THE IMPACT OF SPECTRALLY ASYNCHRONOUS DELAY ON THE INTELLIGIBILITY OF CONVERSATIONAL SPEECH

by

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Conversationally spoken speech is rampant with rapidly changing and complex acoustic cues that individuals are able to hear, process, and encode to meaning. For many hearing-impaired listeners, a hearing aid is necessary to hear these spectral and temporal acoustic cues of speech. For listeners with mild-moderate high frequency sensorineural hearing loss, open-fit digital signal processing (DSP) hearing aids are the most common amplification option. Open-fit DSP hearing aids introduce a spectrally asynchronous delay to the acoustic signal by allowing audible low frequency information to pass to the eardrum unimpeded while the aid delivers amplified high frequency sounds to the eardrum that has a delayed onset relative to the natural pathway of sound. These spectrally asynchronous delays may disrupt the natural acoustic pattern of speech. The primary goal of this study is to measure the effect of spectrally asynchronous delay on the intelligibility of conversational speech by normal-hearing and hearing-impaired listeners.

A group of normal-hearing listeners (n = 25) and listeners with mild-moderate high frequency sensorineural hearing loss (n = 25) participated in this study. The acoustic stimuli included 200 conversationally-spoken recordings of the low predictability sentences from the revised speech perception in noise test (r-SPIN). These 200 sentences were modified to control for audibility for the hearing-impaired group and so that the acoustic energy above 2 kHz was delayed by either 0 ms (control), 4ms, 8ms, or 32 ms relative to the low frequency energy. The

data were analyzed in order to find the effect of each of the four delay conditions on the intelligibility of the final key word of each sentence.

Normal-hearing listeners were minimally affected by the asynchronous delay. However, the hearing-impaired listeners were deleteriously affected by increasing amounts of spectrally asynchronous delay. Although the hearing-impaired listeners performed well overall in their perception of conversationally spoken speech in quiet, the intelligibility of conversationally spoken sentences significantly decreased when the delay values were equal to or greater than 4 ms. Therefore, hearing aid manufacturers need to restrict the amount of delay introduced by DSP so that it does not distort the acoustic patterns of conversational speech.

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PREFACE

I would like to express my sincere gratitude to my family, friends, and my colleagues at both the University of Pittsburgh and Washington University School of Medicine. All of you have played a part in helping me get to where I am today. I couldn't have done this without any of you. At any point in time, one or more of you have all provided me with encouragement to help me face my fears, inspiration to seek new questions and answers, motivation to be a better person, and discipline to finish this dissertation. Thank you.

1.0 INTRODUCTION & SUMMARY

Speech perception is defined as the auditory perception of phonemic spectral and temporal patterns, and the mapping of these acoustic properties to linguistic units. Although the perception of speech seems to be an effortless task for individuals with normal hearing, it is no small feat. The human auditory system is an extraordinary sensory network that is able to perceive sounds ranging from 0 dB SPL to a sound pressure level that is 10 million times greater (140 dB SPL), and frequencies ranging from 20 Hz to 20,000 Hz. This is an enormous range of acoustic inputs received by the cochlea, the auditory sensory end organ measuring only 34-36 mm in length. In addition to the broad range of sensitivity in the auditory system, humans also have the ability to hear numerous sounds in their surrounding environment, whether it is the conversational chatter from neighboring tables at a restaurant or the engine roar of a subway transportation system, and still attend to the conversation at hand. In a matter of milliseconds, the auditory system is able to detect the spectral and temporal properties of speech, despite a wide variety of adverse acoustic environments, and translate these into a meaningful linguistic message.

The perceptual properties of the human auditory system are astoundingly fine-tuned in three dimensions: intensity, frequency, and time. Normal-hearing listeners can discriminate intensity differences of at least 0.5 to 1 dB, pure tone frequency differences of at least 3 Hz (can be less than 1 Hz for complex tones), and temporal gaps of 2-3 ms (Gelfand, 1998). The acoustic properties of speech are defined along the same three dimensions of intensity, frequency, and

time. Speech contains patterns or cues in each of these three areas, making it an acoustically redundant signal. There have been numerous research studies exploring how normal-hearing listeners use each of these cues in speech recognition. As more is understood about the complex mapping of acoustic cues to phonemes and lexemes in the normal-functioning auditory system, then researchers can qualify and quantify the distortions of the neural network in impaired auditory systems.

For hearing-impaired listeners, the distortions of the neural map between acoustical properties and lexemes begin peripherally with reduced access to the amplitude, spectral, and temporal cues in the speech signal. Loss of audibility accounts for most but not all of the deterioration in speech recognition performance for some listeners with hearing loss (Dubno, Dirks, & Ellison, 1989; Hogan & Turner, 1998). In addition to loss of sensitivity, hearing-impaired listeners have loss of frequency resolution (Moore & Glasberg, 1986). Restoration of audibility can be achieved with amplification devices, but the cochlear spectral distortion is not ameliorated. The goal of an auditory rehabilitation program using amplification is to restore perceptual performance of hearing-impaired listeners to that of normal-hearing listeners by enhancing the acoustic cues of the speech spectrum. Current hearing aid technology uses digital signal processing (DSP) to apply gain, compression, and noise reduction algorithms to the incoming acoustic signal. As a result of the frequency dependent amplitude compression as well as the underlying DSP, the speech spectrum is spectrally and temporally distorted before it is further distorted by the damaged cochlea.

DSP is the core of every amplification device on the market today. While DSP enables the device to perform many complex algorithms on the incoming acoustic signal that purportedly increase the speech recognition performance of hearing-impaired listeners, it introduces a delay

to the signal. The delay caused by DSP, or digital delay, is defined as the amount of time necessary for an acoustic signal to pass through the microphone, DSP circuit, and the receiver of the hearing assistive device. Spectrally asynchronous delays are delay values that vary as a function of frequency bands within the speech spectrum, meaning that the arrival time at the ear drum of one group of frequencies is delayed relative to another group of frequencies (e.g., low frequencies are delayed more than the high frequencies). These delays disrupt the speech's spectral and temporal acoustic patterns that potentially serve as cues for phonemic recognition.

Currently, researchers have been asking the question of whether the spectral and temporal distortions introduced by digital hearing aids and their processing schemes have a deleterious effect on the speech perception abilities of hearing-impaired listeners. Before this question can be addressed, a foundation must be laid. First, the literature on perceptual abilities of normal-hearing listeners were explored not only to glean the various acoustical properties of speech that serve as cues to speech perception, but also to gain insight into the various theories behind the perception and translation of these acoustic cues into linguistic units. Secondly, the speech perception abilities and the use of these acoustical cues by hearing-impaired listeners with mild-moderate sensorineural hearing loss were summarized, and the deviations from normal perceptual performance were quantified. As speech is spoken in a conversational manner outside speech perception laboratories and is the target of amplification by hearing aids, a section was devoted to the acoustic characteristics of conversational speech. Then, the spectro-temporal distortions of the speech spectrum caused by the DSP implemented in modern hearing aids were discussed. Lastly, the literature on the consequences of such distortions on the speech perception of hearing-impaired listeners was reviewed.

The literature review led to the following empirical question: does the introduction of spectrally asynchronous delay that is similar to the delay introduced by open-fit digital hearing aids lead to poorer speech intelligibility of conversationally spoken speech by mild-moderate hearing-impaired listeners? This question led to the construction of conversationally spoken recordings of the revised Speech Perception in Noise (r-SPIN) test (Bilger, Neutzel, Rabinowitz, & Rzeczkowski, 1984) low predictability sentences. These stimuli were chosen because they forced the listener to rely on acoustic cues for speech perception rather than sentence context. The new recordings of the r-SPIN sentences were found to carry all of the trademarks of conversational speech such as faster articulation rate and shorter durations for both vowels and and voice onset time of word-initial consonants, yet were highly intelligible to a group of 15 normal-hearing listeners.

Next, the conversational recordings of the r-SPIN sentences were modified so that the stimuli represented what a hearing-impaired listener hears at the output of an open fit hearing aid. Each sentence was filtered so that the onset of acoustic energy above 2 kHz was delayed relative to the original onset of the sentence. This modified stimulus represented the combination of the natural pathway of sound into the ear canal, and the delayed high frequency energy from the output of the digital hearing aid. For hearing-impaired listeners, this delayed high frequency information was amplified in accordance with the listeners hearing loss. A group of normal-hearing and hearing-impaired listeners listened to and repeated the processed stimuli presented randomly from four spectrally asynchronous delayed conditions: 0 ms delay (served as a control condition), 4 ms, 8 ms, and 32 ms. The intelligibility of the final key-word of each sentence was scored for each listener. The data were then averaged for each group and each delayed condition.

The group of normal-hearing listeners was minimally affected by the introduction of spectrally asynchronous delay of energy above 2 kHz. In fact delays of 32 ms did not significantly alter normal-hearing listeners' intelligibility performance. Hearing-impaired listeners were negatively affected by the introduction of spectrally asynchronous delays in that their performance on the identification of the final key-word was significantly poorer with introduction of a delay as short as 4 ms when compared to the control condition (0 ms delay). However, this degradation in performance was very slight showing that hearing-impaired listeners might be fairly tolerant of short spectrally asynchronous delays. However, hearing-impaired listeners were not as tolerant of the 32 ms delay condition, showing that these listeners may rely more heavily on the spectro-temporal cues for speech perception than the normal hearing listeners who were not affected at all by the 32 ms condition. Therefore, hearing aid manufacturers should be conscious of their devices signal processing speed.

2.0 BACKGROUND

2.1 SPEECH PERCEPTION BY NORMALLY-HEARING LISTENERS

2.1.1 Acoustic Cues

Speech is a complex and rapid-changing code containing acoustic cues structured by the articulatory and aerodynamic mechanisms of speech production. Humans' ability to understand conversation in adverse conditions that degrade the speech signal proves that audibility of the entire speech spectrum is not necessary for speech understanding. It is important to ask, "What acoustic properties of speech are most critical to speech recognition?" Over the past 60 years, researchers have been manipulating the spectrum of both natural speech and synthetic speech in order to answer this very question. As a result, there is an abundance of literature regarding the cues in the acoustic pattern of speech that listeners use to perceive phonemes.

Spectrograms (Figure 2-1) display the pattern of speech acoustics along three dimensions: time along the abscissa, frequency along the ordinate, and intensity represented by the darkness or boldness of the bands found within the spectrum. Cues to aid speech perception result from the variance of each dimension with respect to another. These cues are classified as spectral, intensity, temporal, spectro-intensity, tempo-intensity, and spectro-temporal cues. For spectral, intensity and temporal cues, a single acoustic property provides a pattern resulting in

phoneme recognition, while the two remaining domains are held constant (i.e., in spectral cues the frequency patterns serve as cues while intensity and time are held constant). While spectro-intensity, tempo-intensity, and spectro-amplitude cues are defined as dynamic patterns that emerge when one acoustic property varies as a function of another property. For example, an acoustic pattern generated by frequency varying as a function of time is a spectro-temporal cue. Table 2-1 contains an acoustic cue matrix showing the variation of the cues for phonemic perception along the acoustic dimensions.

Table 2-1: Acoustic Cue Matrix

	Spectral	Intensity	Temporal
Spectral	• Vowel perception (formant spacing) • Fricative perception (spectral shape of noise) • Plosive perception (spectral shape of burst)	• Plosive perception (Burst intensity & consonant-vowel amplitude ratio)	• Consonant perception (formant transitions—slope and duration) • Voicing perception (VOT, TOT, EOA)
Intensity		• Voicing perception in consonants (F0) • Perception of nasals (weaker intensity)	• Manner of production • Voicing perception in consonants (amplitude envelope)
Temporal			• Voicing perception in final-position consonants (vowel duration) • Fricative perception (duration of noise) • Manner of production (periodicity)

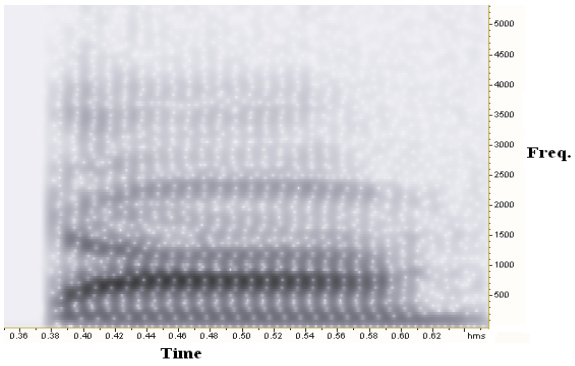


Figure 2-1: Spectrogram of the CV syllable /da/. Frequency is represented on the y axis, while time is represented on the x-axis. Intensity is shown by the darkness of the bars within the spectrogram.

2.1.2 Spectral Cues

Spectral cues result from the resonances (formants) of the vocal tract during speech production. When voicing is present, the spectral cues aid vowel recognition. Vowel perception is dependent on the spectral pattern or spacing of the vowel formats. Delattre and his colleagues (1952) at the Haskins Laboratory, where much of the early work on speech acoustics and perception originated, used synthesized speech to manipulate the frequency spacing of the first and second formant (F1 and F2). They presented the variations of the formant pairs to listeners to find the pattern that resulted in the highest accuracy of vowel identification. Results indicated that listeners use formant spacing (the frequency difference between F1 and F2), as a cue for vowel identification. For example, results indicated that the difference between a listener perceiving the vowel /i/ as in “beet” and /u/ as in “boot” was that /i/ had a much higher F2 and

subsequent formant spacing than /u/. Figure 2-2 shows the formant patterns that serve as spectral cues for each labeled vowel.

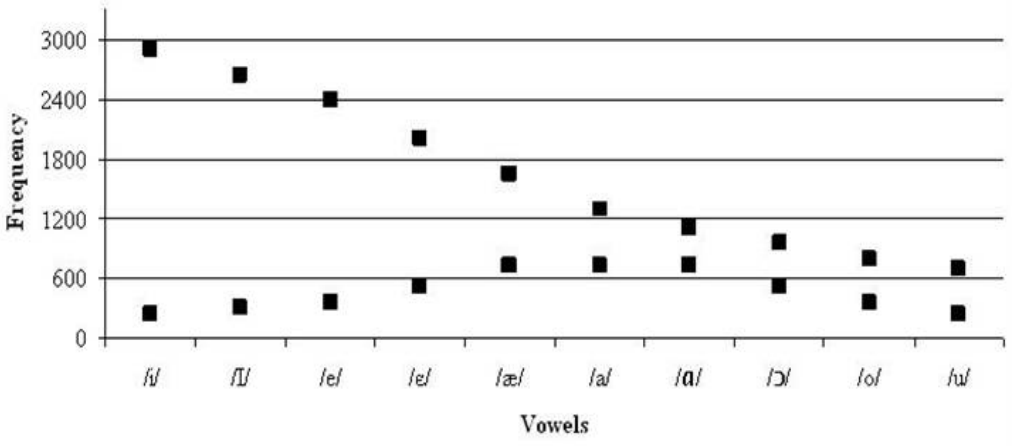


Figure 2-2: The values and relative pattern of F1 (lower bars) and F2 (higher bars) for each of the labeled vowels (from Delattre et. al., 1952).

