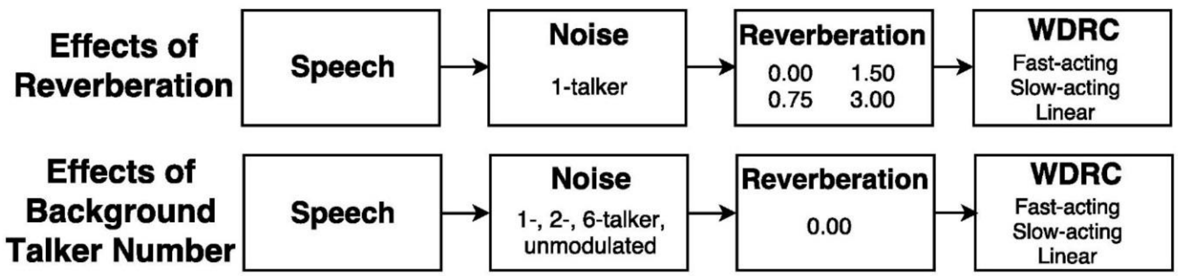


Figure 11. Chapter 3. Acoustic effects of reverberation and WDRC

Mean change in SNR as a result of three different WDRC processing schemes across four different reverberation time conditions with a 1-talker modulated background noise. Each panel depicts a separate input SNR condition. The error bars represent ± 95% Confidence Interval.

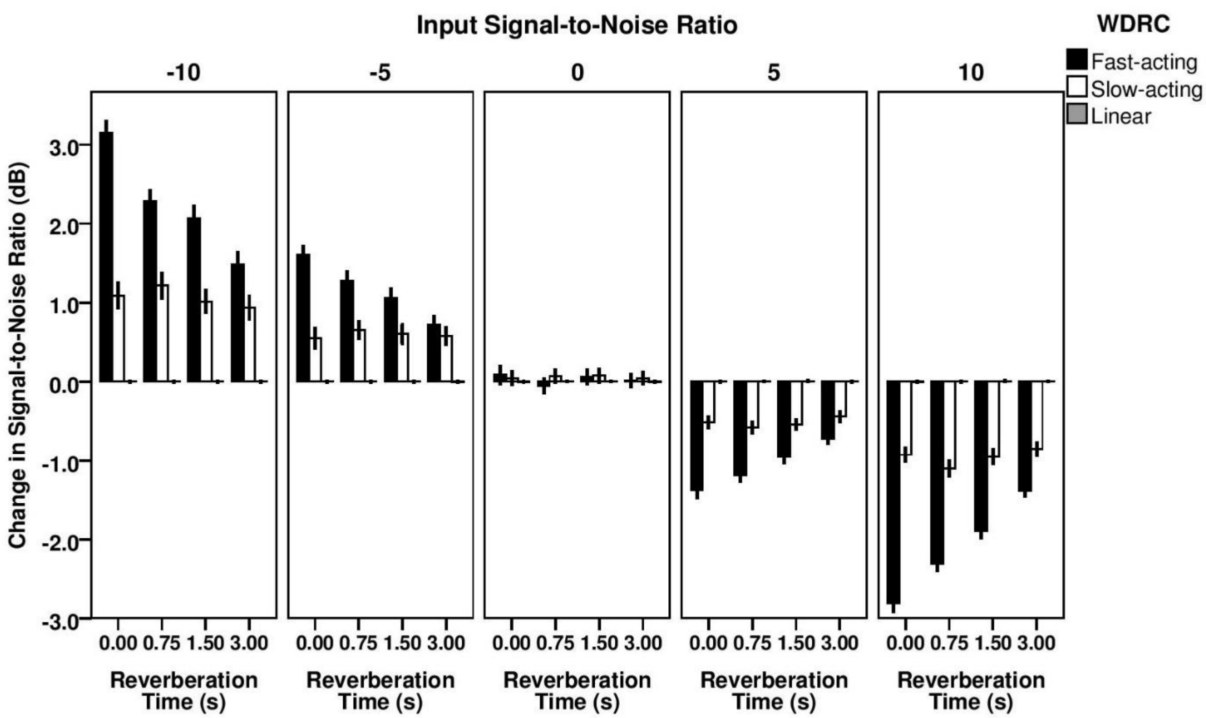


Figure 12. Chapter 3. Acoustic effects of background talker number and WDRC

Mean change in SNR as a result of three different WDRC processing schemes across four background talker number conditions. Each panel depicts a separate input SNR condition. The error bars represent ± 95% Confidence Interval.

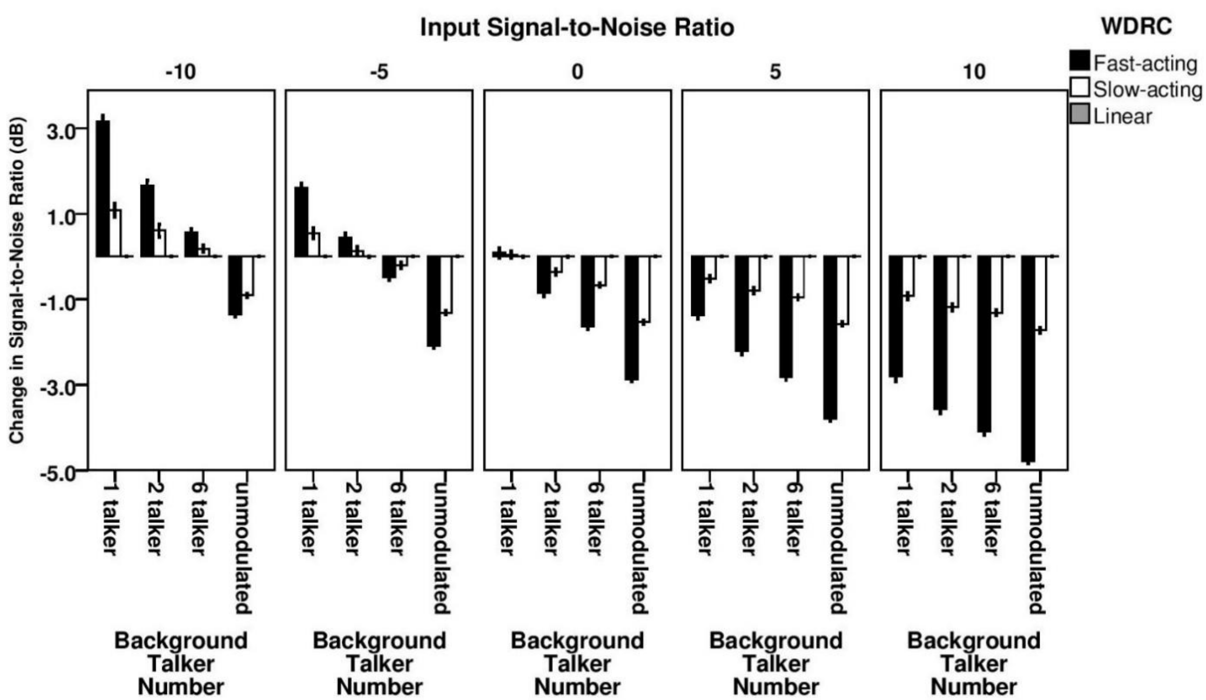


Figure 13. Chapter 4. Participant audiograms II

Mean air-conduction thresholds for each working memory group (high working memory, n=13; low working memory, n=15) for the left and right ears. Error bars represent +/- 1 standard deviation.