Spectral cues also contribute to the perception of fricatives. Due to the site of constriction in the oral cavity during production, the frication noise takes on a variety of spectral resonance patterns (Heinz & Stevens, 1961). The palatal fricatives /ʃ/ and /tʃ/ have the largest resonant cavity, so the high frequency noise spectrum is concentrated above 2 kHz. However the spectral shape of the alveolar fricatives /s/ and /z/ are contained above 4 kHz. For the most anterior-produced fricatives /f/ and /v/, the spectral shape of the noise is more broad-band due to a lack of resonant oral cavity. Listener's can use their perception of the spectral energy of the noise to accurately label the phoneme presented (Harris, 1958; Heinz & Stevens, 1961; Jongman, 1989).

Just as there are spectral cues in the noise of the frication, there are also such cues in the noise of the burst in plosive voiceless consonants. The frequency of the burst in relation to the second formant of the following vowel aids listeners in determining the place of constriction for the consonants /p/, /t/, and /k/. Liberman, Delattre, and Cooper (1952) showed that bursts

containing high spectral energy were perceived as /t/ by listeners, while bursts with spectral energy lower in frequency relative to the vowel's F2 were perceived as /p/. Listener's reported hearing /k/ when the spectrum of the burst was slightly higher than the following vowel's F2.

Perception of nasal consonants is accomplished by spectral cues. Nasal consonants have a perceptual feature called nasal murmur which is an additional resonance around 250 Hz (Mermelstein, 1977). This low frequency spectral cue aids listeners in distinguishing the nasal manner of the consonant being produced.

2.1.3 **Intensity Cues**

Intensity cues are acoustic perceptual features resulting from changes in the amplitude of phonemic production. Intensity is the most salient cue for detecting voicing in vowel and consonant production (Ohde, 1984). Presence of vocal-fold vibration results in overall acoustic patterns that are more intense than those produced without vibration. Intensity cues also are used to differentiate the manner of production between nasal and non-nasal consonants, in addition to the aforementioned spectral cue of nasal murmurs. Fant (1952) found that nasal consonants tend to have formants with weaker intensities than the neighboring vowels. Mermelstein (1977) also found that the lower intensity level of the energy band in the upper formants separate the nasally from the non-nasally-produced consonants.

2.1.4 **Temporal Cues**

Speech contains temporal variations that serve as acoustic cues in phonemic recognition. Temporal cues aid the listeners in perceiving the presence or absence of voicing in consonants

(House, 1961). Raphael (1972) found that for consonants in the final position (VC) lengthening the preceding vowel duration resulted in listeners perceiving more voiced consonants even when the final consonant was actually a voiceless production. When the vowel of the word "bet" was prolonged, the listeners' perception of the final consonant changed to that of a voiced /d/ as in "bed".

The duration of the noise in fricatives provides a cue for the perception of consonant voicing. Voiceless fricatives such as (/f/, /s/, and /ʃ/) tend to have longer durations of noise than their voiced counterparts (/v/, /z/, and /tʃ/) (Baum & Blumstein, 1987; Jongman, Wayland, & Wong, 2000). Temporal cues also distinguish between the perception of fricatives and affricates. A silent duration preceding /ʃ/ in the word "hash" causes the perception to shift toward /tʃ/ as in "hatch" (Raphael & Dorman, 1980).

Another temporal cue called periodicity is an important property of speech. Periodic and quasi-periodic phonemes such as vowels and voiced consonants fluctuate at rates between 50-500 Hz. Aperiodic phonemes such as fricatives and voiceless plosives typically have fluctuation rates above 1 kHz. Listeners use this temporal cue not only to determine voicing and manner of production, but also to determine pitch because the rate of periodicity reflects the fundamental frequency of the voice (Rosen, 1992).

2.1.5 **Spectro-Intensity Cues**

All of the acoustic cues defined thus far have been one-dimensional. The remaining cues can be described as bi-dimensional, meaning that one acoustic feature varies as a function of another. Spectro-intensity cues are those in which changes in the intensity of certain frequency regions elicit different phonetic percepts. For voiceless stop consonants /p/ and /t/, there are

spectral cues that distinguish the place of production with the noise burst of the alveolar consonant /t/ having a higher frequency spectrum than the labial consonant /p/. Ohde and Stevens (1983) conducted a study examining the effect of burst amplitude on the perception of /pa/ and /ta/. The authors constructed a continuum of nine synthetic speech tokens with the spectral cues varying in steps from that of /pa/ to that of /ta/. They found that by enhancing the amplitude of the burst relative to the vowel energy, listeners tend to rate the sound as being /ta/ although the spectral cues correspond with the bilabial /pa/. Conversely, if the amplitude of the burst was decreased relative to the vowel energy, then the listeners were more likely to perceive /pa/, despite the fact that the spectral cues indicated /ta/. Similar relative amplitude cues also exist for the fricative consonants /s/ and /ʃ/ (Hedrick & Ohde, 1993).

2.1.6 **Tempo-Intensity Cues**

The speech spectrum contains gross tempo-intensity variations commonly referred to as the "amplitude envelope". The amplitude fluctuation rate is characterized as having slow rise/fall times with the fluctuation rate being between 2-50 Hz (Rosen, 1992). Van Tasell and her colleagues (1987) explored the role of amplitude envelope on speech perception. In their experiment, the authors generated 19 speech waveform envelope noises that corresponded to one of 19 vowel-consonant-vowel utterances. The noise waveforms were then low pass filtered so as to extract the amplitude envelope. Listeners were then asked to identify the consonant of the filtered noise. The authors found that the listeners' closed-set identification of the 19 consonants was above chance. Error analysis revealed that listeners were able to use the amplitude envelope cue to correctly group the utterances according to manner of production and presence of voicing. Numerous studies involving cochlear implants and narrow-band filtering have further confirmed

the importance of amplitude envelope cues (Warren, Reiner, Bashford, & Brubaker, 1995; Shannon, Zeng, Kamath, Wygonski, & Ekelid, 1995; Shannon, Zeng, & Wygonski, 1998).

2.1.7 **Spectro-Temporal Cues**

In his work describing the temporal information of speech, Rosen (1992) divided speech's temporal aspects into three features. The first two are the previously mentioned periodicity and amplitude-envelope cues. The third feature called the fine-structured cue reflects the variations of the spectrum over short-time intervals. These cues display the rapid spectral movement over time. Examples of these spectro-temporal cues are formant transitions, duration of transitions, and voicing onset cues.

Formant transitions result from the changes in the resonance of the articulatory mechanism as it smoothly transitions from one production stance to another. Figure 2-3 shows the F2 transitions from the consonants /b/, /d/, and /g/ to the vowels /i/, /e/ /ɛ/, /a/, /ɔ/, /o/, and /u/. The F2 transitions from the phoneme /b/ to each vowel is characterized as having a rising transition, while the /g/ to vowel F2 transition is characterized as falling. The direction of the alveolar /d/ phoneme F2 transitions vary as a function of the following vowel (Delattre, Liberman, & Cooper, 1955). By systematically varying the direction and slope of the F2 transition, the listener's perception changes between /ba/, /da/, and /ga/ (Liberman, Harris, Hoffman, & Griffith, 1957).

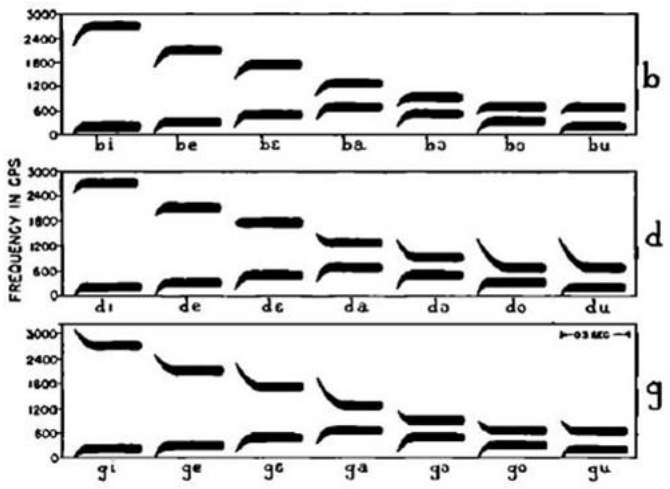


Figure 2-3: The second formant transitions for /b/, /d/, and /g/ and the each of the labeled vowels (from Delattre et. al., 1955).

Stevens (1980) reported that this rapid change in the spectrum occurs over the first 10-30 ms of the utterance. Psychoacoustic experiments have shown that by keeping F1 steady and removing the first 50 ms of the F2 transitions for the syllables /bV/, /dV/, and /gV/, listeners' perception changes in that they report that they no longer hear the consonants /b/ and /g/, but hear /d/ paired with different vowels (Delattre et al., 1955). It is apparent that there is an acoustic wealth of information indicating the place of articulation during the brief formant transition period. Spectro-temporal cues also differentiate the semivowel glides /r/ and /l/, in that /r/ is marked by a low rising F3 and /l/ is characterized by a high falling F3 (Lisker, 1957). Longer formant transitions (> 40 ms) indicate the production of semivowel glides /w/ and /j/. Liberman, Cooper, Shankweiler, and Studdert-Kennedy (1956) found that by extending the transition durations of /bɛ/ and /gɛ/ to 40-50 ms listeners reported hearing /wɛ/ and /jɛ/. Formant transitions greater than 150 ms are perceived as diphthongs (Gay, 1970).

Spectro-temporal cues play a role in determining the presence or absence of voicing in the production of consonants. Stevens and Klatt (1974) found that listeners use two spectro-temporal cues in the perception of voicing. The first cue was the duration of the formant transitions between the consonant and the vowel, with transitions of 10-30 ms perceived as voiced. Voiceless consonants have minimal or even negligible formant transitions. This is due to the transitions occurring during the voiceless burst of the initial plosive. The second cue the authors mentioned for voiced-voiceless distinction was voice onset time (VOT).

Voice onset time is defined as the interval between the release of a stop occlusion and the onset of voicing. Voiced initial stops in which the voicing occurs at the same time or immediately following the release burst tend to have short VOT of no more than 20 ms. Voiceless stops in which the voicing lags behind the release burst have VOT greater than 25 ms (Lisker & Abramson, 1964). There is a clear and distinct categorical boundary between English voiced and voiceless cognate pairs around 20 ms, where consonants with VOT greater than 20 ms tend to be perceived as voiceless and those with VOT less than 20 ms tend to be perceived as voiced. The location of this boundary at 20 ms confirms earlier research on auditory perception of temporal order (Hirsh, 1959).

The frequency characteristic of voicing tends to be the fundamental frequency of the voice while the frequency characteristic of the release burst tends to be high frequency noise. Therefore VOT can be described as the temporal relationship between low and high frequency components. Pisoni (1977) carried out a study using non-speech tonal stimuli to mimic the VOT feature of stop consonants. He used a low frequency 500 Hz tone to mimic the low frequency voicing property and a 1500 Hz tone to represent the high frequency component of the burst and varied their relative tone onset time (TOT) between -50 and + 50 ms. After training, listeners

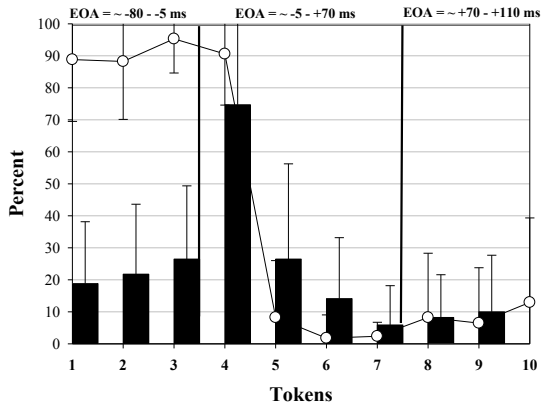
produced sharp identification boundaries around -20 and +20 ms, paralleling the results of the earlier VOT studies.

Recently, Yuan and colleagues (2004) proposed an acoustic cue called Envelope Onset Asynchrony (EOA) that serves to distinguish a voiced consonant from its voiceless cognate in a similar manner as VOT and TOT. EOA uses the time asynchrony between the onset of two frequency bands of energy in the speech spectrum, the first being low passed at 350 Hz and the second being high passed at 3000 Hz to determine whether the consonant is voiced. For an articulated initial voiced consonant, low frequency energy either occurs before or simultaneously with the onset of the high frequency energy in the speech spectrum. In contrast, the onset of low frequency energy tends to follow the onset of high frequency energy for initial voiceless consonants. EOA is derived from subtracting the onset of the high frequency energy band from the onset of the low frequency energy band. Theoretically, the EOA for initial voiced consonants should be a negative value or zero and the EOA for initial voiceless consonants should be a positive value. In the acoustical analysis of two speakers' speech spectrum, the authors found the overall mean EOA of 8 voiced consonants to be -12.4 ms and the overall mean EOA of 8 voiceless consonants to be 142.5 ms.