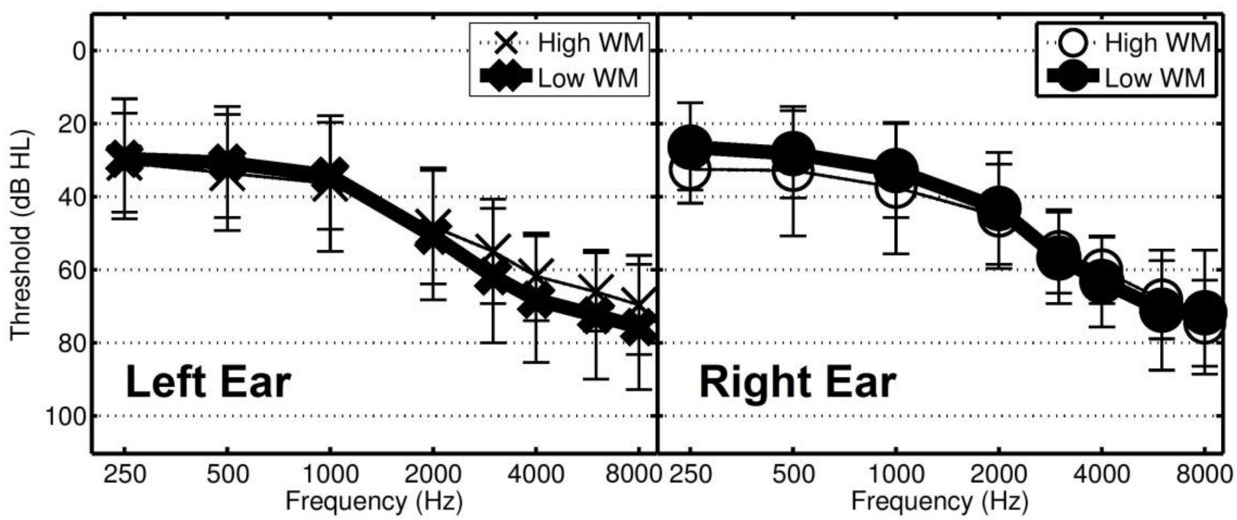


Figure 14. Chapter 4. Raw speech intelligibility results with WDRC

Speech intelligibility scores in with fast-acting and slow-acting WDRC as a function of reverberation time. Results were separated by input SNR (columns) and by working memory group (rows). The error bars represent +/- 1 standard error of the mean.