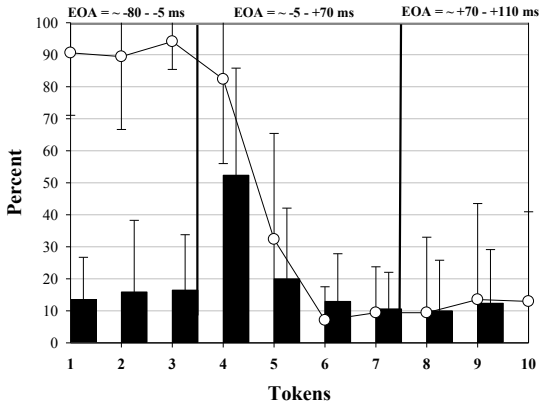
Ortmann, Palmer, and Pratt (2010) tested a group of listeners in order to determine whether EOA is a spectro-acoustic cue used by listeners for determining the presence of voicing in initial consonants. Recordings of six individual consonant-vowel syllables /ba/, /pa/, /da/, /ta/, /ga/, and /ka/ were filtered into two frequency bands, a low frequency band below 350 Hz and a high frequency band above 3000 Hz. For each syllable, these bands were delayed in time relative to one another in 25 ms steps so that an EOA continuum was generated for each CV token. Listeners completed a 2 alternative forced choice labeling and discrimination task for each

continuum. Figure 2-4 shows the normal-hearing listeners' group average labeling and discrimination data for the syllables /pa/, /ta/, and /ka/. For each of the graphs, the percent of the listeners' responses indicating a voiced percept is plotted as a function of EOA changing from a negative value to a positive value. For example, in Figure 2-4a the CV syllable /pa/ is perceived by listeners as /ba/ when the EOA is manipulated to have a negative value (far left of the graph). Overall, the results indicate that as the temporal onset asynchrony between low and high frequency bands of speech is manipulated, listeners' perception of the consonant's voicing properties changed from voiced to that of its voiceless cognate.

a) NH-/pa/



b) NH-/ta/



c) NH-/ka/

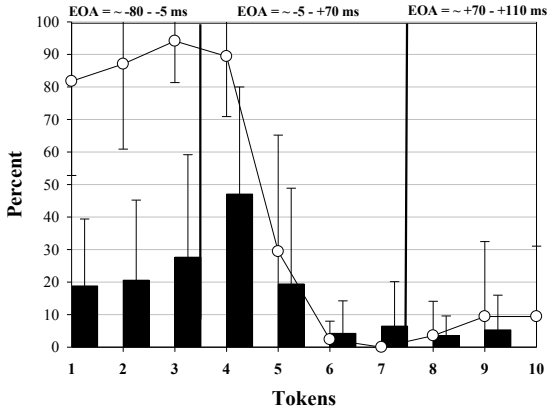


Figure 2-4(a-c): Normal-hearing listeners' identification and discrimination data for the EOA continuums, a) /pa/ b) /ta/ and c) /ka/. For each of the graphs, the x-axis displays the 10 tokens representing the shift in EOA from a more negative value to a more positive value. The space in between each token value represents the adjacent token pairs (i.e., token 1 paired with token 2, token 2 paired with token 3, and so on). The values along the y-axis are in percent. The line graph represents the data from the labeling task, so higher on the y-axis means that the listeners' perception is voiced, while lower values represent a more voiceless percept. The bar graph displays the discrimination data, so higher y-axis values mean that a greater difference between the token pair was detected.

2.1.8 **Coarticulation**

These spectral, intensity, temporal, spectro-intensity, tempo-intensity, and spectro-temporal cues are many of the overlapping acoustic cues that aid listeners in recognizing spoken language. These sub-phonemic cues have a complex map to phonemes in that there is not a one-to one correspondence between these aforementioned cues and the recognized phoneme. The map complexity is due in large part to coarticulation. Coarticulation occurs when two or more phonemes are produced with temporal overlap (i.e., lip rounding of the production of /s/ when saying "soon"). Lindblom (1963) found that the vowel 2nd formant spectral cues (recall Figure 2-2) varied by as much as 70% when the vowel was produced following different consonants /b, d, g/ and with different syllable durations. Amazingly, listeners partaking in everyday conversation with many different speakers are still able to recover the intended phoneme and subsequent message despite all of this acoustic variability in speech (Smith, 2000).

Another lack of perfect relationship between acoustic cues and perceived phonemes is evidenced in the F2 transition cue for the consonant /d/. According to the second row of Figure 2-3, the transition cue for /d/ in the syllable /di/ (far left) is a rising transition. However, the same cue for /d/ in /du/ (far right) is a steeply falling transition. It is apparent that completely opposite cues are eliciting the same percept. Further studies have found that listeners are able to compensate for the acoustic variability of coarticulation (Lindblom & Studdert-Kennedy, 1967; Mann, 1980; Mann & Repp, 1980; Mann & Repp, 1981; Holt, Lotto, & Kluender, 2000). Mann (1980) found that target speech sounds are shifted by the preceding phonetic context. In this experiment listeners identified ambiguous target stimuli as either /da/ or /ga/ when they were preceded by /ar/ or /al/. The spectral cues for each of these individual consonants in isolation are 1) /d/ has a high F3 onset frequency while 2) /g/ has a lower F3 onset, 3) the F3 frequency for /r/

is low frequency compared to 4) /l/ which has a high F3 frequency offset. Coarticulation of the syllables /arda/ would result in the lowering of the F3 frequency for the consonant /d/ by the low frequency offset of the consonant /r/. The syllables /alga/ would have an acoustic pattern in which the third formant of /g/ would be raised by the temporal overlap with the consonant /l/ (See Figure 2-5a-d for a schematic diagram of this coarticulation). The results showed that listeners identified the ambiguous phoneme as /da/ when the precursor syllable was /ar/ and identified /ga/ more often in the context of /al/. Listeners were able to use context of the preceding phoneme (/ar/ & /al/) to correct for the spectral variability of the neighboring phoneme (/da/ & /ga/).

This puzzling relationship between speech acoustics and perception has caused researchers to develop several theories as to how listeners can withstand so much acoustical variation due to coarticulation and still retain phonemic and subsequent lexical recognition. These theories are divided into those that are articulation-based theories and those that are auditory-based theories (See Diehl, Lotto, & Holt, 2004 for a good review of speech perception theories).

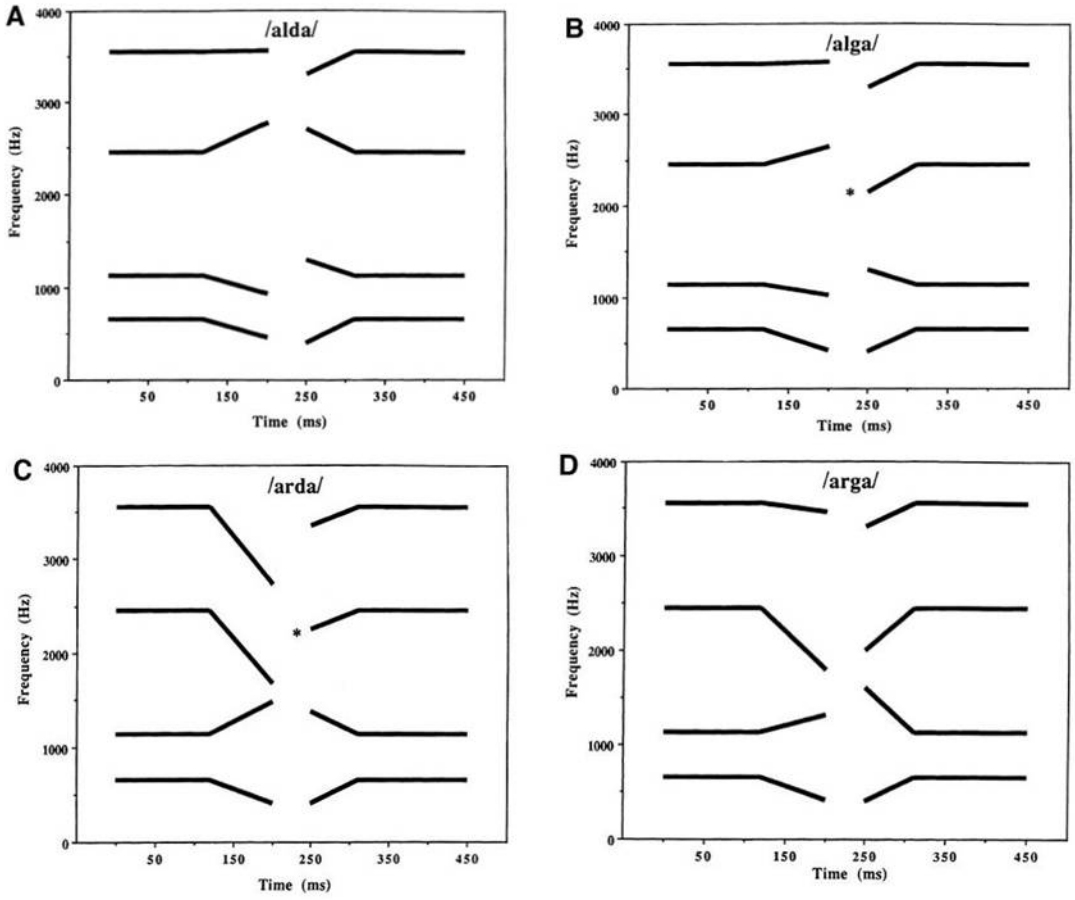


Figure 2-5(a-d): Schematic spectrogram taken from Lotto & Kluender (1998) showing the coarticulatory effects of preceding /al/ on /da/ and /ga/ and /ar/ on /da/ and /ga/. Note the similarities in formant transitions of /ga/ and /da/ in (B) and (C). Also notice the spectral contrast between the third formant of the preceding consonants /l/ and /r/ and the following consonants /d/ and /g/. In (A) and (B), there is more contrast or disparity between F3 in /alga/ than /alda/. In (C) and (D), note the spectral contrast in /arda/ that is not present in /arga/.

2.1.9 Theories of Speech Perception

2.1.9.1 Articulation Based Theories

The Motor Theory of speech perception was developed in the 1960's by the research group at the Haskin's Laboratory (Liberman et al., 1967; Liberman & Mattingly, 1985). This theory was conceptualized after the group spent nearly two decades capturing and linking acoustic properties to phonetic structure. In 1957, Liberman and his colleagues constructed a continuum of synthetic CV syllables varying in the slope of the F2 transition that resulted in a perceptual continuum spanning /ba/ to /da/ to /ga/. These CV syllables were then presented to listeners to either identify the initial consonant or to discriminate between pairs of adjacent stimuli on the continuum. The results displayed a sharp boundary between the perceived phoneme categories corresponding with a peak in discrimination accuracy between categories and then falling to chance within categories. This effect is known as categorical perception. Liberman and his colleagues, knowing the complex and variable map between the absolute values of acoustic cues and phonemic perception, began to explain categorical perception in terms of articulation rather than acoustics. They argued that the boundaries of the categories are more coincidental with the articulator's place of production than the acoustic properties of speech. Because of this seemingly one-to-one correspondence between phonemes and articulation, it was hypothesized that phonemic analysis occurs along the speech neuromotor pathway rather than the auditory pathway.

The Motor Theory accounts for perceptual recovery from coarticulation by stating that articulated phonemes are perceived by listeners via their own neuromotor commands. This way the listener perceives the intended gesture rather than the actual coarticulated gesture made by the speaker. Figure 2-6 displays the model of the Motor Theory of speech perception. In this

model, the auditory pathway is only responsible for receiving the acoustic patterns at the subphonemic level. An important aspect of this theory is that there exists a specialized speech decoder that breaks down the acoustic features into information about the vocal tract shape and articulatory gestures. Liberman and Mattingly (1985) claim that this module, which is unique to humans, enables listeners to recover from the acoustic consequences of coarticulation such as the lowering of F3 onset in /da/ when spoken in the bi-syllable /arda/. The recovered intended gestures are then sent to the neuromotor network where the features are extracted and linearly mapped to the appropriate phoneme. According to the Motor theory, this speech module is responsible for the human listener's ability to separate speech in the presence of multiple talkers in that this module recognizes the presence and dynamics of more than one vocal tract shape and computes the resulting acoustic resonances, then separates the gestures according to vocal tract shape (Dorman, Raphael, & Liberman, 1979).

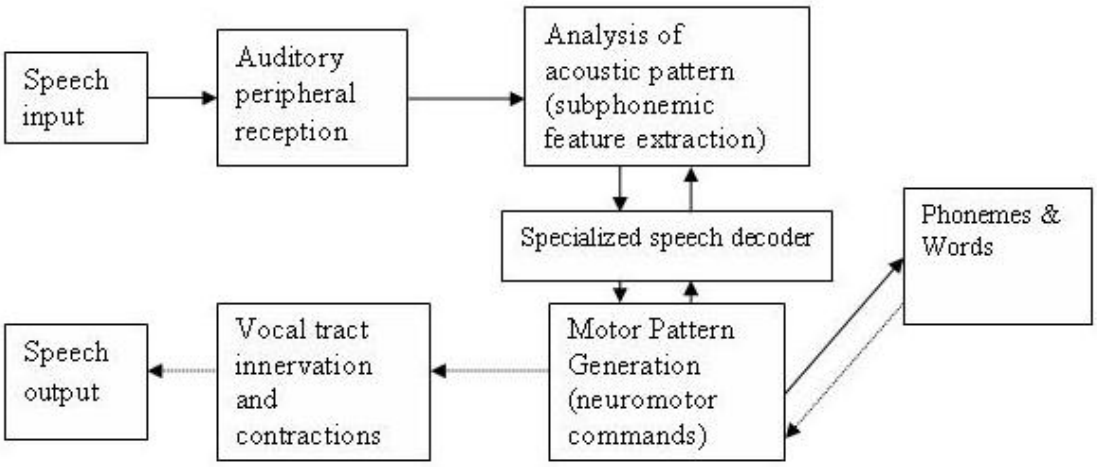


Figure 2-6: A simplified diagram of the Motor Theory of speech perception. Note the use of neuromotor commands for speech production.

The Direct Realist Theory, like the Motor Theory, is a gestural account of speech perception in which perception is linked to production (Fowler, 1986). While motor theorists believe that phonemic recognition occurs by recovery of the intended gestures of speech through a specialized speech decoder and neuromotor commands, the direct realist believes that listeners recover the talker's actual gesture such as lip closure in the syllable /ba/. The Direct Realist Theory does not require features to be extracted from the acoustic signal because the acoustic signal is just a medium to transport gestures. Listeners do not perceive the spectral, intensity, and temporal properties of speech, but rather the speaker's actual articulation of phonemes. This theory draws from the notion that when people rely on their haptic senses to feel and recognize a certain object, the senses do not quantify the amount of pressure on the skin of their fingertips, but rather seek information about the object itself (i.e., round, hard, smooth with stitches, and size = baseball). Therefore it was hypothesized that listeners do not extract the actual acoustic properties of speech to recognize sounds, but rather use the acoustics as a medium to perceive the phonetically structured articulators.

Figure 2-7 displays a schematic diagram of the Direct Realist Theory (Folwer, 1996). This theory explains the perceptual hardiness toward coarticulation, as in the /arda/ example by stating that the syllable /ar/ and /da/ are two separate and independent phonemic units that are co-produced. The listener recovers the two gestures and perceives the /da/ in /arda/ despite the acoustic variability of F3 (Fowler, 2006).

The Analysis by Synthesis Theory, developed by Stevens and Halle (1967) at Massachusetts Institute of Technology, combines both a gestural and an auditory approach to speech perception. In this model (Figure 2-8), a listener perceives the acoustic pattern of speech, then generates a hypothesis regarding the phonemic structure of the utterance. The listener then

rapidly generates an internal auditory model of his own production of the same utterance. If the input's overall acoustic pattern matches his projected ouput, then the listener accepts his hypothetical perception and accurate speech perception occurs. By analyzing the input in terms of his synthesis of the utterance, the listener normalizes the variability due to coarticulation. This theory was abandoned soon after it was created as the authors began to support auditory-based theories of speech perception.

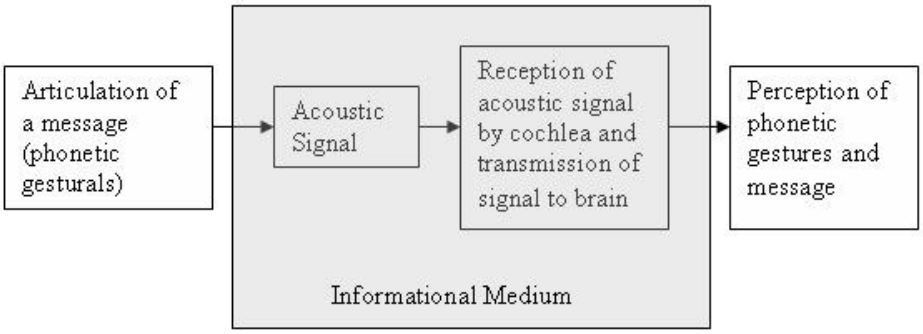


Figure 2-7: A simplified diagram of the Direct Realist Theory of speech perception. Note the lack of acoustic and phonetic feature extraction.

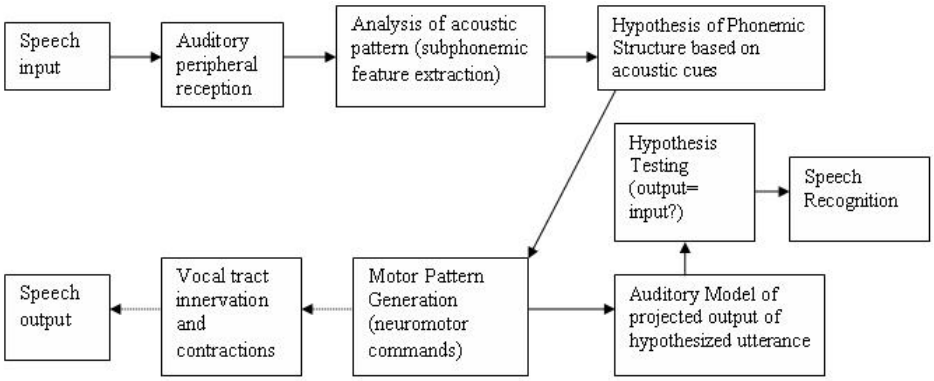


Figure 2-8: A simplified model of the Analysis by Synthesis Theory of speech perception. Note the combination of both auditory and gestural processes for speech perception.

2.1.9.2 Auditory Based Theories of Speech Perception

The Auditory Approach to speech perception states that phonemic and lexical recognition is achieved by the recovery of the acoustic properties of speech by the auditory sensory and cognitive network. Although the Acoustic Approach acknowledges that acoustic cues are generated by and correlated with articulatory gestures, speech understanding does not involve the perception of these gestures, nor is perception tied to production. Instead, acoustic cues are directly encoded as phonemes (Diehl et al., 2004). The Auditory Approach evolved as a result of several findings that challenged the articulation based theories of speech perception. These findings include results that indicated that some invariant acoustic cues to speech perception exist (Blumstein & Stevens, 1979), findings that animals were able to exhibit speech perception abilities (Kuhl & Miller, 1975; Kuhl & Miller, 1979; Kluender, Diehl, & Kileen, 1987), and data that demonstrated that human listeners can perceive non-speech stimuli similarly to speech stimuli (Stevens & Klatt, 1974; Pisoni, 1977; Holt et al., 2000).

Cole and Scott (1974) proposed that listeners' identification of at least three acoustic cues is crucial to accurate speech perception: invariant acoustic cues, context-conditioned cues, and waveform envelope cues. They described invariant cues as the acoustic cues that accompany a particular phoneme in any vowel context. Blumstein and Stevens (1979) stated that these invariant cues occur in the first 10-20 ms after the release of stop consonants. The authors analyzed the onset spectra of stop consonants paired with different vowels spoken by different speakers. They found three general templates that classified voiced and voiceless stop consonants according to place of articulation. The onset spectra of labial /b, p/ tend to have a flat or falling amplitude-frequency pattern, while alveolar /d, t/ possess an amplitude rising spectra. Velar consonants /g, k/ display a compact mid-frequency energy spectra. These three general

templates correctly identify the place of articulation 85% of the time across many speakers. An additional perceptual study found that listeners can appropriately categorize synthetic CV stimuli constructed with the differing onset spectral templates (Blumstein & Stevens, 1980).

Another invariant cue mentioned was VOT to indicate the presence of voicing during stop consonant production. In quiet listening environments, although both VOT and the duration of F1 transition are cues, the duration of VOT alone is a salient cue to indicate voicing (Stevens & Klatt, 1974; Lisker, 1975; Summerfield & Haggard, 1977). In noisy environments the low amplitude of the spectral burst is obliterated by the spectrum of the background noise. In line with Cole and Scott's theory that listeners use a combination of acoustic invariant and contextual cues, Jiang, Chen, and Alwan (2006) found that listener's perception of voicing in CVs depended on the onset and duration of the first formant transition.

Research findings that chinchillas (Kuhl & Miller, 1975; Kuhl & Miller, 1978) and quails (Kluender et al., 1987; Lotto, Kluender, & Holt, 1997) can perceive speech contradicts the assumption by the Motor Theory of speech perception that humans possess a specialized speech decoder. It appears that animals, without the mechanisms to produce speech, were able to perceive speech. Both the Motor Theory and the Direct Realist theory of perception were refuted further by findings showing similarities between perception of speech and non-speech stimuli by human listeners. Since non-speech stimuli such as pure tones and noise bursts are not made by articulatory gestures, a listener's ability to be influenced by and perceive non-speech stimuli as speech supports the notion that perception results from the recovery of acoustic information not gestures (Holt & Kluender, 2000)

How does the Auditory Approach to speech perception explain listeners' perceptual resistance to the acoustical effects of coarticulation? In the previous example of /arda/ and /alga/,

the spectro-temporal cues for the consonants /d/ and /g/ are very similar due to the context of /r/ and /l/ (Figure 2-5). Yet listeners are able to resist the assimilative context to perceive /d/ and /g/. As in the presence of noise when the invariant acoustic cues are ambiguous, listeners must rely on context-conditioned cues for phonemic identification. The Auditory Approach (Figure 2-9) points to spectral contrast as a cue within context to aid listeners in neutralizing the assimilated effects of coarticulated phonemes (Lotto et al., 1997; Lotto & Kluender, 1998; Holt et al., 2000; Holt, Lotto, & Kluender, 2001; Diehl et al., 2004). Acoustic spectral contrast occurs when there are frequency differences in neighboring phonemes. Auditory perceptual contrast mechanisms may exaggerate these differences so that accurate phonemic recognition is maintained across context due to coarticulation (Holt & Kluender, 2000).

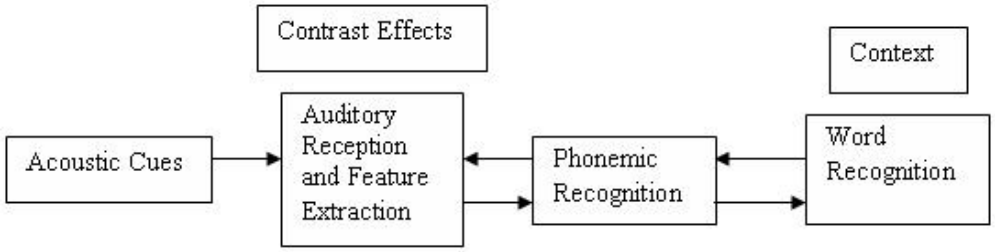


Figure 2-9: A simplified model of a General Approach to speech perception. Boxes above the stages in the model indicate cues that can shift the perception of speech.

Spectral contrast can explain the results of Mann (1980) in which the perception of /da/ increases (/ga/ responses decrease) in the context of the preceding /ar/. Figure 2-5c and d show the schematic graphs generated by Lotto and Kluender (1998) of the first four formant transitions for the /arda/ and /arga/ stimuli that were used in the Mann (1980) study. In the syllable /arda/, the spectral disparity between the offset of the /r/ and the onset of the /d/ F3 transition is greater than that in the syllable /arga/. The spectral contrast between the neighboring phonemes in /alga/

is shown in Figure 2-5b while Figure 2-5a shows the F3 spectral continuity of /alda/. This general auditory mechanism of spectral contrast may be a valuable tool in predicting the phonemic pattern of coarticulated speech despite its assimilative acoustic effects. Evidence for the role of spectral contrast has been observed for phonemic syllables (Mann & Repp, 1981a,b; Repp & Mann, 1980, 1981), lexemes (Elman & McClelland, 1988), and non-speech stimuli (Stephens & Holt, 2003; Wade & Holt, 2005; Holt, 2005). Because perceptual accommodation for coarticulation is evident at the subphonemic, phonemic, and lexical stages of auditory perception, this suggests that speech perception is an interactive process with a bi-directional flow of information between higher level cognition and the perception of the acoustical properties of the sound (McCelland, Mirman, & Holt, 2006).

2.1.10 **Neurophysiology of Speech Perception**

Regardless of the model for the perception of speech, the phonemic acoustic pattern has to be received by the ear and converted into the neural code that represents, depending on the model, the intended gesture, the actual gesture, or the acoustic properties corresponding to the target phoneme. The cochlea is a frequency analyzer consisting of rows of sensory hair cells along its basilar membrane. These hair cells serve as transducers converting mechanical energy to electrical impulses at the synapse of the attached auditory nerve fibers. Each hair cell and corresponding nerve fiber is a band pass filter with a maximum sensitivity to a specific or characteristic frequency (CF). The cochlea is tonotopically organized in that the CF of nerve fibers is determined by the location of the hair cell on the basilar membrane. The spectral properties of the speech signal are encoded by the place of neural activation along the cochlea

and this place code is preserved along the neural pathway to higher auditory centers (Hackney, 2006). The rate of neural discharge indicates acoustic energy, and the energy onset/offset.

Delgutte and Kiang (1984a,b,c,d) published a series of papers on their work of quantifying how speech is encoded in the auditory nerve. In these experiments, the authors recorded the responses of single nerve fibers with different CF's to a variety of speech stimuli. The auditory nerve responses to vowel and fricative stimuli can be described either in terms of average rate of discharge or by the fine time patterns of spike (neuronal discharge) activity. Peaks in the discharge rate occur in the nerve fibers whose CF corresponds to each of the vowel's formants, while the fine time pattern of the spike activity reflects the periodicity of the signal (Delgutte & Kiang, 1984a). The place of maximal rate of neuronal discharge for voiceless fricatives corresponded to the high frequency region of frication noise. The temporal pattern of neuronal discharge of fibers with lower frequency CF's does not display any form of modulation, indicating that the signal is aperiodic (Delgutte & Kiang, 1984b).

For consonants, the auditory nerve fiber has to respond to the rapid amplitude and spectral changes in the spectrum. There is evidence that nerve fibers use short-term adaptation as a mechanism to enhance their sensitivity to the dynamic characteristics of speech (Smith, 1979; Delgutte & Kiang, 1984c). Neural short-term adaptation occurs after the onset of an acoustic stimulus causes a rapid increase in the spike rate of neuronal discharge. Immediately after this sharp peak in activity, the nerve fibers gradually adapt to the signal by decreasing the rate of fire over time. This adaptation allows the auditory nerve fiber to be able to increase discharge rate when acoustic changes occur in the stimulus. Delgutte and Kiang (1984c) measured the effect of preceding context on the neural response to the consonant vowel transition corresponding to /da/. The 10 acoustic stimuli in their experiment consisted of /da/, and nine stimuli with the formant

transitions for /da/ preceded by the context of /a/, /i/, /u/, /n/, /s/, /sh/, /st/, and /d/ with the upper formants (F4 & F5) enhanced. The authors measured the response of an anesthetized cat's auditory nerve fibers to each of these stimuli. The authors not only found evidence of neural short-term adaptation, but also found that discharge rates during the transitions decreased for those nerve fibers with CF's corresponding to a frequency region that was present in the preceding context. When the /da/ transitions followed the phoneme /sh/, the neural response for the transitory period was reduced in the high CF-fibers. When the /da/ transition was preceded by a predominately low frequency energy /n/, the neural response to the transition was reduced in the low CF-fibers.

It is reasonable to posit that neural adaptation may explain the mechanisms of spectral contrast. While the frequency regions that are shared between the coarticulated phonemes would be suppressed, thus enhancing the regions in which there is little or no spectral overlap (Holt & Kluender, 2000). In the example of /arga/, the lower frequency F3 of the /ar/ would theoretically cause short term adaptation on the third formant of the /ga/ (Figure 2-5d). The F3 of /da/ is a higher frequency than the F3 of the /ar/, so that different nerve fiber bundles are firing rather than the already adapted fibers (Figure 2-5c). This enhancement would cause listeners to favor the perception of /da/ in the context of /ar/.

Although, neural adaptation is an attractive source of the contrast effect, there is evidence that contrast effects may arise from a more central auditory mechanism. Specifically, neural adaptation is a monaural mechanism yet studies have shown that contrast effects can result from dichotic presentations with the context cue delivered to one ear and the target stimuli delivered to the opposite ear (Holt & Lotto, 2002). Also, neural adaptation dissipates after 50-100 ms (Delgutte, 1980), while contrast effects linger after 400 ms (Holt & Lotto, 2002). Animal studies

further ruled out the role of the peripheral mechanism at the level of the auditory nerve and cochlear nucleus in spectral contrast effects (Holt, Ventura, Rhode, Behesta, & Rinaldo, 2000).

The acoustic dimensions of frequency, amplitude, and time are encoded in the auditory nerve fibers. This neural pattern of the speech spectrum is transmitted through the brainstem to the auditory cortex where it is cognitively processed and translated into lexemes. There is evidence that the tonotopicity from the cochlea to the auditory nerve fibers is somewhat maintained in the auditory cortex (Cheung, Bedenbaugh, Nagarajan, & Schreiner, 2001). The circuitry of the auditory cortex is extremely complex with ascending, descending and lateral connections. The auditory cortical map and its functional relationships are not fully understood and is a topic of interest and debate among researchers in the fields of neuroanatomy and neurophysiology (Budinger & Heil, 2006).