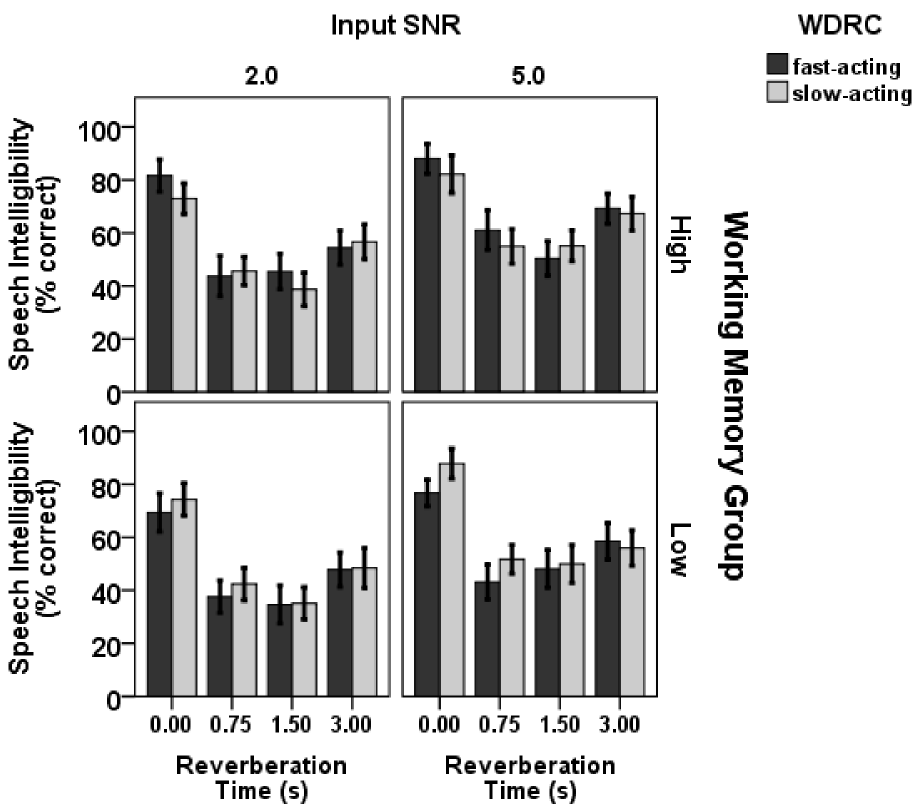
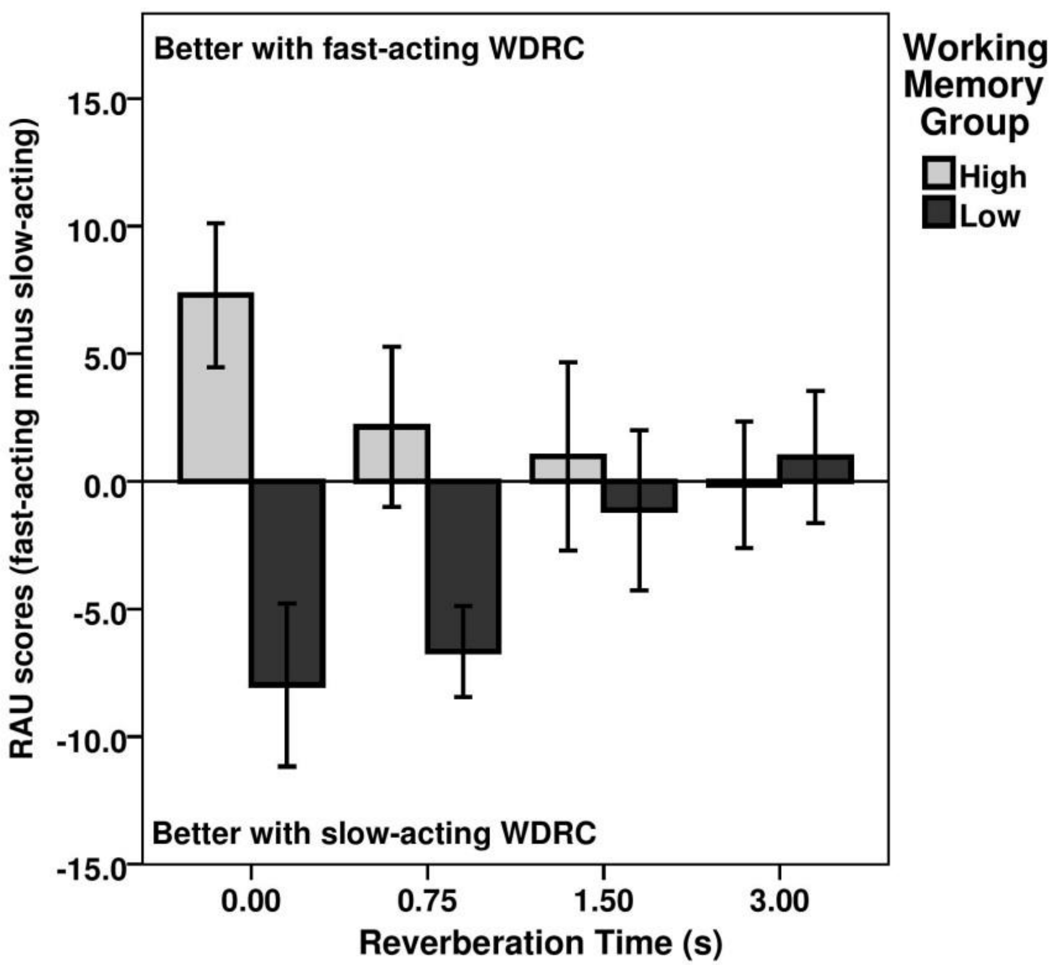


Figure 15. Chapter 4. Transformed speech intelligibility results with WDRC

Mean difference in speech intelligibility scores (RAU-transformed fast-acting WDRC speech intelligibility – RAU-transformed slow-acting WDRC speech intelligibility) for high and low working memory groups as a function of reverberation time. Scores were collapsed across input SNR condition. Positive scores reflect better performance with fast-acting WDRC, and negative scores reflect better performance with slow-acting WDRC. The error bars represent +/- 1 standard error of the mean.



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APPENDIX

Appendix A. Chapter 2. Subjective rating scales for the story listening task

Listening Effort Ratings
7. No Effort
6. Very Little Effort
5. Little Effort
4. Moderate effort
3. Considerable Effort
2. Much Effort
1. Extreme Effort

Speech Naturalness Ratings
7. Completely Natural
6. Natural
5. Somewhat Natural
4. Neither Natural nor Unnatural
3. Somewhat Unnatural
2. Unnatural
1. Completely Unnatural

Background Noise Comfort Ratings
7. Completely Comfortable
6. Comfortable
5. Somewhat Comfortable
4. Neither Comfortable nor Uncomfortable
3. Somewhat Uncomfortable
2. Uncomfortable
1. Completely Uncomfortable

VITA

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