2.2 SPEECH PERCEPTION BY LISTENERS WITH MILD-MODERATE SENSORINEURAL HEARING LOSS

Sensorineural hearing loss causes a loss of sensitivity to sound. Elevated sensitivity thresholds increase the amount of difficulty in understanding speech. For listeners with mild-moderate hearing impairment, this loss of audibility accounts for most of the detriment in speech recognition performance (Dirks, Bell, Rossman, & Kincaid, 1986; Humes, Dirks, Bell, Ahlstrom, & Kincaid, 1986; Zurek & Delhorne, 1987; Dubno et al., 1989; Ching, Dillon, & Byrne, 1998). However, several studies have reported that audibility is not the only contributor for degradation of speech understanding in listeners with a more severe hearing impairment (Dubno et al., 1989; Rankovic, 1991; Ching et al., 1998; Hogan & Turner, 1998; Turner & Cummings, 1999). The

poorer than predicted performance on speech recognition tasks have been attributed to the reduction of frequency resolution in the damaged cochlea. Hearing-impaired listeners generally have auditory filters that are more broadly tuned than normal-hearing listeners (Tyler, Wood, & Fernandes, 1982; Glasberg & Moore, 1986; Moore & Glasberg, 1986). However, listeners with a mild-moderate hearing impairment may perform as well as normal-hearing listeners on spectral and temporal resolution tests (Tyler, Hall, Galsberg, & Patterson, 1984; Glasberg & Moore, 1986; Thibodeau & Van Tasell, 1987; Summers & Leek, 1995). Because mild-moderate hearing-impaired listeners' difficulty understanding speech in quiet environments is more related to lack of audibility than poor spectral and temporal resolution, it can be hypothesized that these listeners can perceive the spectral, amplitude, and temporal pattern/cues of speech similarly to normal-hearing listeners once audibility has been achieved.

2.2.1 **Spectral Cues**

Spectral cues aid normal-hearing listeners in the perception of vowels. Nabelek and her colleagues (1993) conducted an experiment to test whether normal-hearing and mild sloping to moderately-severe hearing-impaired listeners differed in their perception of an /I – ε/ vowel continuum once audibility was accounted for. Both groups of listeners listened to several different vowel continuums with either a steady-state spectral cue or a formant transitional spectro-temporal cue in three listening environments: quiet, noise, and reverberation. The listeners were asked to label what they heard as being an /I/ or an /ε/. The location of the boundary and slope of the identification functions for each continuum in each of the background environments were compared between the groups. No significant differences were found between the groups regardless of the acoustic cues (steady-state or transitions) or the background

noise (quiet, noise, or reverberation). The results indicate that hearing-impaired listeners can use spectral cues to identify the vowels /I – ɛ/.

Spectral information is critical to the perception of fricatives. Hearing-impaired listeners tend to have a lot of difficulty in the perception of fricatives due to the combination of the weak intensities of these phonemes and the presence of high frequency hearing loss. Listeners with hearing loss can use the spectral cue once it is made accessible to them through amplification. Hearing-impaired listeners listening to the /s/, /f/, and /θ/ spoken in a CV context by male, female, and child speakers, can correctly identify the fricative phoneme more than 80 % of the time as long as audibility is achieved through 9 kHz (Stelmachowicz, Pittman, Hoover, & Lewis, 2001).

2.2.2 **Intensity Cues**

Intensity cues, in addition to spectral cues, aid in the perception of manner of phonemes such as voiced consonants and nasality. Miller and Nicely (1955) analyzed the error patterns of normal-hearing listeners' perception of sixteen consonants. The listeners listened and categorized these consonants under listening conditions comprised of either quiet or noisy environments, and low pass filtered speech. The authors found that the listeners ability to identify the voicing or the nasality of articulation was resistant to the deleterious effects of background noise, whereas place of articulation is severely affected by environmental manipulations. Studies analyzing the consonant confusion patterns of mild-moderately hearing-impaired listeners demonstrate results in which the classification of nasality and voicing is least affected by the sensorineural hearing loss while the number of errors for place of articulation is elevated (Dubno, Dirks, & Langhofer, 1982; Turner & Brus, 2001).

2.2.3 **Temporal Cues**

For normal-hearing listeners, temporal cues aid listeners in the perceptual distinction between fricatives and affricates (Kluender & Walsh, 1992). Hedrick (1997) found that when the duration of the noise in the voiceless fricative /ʃ/ is shortened from 140 ms to 50 ms, both normal hearing and moderately hearing-impaired listeners tend to label the consonant as /tʃ/. Another study manipulated durational cues by inserting a silent gap of varying durations between the phoneme /s/ and the onset of the following vocalic phoneme of the word "say" to produce a perceptual continuum ranging from "say" to "stay" (Nelson, Nittrouer, & Norton, 1995). This study found that as long as the spectral-temporal cues (formant transitions) were not ambiguous, the boundary between the perception of "say" and "stay" was at the gap duration of 18 ms for normal-hearing and hearing-impaired listeners alike. In addition, psychoacoustic measures of gap detection of noise bursts were analyzed between the groups of listeners. When intensity level was equated in sensation levels between normal-hearing and mild-moderate hearing-impaired listeners, no significant difference was found between the two groups of listeners. It appears that listeners with sensorineural hearing loss restricted to mild-moderate severity can detect and use the temporal cues related to speech.

2.2.4 **Spectro-Intensity Cues**

According to Ohde and Stevens (1983) spectro-intensity cues serve to help listeners identify voiceless stop consonants. Normal-hearing listeners exhibit a labial /p/ bias when the amplitude of the consonant's burst is lower than the following vowel's fourth and fifth formant. Several authors have tested to see whether burst amplitude manipulations affect hearing-impaired

listeners CV identification scores (Gordon-Salant, 1987; Montgomery & Edge, 1988; Kennedy, Levitt, Neuman, & Weiss, 1998; Hedrick & Younger, 2001). These studies found that by enhancing the amplitude of the consonant so that the consonant to vowel ratio (CVR) is increased, hearing-impaired listeners' performance on CV syllable identification tasks improved by as much as 45.8 percentage points for specific phonemes (Kennedy et al., 1998).

A confounding factor in these results is that increasing the amplitude of the consonant increases the audibility of the consonants' spectral properties for hearing-impaired listeners. Sammeth, Dorman, and Stearns (1999) conducted a study that manipulated CVR by holding the consonant audibility constant while reducing the amplitude of the following vowel by 6 and 12 dB. They concluded that CVR enhancement by vowel reduction did not improve recognition performance of voiceless stop for either normal-hearing or mild-moderate hearing-impaired listeners. This study did find that hearing-impaired listeners, like the normal-hearing listeners, exhibit a labial /p/ bias when the spectral burst was lower in amplitude. Mild-moderate hearing-impaired listeners can use intensity cues that vary as a function of frequency in speech perception.

2.2.5 **Tempo-Intensity Cues**

The cues within the temporal waveform of speech are very important to hearing-impaired listeners. Studies using CV stimuli constructed so that all spectral information is reduced or removed and the overall amplitude envelope is preserved have shown that hearing-impaired listeners perform just as well as normal-hearing listeners in consonant identification tasks (Turner, Souza, Forget, 1995; Turner, Chi, & Flock, 1999; Lorenzi, Gilbert, Carn, Garnier, & Moore, 2006). Turner, Chi, and Flock (1999) further studied the role of temporal envelope cues

by dividing VCV stimuli into two or more frequency bands, then creating two amplitude envelopes for each stimulus. When these bands of temporal patterns were given to normal-hearing and hearing-impaired listeners, hearing-impaired listeners' consonant recognition was worse than that of normal-hearing listeners. Although hearing-impaired listeners can use tempo-intensity cues, it appears that when given several temporal patterns corresponding to different frequency regions of the stimulus, they are not able to integrate this information (Healy & Bacon, 2002). This spectro-tempo-intensity cue has been labeled as the temporal fine structure (TFS) of speech. It appears that hearing-impaired listeners' inability to use the TFS as a cue affects the listener's ability to perceive speech in background noise (Qin & Oxenham, 2003; Lorenzi et al., 2006)

2.2.6 **Spectro-Temporal Cues**

Spectro-temporal cues consist of formant transitions, transition durations, and voicing onset cues. For normal-hearing listeners, formant transitions are critical to their perception of the place of articulation. Listeners with sensorineural hearing loss have difficulty identifying the place of articulation for stop consonants (Owens, Benedict, & Schubert, 1972; Walden, Schwartz, Montgomery, & Prosek, 1981; Turner, Fabry, Barrett, & Horwitz, 1992; Turner & Brus, 2001). Lindholm, Dorman, Taylor, and Hannley (1988) examined the perceptual importance of three acoustic cues to the perception of voiced stop consonants by normal-hearing and mild-moderate hearing-impaired listeners. The stimuli consisted of /bæ/, /dæ/, and /gæ/ with the appropriate formant transitions, burst spectral templates, and rate of frequency change. The authors spliced the formant transitions and burst spectral properties out of each CV. They also calculated the rate of frequency change for each CV and found that for /bæ/ and /dæ/, the transition occurred over 5

ms while /gæ/ occurred over 20 ms. Then, the authors combined the cues in a matrix format to produce 18 combination stimuli. For example one stimulus consisted of the /bæ/ transition, paired with a /d/ burst spectrum, and a transition rate of 20 ms. These 18 combination stimuli were then given to the listeners in an identification task. The results showed that normal-hearing listeners mostly rely on the formant transition cue to identify the target phoneme regardless of the conflicting cues. Hearing-impaired listeners relied on formant transition cues less than the normal-hearing listeners. The impaired listeners' identification performance was more influenced by the spectral shape and temporal properties of the signal. It appears that the impaired auditory system has some degree of difficulty using the rapidly changing formant transitions as a cue to speech perception (Zeng & Turner, 1990; Turner, Smith, Aldridge, & Stewart, 1997).

Although hearing-impaired listeners demonstrate difficulty in using the transitions as cues, such is not the case for voice onset timing cues. These timing onset cues help listeners differentiate between voiced and voiceless cognate pairs. Johnson, Whaley, and Dorman (1984) found that listeners with mild-moderate hearing impairment did not significantly differ from listeners with normal sensitivity in the perception of voice onset time (VOT). Similarly, hearing impairment does not affect the perception of envelope onset asynchrony (EOA). Ortmann, Palmer, and Pratt (2010) examined the influence of EOA manipulations on the perception of voicing in a group of mild-moderate hearing-impaired listeners in addition to the previously mentioned group of normal-hearing listeners. Using the same stimuli and procedure as with the normal-hearing group, the authors found that the perception of voicing in stop consonants is influenced by the degree of temporal asynchrony. Figure 2-10 displays the hearing-impaired groups' average labeling and discrimination data for /pa/, /ta/, /ka/. The hearing-impaired

listeners did not significantly differ from the normal-hearing listeners (Figure 2-4) in their use of EOA as a cue to voicing distinction.

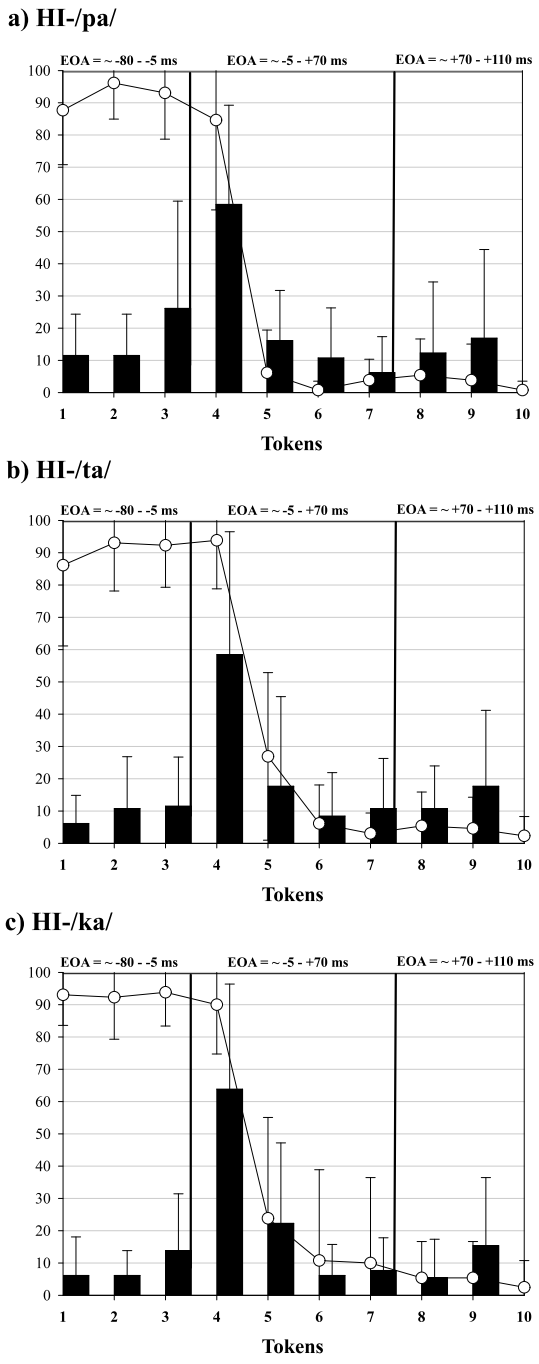


Figure 2-10 (a-c): Hearing-impaired listeners' identification and discrimination data for the EOA continuums, a) /pa/ b) /ta/ and c) /ka/. For each of the graphs, the x-axis displays the 10 tokens representing the shift in EOA from a more negative value to a more positive value. The space in between each token value represents the adjacent token pairs (i.e., token 1 paired with token 2, token 2 paired with token 3, and so on). The values along the y-axis are in percent. The line graph

represents the data from the labeling task, so higher on the y-axis means that the listeners' perception is voiced, while lower values represent a more voiceless percept. The bar graph displays the discrimination data, so higher y-axis values mean that a greater difference between the token pair was detected.

2.2.7 Hearing-Impaired Perceptual Performance and Models of Speech Perception

Listeners with mild-moderate hearing impairment appear to use cues in each of the acoustic dimensions to a certain extent once the speech signal is audible. Impaired auditory systems tend to rely more on spectral shape and temporal cues, than rapidly changing formant transitions. Articulation-based models of speech perception, such as the Motor Theory and Direct Realist Theory account for poorer speech perception in hearing-impaired listeners by stating that cochlear damage generates ambiguous gestures (intended or actual). Auditory-based theories state that the damaged cochlea distorts the acoustic properties of speech and causes ambiguity in the auditory neural mapping of features to phonemes/lexemes. Each of the models of speech perception can provide an explanation for hearing-impaired listeners' performance.

Little is known about the strategies hearing-impaired listeners use to differentiate acoustic similarities brought on by coarticulation. Although hearing loss interferes with listeners' access to the spectral and spectro-temporal properties of speech, mild-moderate hearing-impaired listeners can use them to a certain extent (Lindholm et al., 1988). As coarticulation is rampant in conversational speech, it is reasonable to assume that listeners with mild-moderate hearing loss consistently use coarticulation to recover the intended phoneme. Auditory-based theories of speech perception promote the role of contrast effects to predict the phonemic pattern of coarticulated speech. It is not known whether hearing-impaired listeners are influenced by spectral contrast. Hearing-impaired listeners use or lack of use of spectral contrast could potentially shed light on the neural network mechanisms (peripheral v central) involved in

spectral contrast and could potentially strengthen the argument for one or more models of speech perception.

2.3 ACOUSTICS OF CONVERSATIONAL SPEECH

The majority of speech perception studies use stimuli such as synthesized syllabic tokens, naturally produced syllabic tokens, clearly spoken words, or rehearsed and read sentences or passages. While these tokens allow for the experimenter's control of the acoustic characteristics of the stimuli, they are not wholly representative of the speech that listeners are exposed to daily. Daily communication consists of speech spoken in a conversational manner. There are intelligibility differences between conversational speech and speech spoken in a clear or distinctive manner for both normal and hearing-impaired listeners (Payton, Uchanski, & Braida, 1994; Schum, 1996; Uchanski et. al., 1996; Krause & Braida, 2002; Liu, Del Rio, Bradlow, & Zeng, 2004; Liu & Zeng, 2006). Picheny and his colleagues (1985) found that there is a decrease in the intelligibility of conversationally produced nonsense sentences compared to similar sentences spoken more clearly. Also, the acoustic information within conversationally produced speech is different from clearly produced speech along the static dimensions of frequency, intensity, and time and the dynamic dimensions of spectro-intensity, intensity-temporal, and spectro-temporal variations (Picheny, Durlach, & Braida, 1986; Krause & Braida, 2004).

2.3.1 Differences in Static Cues Between Conversational and Clear Speech

2.3.1.1 Spectral and Intensity Cues

Static acoustic cues for speech perception result from a single acoustic property such as frequency, intensity, or time information. Spectral differences between conversational speech and clear speech are lower fundamental formant frequency values and elimination of the spectrally rich burst information in consonant plosives in conversational speech (Picheny et. al., 1986; Krause & Braida, 2004). In their analysis of conversational speech, Picheny et. al. (1986) found that 60% of plosive bursts in the word final position were eliminated. Krause & Braida (2004) performed a similar acoustical analysis of clear and conversational speech, and reported that conversational speech has less relative energy above 1 kHz. Although the lack of high frequency spectral information in conversational speech contributes to poorer speech perception, it does not account for the total decreased intelligibility of conversationally spoken speech over clearly produced speech.

2.3.1.2 Temporal Cues

Temporally, conversational speech is drastically different from clear speech. Conversational speech ranges between 160 to 200 words per minute or 3-4 syllables per second, which is twice as fast as clearly produced speech (Picheny et. al., 1986). Not only are there fewer pauses during conversational speech, but also the overall rate of articulation increases (Picheny et. al., 1986, Picheny et al., 1989; Byrd & Tan, 1996; Uchanski et al., 1996). Picheny, Durlach, and Braida (1989) continued their series of studies examining the intelligibility differences between clear and conversational speech by focusing on the role of speaking rate. The authors artificially slowed down the rate of a spoken sentence until its overall duration was equal to the duration of

the same sentence spoken clearly. They also temporally compressed clearly produced speech so that its overall duration was the same as conversationally produced speech. If speaking rate is a determining factor in the intelligibility advantage of clear speech over conversational speech, then slowing down conversational speech should increase intelligibility and increasing the rate of clear speech should decrease intelligibility. The results of this study did show that shortening the duration of clearly produced sentences decreased the intelligibility of the sentences. However, temporally expanding the duration of conversational speech so that it was overall temporally equal to clear speech did not improve performance. In fact, the intelligibility of artificially slowed conversational speech was worse than unprocessed conversational speech.

Uchanski et. al. (1996) proposed that perhaps this failure to find an intelligibility advantage by slowing down the rate of conversational speech was due to the uniform expansion algorithm used to adjust the speaking rate. In their analysis of durational differences between conversational and clear speech, they found differences in *phonemic segmental* durations between the two speaking styles. They reported that while short vowels and voiced plosives increase 29% and 43% in duration for clear speech over conversational speech, clearly produced unvoiced fricatives and semivowels lengthen by 91% and 103% in comparison to conversational speech. Uchanski and colleagues (1996) used a non-uniform time-scaling technique to artificially slow down speech so that the phonemic segmental durations of conversationally produced sentences were equal to that of clearly produced sentences. They also used this same time-scaling algorithm to speed up clear speech so that it was equal in segmental duration to conversational speech. Their results were similar to Picheny et al. (1989) in that artificially slowing down conversational speech resulted in poorer speech intelligibility for both normal hearing listeners in background noise and hearing-impaired listeners. Speeding up clear speech

also resulted in poorer intelligibility scores than those of non-processed conversational speech for both groups of listeners.

Lui and Zeng (2006) inserted small gaps between the phonemic segments of conversationally spoken speech so that the durations of the sentences were equal to that of clearly spoken sentences. The intelligibility of the gap-inserted conversational sentences by normal-hearing listeners in background noise did increase relative to the unaltered conversational sentences. However, the intelligibility of their gap-inserted conversational speech was significantly poorer than clearly produced sentences. The authors concluded that the uniform and non-uniform signal processing used in the previous studies by Picheny et. al. (1989) and Uchanski et. al. (1996) to alter either the overall or segmental durations of speech introduced some extraneous distortions that resulted in poorer than predicted results (Lui & Zeng, 2006). They also concluded that the insertion of gaps increased the amplitude modulation and allowed more time for the efficient phonemic processing by the listeners (Fu, 2002)

Krause and Braida (2002) further examined the role of speaking rate by using naturally produced speaking rate alterations of both clear and conversational speech. The authors trained five speakers with significant public speaking experience to produce clearly articulated speech at their internally defined slow, normal (conversational rate), and quick speaking rates. The speakers also were instructed to read aloud nonsense sentences in a "conversational manner" at each of the three rates. The speakers were given intensive training regarding the differences between the two speaking styles and speaking rates. The intelligibility scores from two listeners with normal hearing in background noise indicated that there is a benefit of clear speech as the speaker's rate increases. For each of the five talkers, producing speech in a clear manner yielded higher intelligibility scores than the productions of conversational style speech, even when the

speaking rate between the two speaking styles was roughly equal to 200 words per minute. Although the rate of articulation is a primary difference between clear and conversational speech, the secondary fine acoustic differences between clear and conversational speech play larger roles in intelligibility.

2.3.2 **Differences in Dynamic Cues Between Conversational and Clear Speech**

2.3.2.1 Spectro-Intensity Cues

The dynamic cues of speech perception are those described as bi-dimensional such as spectro-intensity, tempo-intensity, and spectro-temporal cues. The faster speaking rate of conversational speech causes the acoustical properties of the bi-dimensional cues to differ from those of clear speech. One of the secondary effects of a faster speaking rate is an increase in phonemic coarticulation. Byrd and Tan (1996) used electropalatography to measure speaker's tongue-palatal contact when speaking the sentence, "Say baC_1 C_2ab again" (C_1 and C_2 are two different consonants) at a normal and a fast rate. They found that as speaking rate increases, the duration of tongue-palatal contact decreases for the consonants and that coarticulation occurs. Coarticulation was documented by the compromised tongue-palatal contact location between the two articulated consonants. The articulation of the first consonant /d/ in the utterance /ba*d g*ab/ caused the place of constriction for the following /g/ to be more frontal and less intense at conversational speaking rate than when /ba*d g*ab/ was spoken at a slower speaking rate. There was also evidence that the second phoneme /g/ influences the place of articulation of the preceding phoneme /d/. This alteration of tongue-palatal constrictions in conversational speech results in spectral and spectro-intensity changes of plosive bursts. The weakening or deletion (Picheny et. al., 1986; Krause & Braida, 2004) of the plosive burst intensity and the shift in the

plosive's center frequency disrupt the acoustic cue patterns found in speech (Blumstein & Stevens, 1979; Blumstein & Stevens, 1980; Ohde & Stevens, 1983) and could lead to phonemic ambiguity in conversational speech.

2.3.2.2 Tempo-Intensity Cues

The increase of articulation rate in conversational speech alters the amplitude envelope of the speech signal from that of slower, clearly spoken speech. Amplitude fluctuations over time are cues for stop consonant identification due to the brief periods of silence prior to the release of the burst (van Tassel et. al., 1987). Clearly spoken speech has greater temporal amplitude modulation than conversational speech (Krause & Braida, 2002; Liu et. al., 2004; Liu & Zeng, 2006). Compared to clearly spoken speech, the boundaries between syllables are not as distinct in conversationally spoken sentences. In conversational speech the plosive bursts are either omitted (Picheny et. al., 1986, Krause & Braida, 2004) or slurred together (Byrd & Tan, 1996). The faster articulation rate of conversational speech leads to fewer and smaller gaps between syllables and words, which translates to a shallower depth for modulation frequencies below 3-4 Hz than clearly produced speech (Krause & Braida, 2004). Conversational speech does not obliterate the tempo-intensity cue for phonemic identification, but it could make this cue slightly less distinctive. Although listeners with normal hearing rely on other acoustic cues to correctly identify consonants when these tempo-intensity cues are compromised (Christensen & Humes, 1997), listeners with hearing impairment rely heavily on these tempo-intensity cues (Lindbolm et. al., 1988; Summers & Leek, 1992).

2.3.2.3 Spectro-Temporal Cues

Spectro-temporal cues occur when frequency varies as a function of time. Voice onset time (VOT) and formant transitions are two examples of spectro-temporal cues. The dynamic formant movement has been shown to influence listeners' perception of consonants (Delattre et. al., 1955) and vowels (Hillenbrand et al., 1995). In conversational speech the vowel space, which is measured by the frequency area between the first and second formants of uttered vowels, is reduced when compared to clearly produced speech (Picheny et. al., 1986; Moon & Lindblom, 1994; Ferguson & Kewley-Port, 2002). Ferguson and Kewley-Port (2002) analyzed the acoustic properties of 10 vowels uttered by a single speaker in both a conversational and clearly spoken manner. They found that the magnitude of the dynamic formant (F1 and F2) movement (amount of formant frequency change over time) was significantly smaller in conversationally produced speech than clearly produced speech. The combination of the shorter duration of conversational speech and the smaller vowel space of the speaker's formant variability alters the duration and slope of the formant transitions. Normal-hearing listeners give greater perceptual weight to rapid formant transitions than the other acoustic cues in identifying consonants (Lindholm et. al., 1988; Hedrick & Jesteadt, 1996; Hedrick & Younger, 2001, 2007). Hearing-impaired listeners can perceive and use the rapid formant transitions, but their contributions to speech intelligibility are smaller than that for normal-hearing listeners. Hedrick and Younger (2007) demonstrated that hearing-impaired listeners performed similarly to normal-hearing listeners on the use of formant transitions to identify /p/ in a quiet listening environment. Once the formant transition was degraded by background noise or reverberation, the hearing-impaired listeners gave formant transitions less perceptual weight while normal-hearing listeners continued to use the transition information to identify phonemes. The shortening of formant transitions during conversational

speech could degrade this frequency-temporal cue for hearing-impaired listeners so that they may no longer rely on it for phonemic perception.

In their acoustical analysis of clear and conversational speech, Picheny et. al. (1986) found that VOT was shorter for speech spoken in a conversational manner. For conversational speech, the VOT in stressed word-initial voiceless plosives was an average of about 80 ms as opposed to an average of 160 ms for clearly produced speech. Previous studies have reported shorter VOTs for voiceless plosives in speech spoken at a conversational rate (Klatt, 1975). Klatt (1975) reported an average VOT value of 47 ms for /p/ in the word-initial position (as in "pat"), and 12 ms for /p/ in the word-initial consonantal cluster of /sp/ (as in "spoon"). Krause and Braida (2004) also analyzed VOT differences in their corpus of conversationally and clearly spoken sentences by two speakers with public speaking experiences. The two speakers were chosen because they were able to produce "clear" speech at a conversational speaking rate of 200 words per minute. These were the same speakers that were used in their previous study which demonstrated that there is a 14 point intelligibility advantage in speech spoken clearly even when it is spoken at the same rate as speech produced in a conversational manner (Krause & Braida, 2002). The authors found that of the two speakers analyzed, only one had shortened his/her word initial voiceless stop consonants' VOT in conversational speech (Krause & Braida, 2004). Since VOT tends to be shortened in conversationally spoken speech, it is likely that envelope onset asynchrony (EOA) also is affected in that the high frequency band containing the burst and aspiration are closer to the onset of the voicing of the vowel for voiceless plosives.

In conclusion, conversationally spoken speech varies greatly from clearly produced speech. Although the increase of articulation rate is the most apparent difference between clear and conversational speech, it does not account for total reduction of speech intelligibility

(Picheny et. al., 1989; Uchanski et. al., 1996; Krause & Braida, 2002, 2004; Lui & Zeng, 2006). Conversational speech contains the very same static and dynamic acoustic cues as clearly produced speech, but the distinctiveness of these cues is degraded. When assessing either a listener's real-world speech intelligibility or the impact a signal processing scheme has on speech intelligibility, it is important to choose speech stimuli that represent the acoustic properties of conversational speech.

2.4 DIGITAL SIGNAL PROCESSING AND THE SPEECH SPECTRUM

Listeners with mild-moderate hearing impairment use the spectral, intensity, and temporal patterns in speech as cues for speech recognition. It is important to consider how amplification devices and signal processing schemes can manipulate and change these acoustic properties. There is a volume of literature examining the effects of various algorithms such as wide dynamic range compression, noise reduction algorithms, and adaptive directional microphones on speech perception in hearing-impaired listeners (Ricketts & Henry, 2002; Souza, 2002; Chung, 2004; Souza, Jenstad, & Folino, 2005; Jenstad & Souza, 2005; Palmer, Bentler, & Mueller, 2006; Bentler & Chiou, 2006). These algorithms can affect the amplitude envelope cues and spectro-intensity cues of speech, and can be deleterious to individuals with more severe hearing impairment. However, underneath these algorithms lies a source of signal distortion that is not as well studied. The digital signal processing (DSP) chip, which is inherent to every hearing aid sold in the United States today, introduces a delay to the speech signal that could possibly disrupt some of the spectro-temporal cues found in speech.

The delay at the output of digital hearing aids is a result of the combination of converter delay and processing delay. Converter delay is the delay caused by the analog/digital converter, which delays the signal approximately 0.7 ms across the spectrum of the incoming signal. Processing delay is defined as the resultant delay arising from the DSP algorithm that divides the signal into different frequency bands. Processing delay can be defined as either spectrally synchronous or spectrally asynchronous depending on the algorithm employed. Kates (2005) described three basic types of digital processing. First, there are DSP circuits that employ a time domain filter bank algorithm to divide the incoming signal. Time domain filtering introduces a spectrally asynchronous delay that delays the low frequency output relative to high frequency output. A second type of signal processing uses frequency domain filtering or fast Fourier transforms (FFT) to divide the incoming signal. FFT technique buffers or stores the incoming signal for analysis. The resultant output of a DSP circuit employing FFT has a spectrally synchronous delay, meaning all frequencies are delayed a value determined by the size of the input buffer. A third DSP employs digital frequency warping. Warping combines the use of overlapping all-pass filters and FFT. Digital frequency warping introduces a spectrally asynchronous delay that delays the low frequency information relative to the high frequency information.

Figure 2-11(a-d) displays the delay values of four brands of digital hearing aids fit with a closed-earmold: Siemens Triano, Widex Diva, Phonak Claro, and Resound Canta. Each hearing aid was programmed to the NAL-NL1 target for a 50 dB flat hearing loss. Two hearing aids of each manufacturer were used for test/retest reliability purposes. These measurements were taken from the ear canal of the Knowles Electronic Mannequin for Acoustic Research (KEMAR) who was fit with each hearing aid. The recordings for these measurements were obtained using an

Etymotic Research ER-11 microphone and a Zwislocki coupler inside of KEMAR. In an anechoic chamber, KEMAR was positioned in front of a loudspeaker at 0° azimuth. The delay of the amplification device was measured by subtracting the arrival time at the microphone of an impulse sound generated from a speaker without a hearing aid present from the arrival time of the impulse sound at the microphone with a functional hearing aid present. The Siemens Triano and Widex Diva hearing aids, which employ a time-domain filter bank algorithm, show signs of spectrally asynchronous delays with the low frequencies delayed relative to the high frequencies, while Phonak Claro and Resound Canta hearing aids with frequency domain filtering algorthms displayed a constant delay value across all frequencies. These delay values agree with those reported by Dillon and his colleagues in 2003.

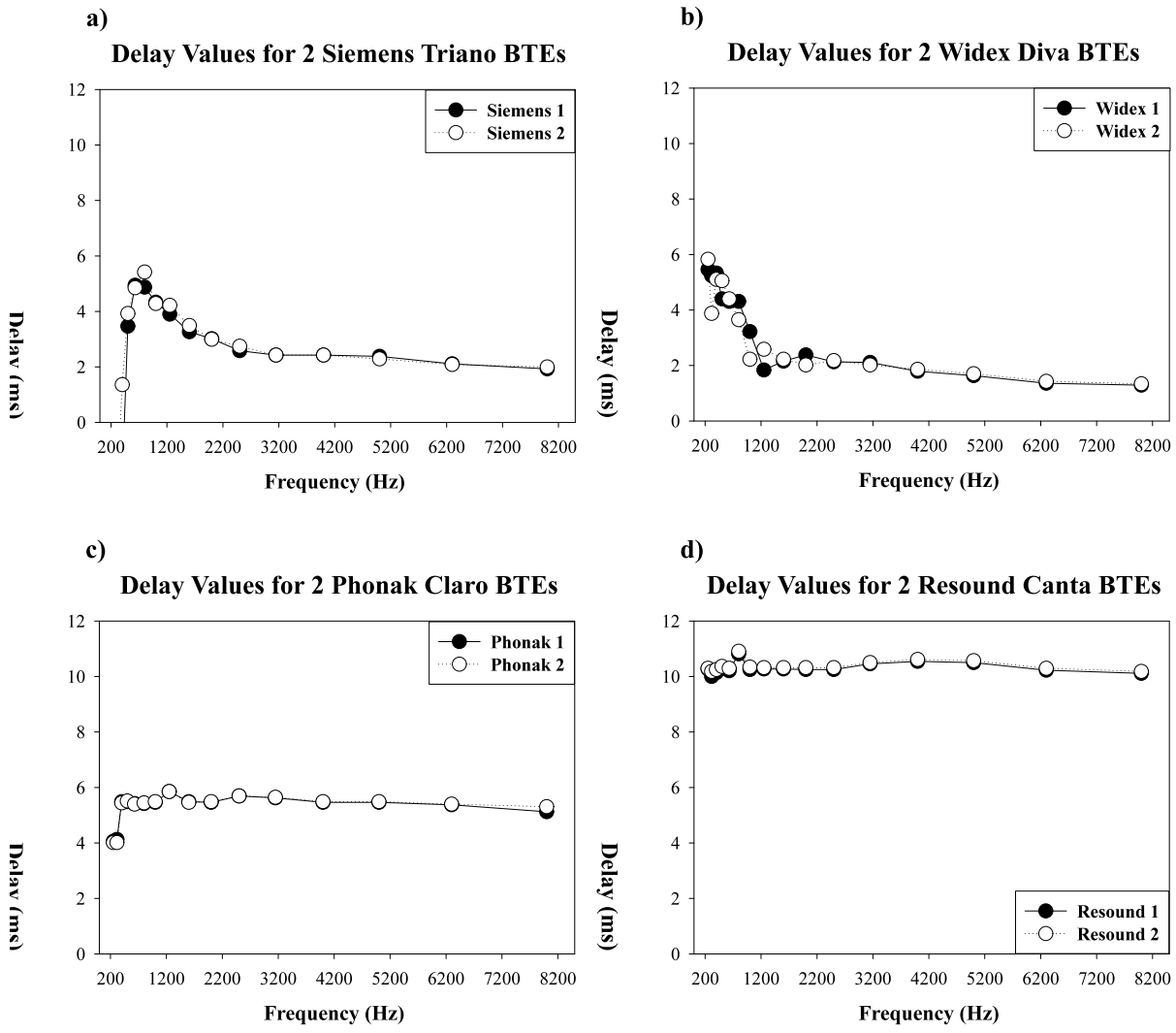


Figure 2-11: Measured delay values for a) Siemens Triano BTE b) Widex Diva BTE c) Phonak Claro BTE and d) Resound Canta BTE. The x-axis displays frequency in Hz and the y-axis displays delay values in ms.

In addition to digital delay, there may be a physical delay caused by the fitting of the hearing aid. With advances in feedback cancellation algorithms, audiologists are able to fit digital hearings with an open-fit earmold. The advantages of the open-fit platform include minimal occlusion and provision of amplification only in the region of hearing loss for individuals with primarily a high frequency hearing loss. Open fit earmolds allow for two pathways of sound transmission to the eardrum. The openness of the earmold allows the natural

direct pathway of sound to reach the eardrum directly. All frequencies reach the eardrum, but the high frequency energy is attenuated by the high frequency hearing loss of the listener (Mueller & Ricketts, 2006). Amplified high frequency energy is delivered to the ear via the digital hearing aid by the open-fit earmold. The arrival time of the amplified high frequencies is delayed relative to the sound that arrived at the eardrum via the direct pathway. As the low frequencies are the most, and often times the only frequencies, audible to hearing-impaired listeners from the unamplified direct pathway, there is a an asynchronous delay between the low frequencies and the amplified high frequencies at the tympanic membrane.

As a result, the sound at the eardrum is spectrally asynchronous due to the combination of the direct air conduction pathway and the output of the digital hearing aid. Figure 2-12 shows measurements of the effects of an open-fit earmold on the delay values at the output of the amplification device. These measurements were taken from the ear canal of the Knowles Electronic Mannequin for Acoustic Research (KEMAR) who was fit with the same hearing aid twice, once with an occluding earmold, and then with an open fit earmold. The recordings for this measurement were obtained using an Etymotic Research ER-7 microphone and a Zwislocki coupler inside of KEMAR. In an anechoic chamber, KEMAR was positioned in front of a loudspeaker at 0° azimuth. The relative delay of the amplification device was measured by subtracting the arrival time at the microphone of an impulse sound generated from a speaker without a hearing aid present from the arrival time of the impulse sound at the microphone with a functional hearing aid present.

A MATLAB code calculated the relative delay between the peak-to-peak amplitude of the impulse sound between unaided and aided conditions. With an occluding earmold, there is a flat delay of the acoustic signal to the ear of 6.5 ms between 500 and 8000 Hz. The same hearing

aid fit with an open-fit earmold generated a spectrally asynchronous delay with essentially no delay up to 2000 Hz, then abrupt rise to a 6.5 ms delay through 6000 Hz. There is no delay in the low frequencies because the direct pathway of sound via the open-fit earmold causes the peak amplitude between unaided and open-fit aided condition to occur at the same time. The peak amplitude for the high frequencies is the delayed output of the hearing aid device. This does not mean that all high frequency energy is delayed in an open fit hearing aid. Unamplified high frequency energy does enter the ear canal, but the equation for calculating relative delay only uses the measured maximum peak amplitude of each frequency as the arrival time. A third measurement was made with the open-fit hearing aid turned off. This measurement shows that the direct pathway of sound to the ear canal is not delayed relative to a true open ear with no hearing aid present (Figure 2-12). However, this unamplified high frequency energy, though not delayed, is not audible to hearing-impaired listeners most of the time. The resulting delay of the audible high and low frequency energy of the input signal is spectrally asynchronous.

Recently, with the advent of aggressive feedback cancellation algorithms, more clinicians are opting for open-fitting schemes for their patients. The open-fit amplification device delivers amplified high frequency energy to the ear, while the audible low frequency energy travels through the open ear canal. This hearing aid style is an attractive solution for the sloping high-frequency hearing loss that is commonly caused by aging and noise exposure. Due to the rise in the popularity of open fit hearing aids, manufacturers are inventing hearing aid devices specifically designed for the open-fit platform.

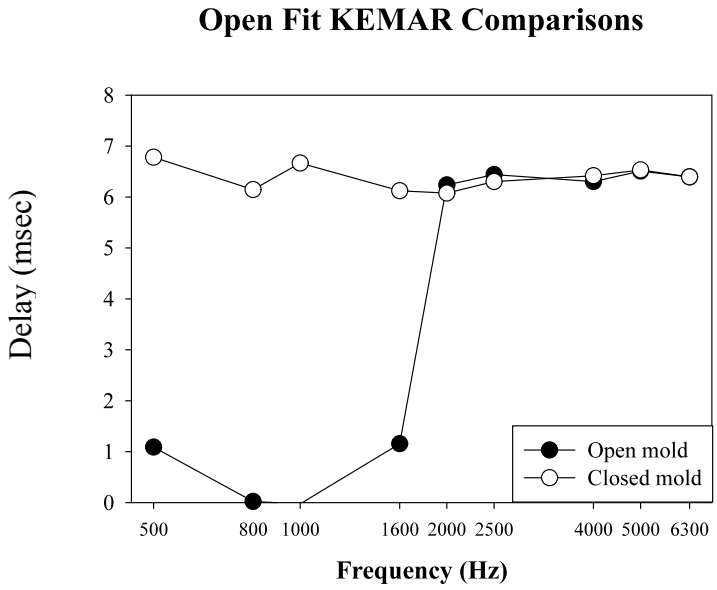


Figure 2-12: The real-ear measurements showing the delay as a function of frequency with a closed-fit and an open-fit earmold attached to the same hearing aid. The x-axis displays frequency and the y-axis displays the delay values of the hearing device. For the open-fit earmold the high frequencies were delayed by the DSP hearing aid, causing a spectrally asynchronous delay.

2.4.1 Consequences of Acoustic Delay—A brief review of the literature

Recently, with the increase in digital hearing aid products and open fitting schemes, researchers have been interested in quantifying the deleterious effects of acoustic delay. Although the research literature examining the effects of acoustic delays on speech perception and production is variable (See Table 2-2 for a review of findings), there is a conclusion that spectrally asynchronous delays are more detrimental than spectrally synchronous delays to the listener's tolerance and performance (Greenberg, Arai & Silipo 1998; Grant & Greenberg, 2001; Stone & Moore 2003).

Stone and Moore conducted a series of studies (1999, 2002, 2003, 2005, 2008) to examine the perceptual consequences of acoustic delay. In their studies they used three different

outcome measures: a 7-point rating scale to rate the subjective perception of annoyance due to acoustic delay, a vowel-consonant-vowel (VCV) identification task, and a speech production measure. For each outcome measure, a different tolerable delay value was obtained. Overall their results indicated that increasing auditory delay has a negative impact on listeners' perception. Specifically, their results indicated that participants are least tolerant of the qualitative effects of spectrally asynchronous acoustic delays. Participants rated delays as short as 9 ms as disturbing, even though the acoustic delays did not begin to disrupt the participants' speech identification abilities until about 15 ms (Stone & Moore 2003). Speech production was not affected by spectrally asynchronous delays, but it was affected by synchronous delay greater than 30 ms (Stone & Moore, 2002, 2003). When Stone et al. (2008) manipulated asynchronous delay so that it was similar to the delay introduced by an open-fit hearing aids, results indicated that listeners were even less tolerant of delay with values of 5-6 ms being rated as disturbing.

Many studies involving acoustic delays used sentence materials as the measure of intelligibility performance. Greenberg et al (1998) found that speech recognition performance is relatively unaffected until the acoustic delay exceeds 50 ms. The increased tolerance for spectrally asynchronous delays is most likely due to the acoustic redundancy found in sentence material as opposed to the VCV clusters used by Stone and Moore. Despite this increase of acoustic redundancy, listeners are still affected by these delays. Acoustic delays interfere with auditory-visual speech recognition (Grant & Seitz, 2000). For auditory-visual speech perception, the allowable delays can be as much as 160 ms before speech recognition of sentences deteriorates (Grant & Greenberg, 2001).

It is important to note the type of acoustic delay the above researchers used in their study. Stone and Moore used synchronous delay values (1999, 2002, 2005), a spectrally asynchronous

delay with the low frequencies being more delayed than high frequency energy (2003), and a spectrally asynchronous delay with the high frequencies being delayed relative to low frequencies (2008). Grant and Greenberg (2001) and Greenberg et al (1998) generated spectrally asynchronous sentence stimuli in which two mid frequency energy bands were delayed relative to two lateral bands of energy. Grant and Seitz (2000) used synchronous delay values in testing auditory-visual perception. Although these studies give information about the effects of spectrally-asynchronous delays, only one (Stone et al., 2008) introduced signal manipulations that mimic the acoustic delay values at the output of an open fit digital hearing aid. The resultant delay of an open-fit device is asynchronous in that the amplified high frequency energy above 2000 Hz is delayed, while the low frequency energy is not delayed at all (Figure 2-12). This single study indicated that listeners might be more susceptible to the subjective consequences of this type of delay. It would be useful to know if there are any objective consequences of such a delay.

Table 2-2: Review of research regarding the impact of auditory delay

Reference	Outcome Measure	Stimuli Type & Presentation Modality	Type of delay	Hearing function of subjects	Maximum tolerable delay
McGrath & Summerfield, 1985	Performance on a sentence recognition task	Video recorded sentence material with and auditory presentation of F0 pulse train Presented Auditory-Visually	Spectrally synchronous	Normal hearing subjects	40 ms
Grant & Seitz, 1998	Performance on a sentence recognition task in background noise	Auditory and Video recorded sentence materials with a fixed SNR Presented Auditory-Visually	Spectrally synchronous	Mild to severe sloping sensorineural hearing loss	200 ms

Arai & Greenberg, 1998	Performance on a sentence recognition task	Audio recorded sentence material filtered in to 19 ¼ octave channels Presented Audition only	Spectrally asynchronous These 19 channels were then delayed with respect to each other to create "jittered" speech	Normal hearing subjects	140 ms
Greenberg, Arai, & Silipo, 1998	Performance on a sentence recognition task	Audio recorded sentence material then filtered into 4 1/3 octave bands Presented Audition only	Spectrally asynchronous delays with the mid frequency bands varied relative to the lateral bands	Normal hearing subjects	50 ms
Stone & Moore, 1999	7 point rating scale of disturbance due to delay of acoustic signal	Subjects listened to a recorded passage of read text Presented Audition only	Spectrally synchronous	Normal hearing subjects with simulated hearing loss	20 ms
Agnew & Thornton, 2000	Subjects manually adjusted the amount of group delay introduced by the aid by adjusting a slider on a computer. Subjects adjust the amount of delay until they were just able to notice the delay and further increased the delay until they reported it to be "objectionable"	Subjects spoke and rated the effect of delay on the sound of their own voice.	Spectrally synchronous	Normal hearing	3-5 ms was "noticeable" > 10 ms was "objectionable
Grant & Greenberg, 2001	Performance on a sentence recognition task	Auditory and Video recorded sentence materials Audio consisted of two spectral slits rather than the full bandwidth of speech	Spectrally synchronous	Normal hearing subjects	160 ms

Stone & Moore, 2002	7 point rating scale of disturbance due to delay of acoustic signal Laryngographic measures of speech production	Subjects read aloud a passage of written text Presented Audition only	Spectrally synchronous	Normal hearing subjects	20 ms for disturbance rating 30 ms for speech production disruption
Stone & Moore, 2003	7 point rating scale of disturbance due to delay of acoustic signal Speech perception performance score Measurement of speech production rates	Subjects read aloud a passage of written text VCV syllables for speech perception measures Presented Audition only	Spectrally asynchronous delays Low frequencies were delayed relative to high frequencies	Symmetric, bilateral moderate sensorineural hearing loss	9 ms for disturbance rating 15 ms for decreased performance on VCV identification task Speech production was not affected by the delays introduced in this study
Stone & Moore, 2005	7 point rating scale of disturbance due to delay of acoustic signal	Subjects read aloud a passage of written text	Spectrally synchronous	Symmetric, bilateral sensorineural hearing loss	Slight HL = 23 ms Mild HL = 15 ms Moderate HL =32 ms
Stone, Moore, Meisenbacher, & Derleth, 2008	7 point rating scale of disturbance due to delay of acoustic signal	Subjects listened to 5 second recordings of continuous discourse	Spectrally synchronous & Spectrally asynchronous simulating the delay found in open-fit hearing aids	Normal hearing subjects One condition involved a simulated hearing loss	5-6 msec for gain plus spectrally synchronous delay 5 msec for spectrally asynchronous delay with a 2k Hz high frequency delay Results for simulated hearing loss and high frequency delay/gain inconclusive

Envelope Onset Asynchrony (EOA) is the time asynchrony between high and low frequency energy onset in naturally produced speech (Yuan et al., 2004). Voiceless CVs tend to

have positive EOA so that high frequency energy onset precedes low frequency energy. If you manipulate the EOA of a naturally produced voiceless CV by delaying the high frequency energy onset, the perception changes to that of its voiced cognate (Ortmann et al., 2010). It may be possible that the introduction of the asynchronous delay by the open fitting platform could interfere with the wearer's perception of voicing. The audible low frequency energy of the syllable /pa/ could travel via the ear canal and arrive at the eardrum prior to the arrival of the amplified high frequency energy thus causing the perception to be more like /ba/. Further research in which the asynchronous delay is manipulated in a similar fashion to digital hearing aid devices needs to be conducted in order to confirm this hypothesis.

2.4.2 **Summary and Empirical Question**

The review of speech perception research points to many acoustic cues used by listeners to aid in phonemic recognition. The output of the human articulator contains acoustic patterns that vary in frequency, intensity, and time. The structured variance of these dimensions with respect to one another form the cues listeners use to perceive speech. In reviewing the literature on the speech perception ability of normal-hearing listeners and hearing-impaired listeners with mild-moderate sensorineural hearing loss, it was found that in quiet listening environments both groups of listeners use the same acoustic cues for speech perception. Hearing-impaired listeners tend to rely on temporal, tempo-intensity, and some spectro-temporal properties of speech for speech perception (Johnson et al., 1984; Lindholm et al., 1988; Turner et al., 1995, Ortmann et al., submitted).

In everyday situations hearing-impaired listeners are exposed to conversationally spoken speech that is different from the clearly spoken speech or synthetic speech that is used in

audiometric speech perception tests. The most obvious acoustical differences between clearly produced and conversationally spoken speech are the faster articulation rate and fewer pauses of conversational speech (Picheny et al., 1986; Uchanski et al., 1996; Krause & Braida, 2002; Liu & Zeng, 2006). In addition to the fast rate of conversational speech, there are acoustic alterations at the phonemic level. The spectral bursts of word-final plosive consonants are often times shortened or omitted (Picheny et al., 1986), the temporal distinction or gaps between syllables and words are smaller (Picheny et. al., 1986; Byrd & Tan, 1996), and the dynamic spectro-temporal cues, such as formant transitions and voice onset time are shorter (Picheny et. al., 1986; Ferguson & Kewley-Port, 2002, 2004). The degradation of these acoustic cues by conversational speech increases perceptual ambiguity by hearing-impaired listeners (Picheny et. al., 1985; Payton et. al., 1994;). While normal-hearing listeners can capitalize on the redundancy of these cues, hearing-impaired listeners are not as fortunate.

When assessing the effect of current signal-processing schemes on speech intelligibility, it is important to use conversational speech as stimuli in order to not only capture the "real world" hearing aid benefit by the hearing-impaired listener, but also to see the interaction between the hearing device and conversational speech's rapidly changing acoustics. One possible interaction stems from the digital delay introduced by digital signal processing in combination with open-fitting schemes. It is hypothesized that these spectrally asynchronous delays disrupt the temporal cues hearing-impaired listeners need to accurately perceive conversational speech.

This review of the literature leads to the following question.

- Does the introduction of spectrally asynchronous delay that is similar to the delay introduced by open-fit digital hearing aids, lead to poorer speech intelligibility of conversationally spoken speech by mild-moderate hearing-impaired listeners?
    - If so, how much spectrally asynchronous delay can be tolerated before speech intelligibility is affected?

3.0 METHODS

In order to answer the aforementioned empirical questions, careful consideration was taken to ensure that the proposed project would be constructed so that variability due to extraneous factors would be minimized. A Pre-Experiment validated the stimuli used in the Main-Experiment that focuses on the question, "Does the introduction of spectrally asynchronous delay that is similar to the delay introduced by open-fit digital hearing aids, lead to poorer speech intelligibility of conversationally spoken speech by mild-moderate hearing-impaired listeners?" The following section outlines the characteristics of the speech stimuli, signal processing conditions, study participants, and procedures for administration of the protocol for both the Pre-Experiment and the Main-Experiment. A description of the statistical analysis and a list of the possible outcomes are presented.

3.1 PRE-EXPERIMENT

3.1.1 **Speech materials**

The speech materials used for this research study are the sentences from the revised Speech Perception in Noise (r-SPIN) test (Bilger, Nuetzel, Rabinowitz, & Rzeczkowski, 1984). The r-SPIN consists of eight lists of 50 sentences. Within each list are 25 high predictability and 25

low predictability sentences. High predictability sentences are those in which sentence context serves as a cue for determining the final monosyllabic word of the sentence. An example of a high predictability sentence is "The doctor prescribed the DRUG". Using sentence context, listeners can identify the final word as "drug" even if the acoustical properties of the word are degraded by background noise. A low predictability sentence provides limited linguistic context to cue the final word. For example, "She has known about the DRUG" is a low predictability sentence. Lack of contextual cues will make the final word harder to identify than when it is in the high predictability sentence. Although the word "drug" is the same in both sentences, listeners are more reliant on the acoustic characteristics of the word when it is in the low predictability sentence. Because the purpose of this research study is to see whether the acoustical consequences of spectrally asynchronous delay impact speech intelligibility, only the 25 low predictability sentences from each of the eight lists were used. This ensured that the listeners relied on acoustic cues for speech perception, not sentence context. Appendix A includes all of the low predictability sentences in their appropriate list.

Sentence material, as opposed to single word lists, was chosen so that the stimuli would best reflect conversational speech. Sentences also capture word and syllabic boundaries, which could be less distinct with the addition of spectrally asynchronous delay. The r-SPIN sentence materials were chosen not only because of the ability to control for the use of sentence context, but also because all of the sentences have been subject to rigorous psychometric testing to control for word familiarity, word frequency, prosodic factors, and phonetic content within and across each list of sentences (Kalikow, Stevens, & Elliot, 1977; Bilger et. al., 1984).

In the development of the SPIN, Kalikow and his colleagues (1977) formed an initial corpus of 1,148 homogenous sentences constrained to 5-8 words and 6-8 syllables each. These

sentences were constructed so that 669 sentences had a highly predictable final monosyllabic word and 479 sentences had low predictability for identification of the final word. All the words were controlled for word frequency in the English language. After conducting several speech perception tests with these sentences using normal- hearing listeners as participants, the authors threw out several hundred sentences due to either poor intelligibility of the sentences or low key word familiarity. The authors also ensured that phonetic content of the sentences and final keywords were typical of conversational language. The final products of this culling process were eight equivalent lists of 50 low and high predictable sentences (Kalikow et. al., 1977). Bilger and his group (1984) revised the original SPIN to ensure list equivalency among the low probability sentences. They also standardized the test with + 8 signal to noise ratio on 128 listeners with sensorineural hearing loss.

Kalikow et. al. (1977) compared the intelligibility of key words in high versus low predictability context presented with a signal to noise ratio of +10 dB. For a group of 81 normal-hearing listeners, the average score for low predictability (LP) key words was 88% correct. If the LP sentences were given to normal –hearing listeners in a quiet environment, the average score should be greater than 88%. This study's recorded LP sentences were presented to a group of normal-hearing listeners to ensure that the key words of the sentences were intelligible under optimal conditions.

3.1.2 **Conversational recordings**

A single male speaker with experience in radio and television public speaking was used for the recording of the LP sentence materials. The speaker was seated in a sound treated booth with his mouth 8 inches away from a mounted Audio-Technica cardiod-dynamic microphone (bandwidth

= 60 to 13,000 Hz). The microphone was routed to a Panasonic digital recorder with settings for a mono recording at a 44.1 kHz sampling rate. The sensitivity of the microphone was adjusted to prevent any peak clipping of the speaker's voice. The speaker was instructed to say each sentence at least four times. For the first sentence utterance, the speaker was instructed to read the sentence aloud in a clear manner. The following instructions from Schum (1996) were given.

> "Imagine that you are speaking to a person that you know is hearing-impaired. I want you to speak as clearly and precisely as possible. Try to produce each word as accurately as you can."

Once the satisfactory recording of a clearly produced sentence was made, as indicated by the speaker speaking slowly, carefully enunciating each word, then he was instructed for conversational speech. The speaker was told to memorize the sentence, and to say each sentence within conversation three times. He was instructed as follows.

> "Speak naturally as you would in conversation with your friends. Conversational speech is different from the clearly spoken speech you used before. For example, you tend to talk faster in conversation. Keep this in mind as you say these sentences again. I want you to sound as natural and conversational as possible."

For each sentence, the speaker was engaged in a conversation about the sentence. He said the sentence as memorized and immediately followed the sentence with another, so as to keep the natural flow of conversation. For example, if the sentence was "I was considering the crook", the speaker said, "I was considering the crook. He broke into my house the other day." Not only does the additional sentence kept the natural flow of conversation, but it also minimized the speaker's tendency to stress the final spoken word. The second sentence was given to the speaker

and was 10-12 syllables in length, so that all of the sentence pairs were similar in syllabic structure. The speaker was not given any guidelines as to how fast or "conversational" he should speak. Krause and Braida (2002, 2004) showed that there are individual differences in the articulation rate of conversational speech. The rules for the recording were that it must sound as natural as possible, that it must be spoken effortlessly, and that it is faster than his clear speech productions. When the recording session was complete, there were four recordings (one sentence will be clearly spoken and the three others will be spoken conversationally) of each of the 200 sentences.

3.1.3 **Analysis of Conversational Speech**

All soundfiles were edited in Adobe Audition 2.0. For each sentence recording, the soundfiles were low pass filtered (10 kHz cutoff) to remove any high frequency noise. The single clear production and the three conversational productions of each sentence were excised and saved in individual files. For each sentence, only one of the three conversational productions was chosen as the final sentence stimulus. The determinants for the final conversational recording stimulus were the speaking rate (words per minute, wpm), the articulation rate (syllables per second, syl/s), vowel duration, and VOT. Picheny et. al. (1986) found that the average rate of conversationally spoken speech was 200 wpm. Krause and Braida (2002) found that some talkers spoke conversationally at rates up to 315 wpm. The conversationally spoken sentences chosen to be part of the stimuli had average speaking rates between 200 and 315 wpm. These sentences were analyzed further with measurements of vowel duration and VOT. Ferguson and Kewley-Port (2002) reported that the vowel duration of conversational speech is roughly 50% that of clear speech. The VOT of word-initial voiceless plosives is shortened in conversational speech as

well (Klatt, 1975; Picheny et. al., 1986; Krause & Braida, 2004). The chosen conversational sentence presented vowel durations that were roughly half that of its clearly produced counterpart and a measureable decrease in the VOT of voiceless plosives. The speaking rate, vowel duration, and VOT of each of the recordings, clearly spoken, conversationally spoken, and the original r-SPIN recordings (Bilger et al., 1984) were analyzed and compared for differences.

Picheny et al. (1986) reported that word-final plosives sometimes are not released in conversational speech. As the listeners are being scored on the recognition of the final key word of the r-SPIN LP sentences, it was a criterion that the final word had to contain acoustic cues for every phoneme in the word. The change in speaking rate and durations of vowels and VOT, as well as the availability of phonemic cues in the final word served as the criteria for choosing each of the conversationally produced token of the r-SPIN LP test items. Once each sentence was chosen, the intensity levels were equated on average RMS via Adobe Audition. Each sentence was assigned into the appropriate equivalent list designated by Kalikow and colleagues (1977). There were 8 lists of 25 sentences for a total of 200 conversationally spoken sentences (see Appendix A). These 200 conversationally spoken sentences were presented to 15 normal-hearing listeners to test for the intelligibility of the final target word of each sentence.

3.1.4 **Subjects**

Fifteen normal-hearing participants were tested for the validation of the conversationally recorded sentences. The participants were recruited from the community of the St. Louis area, as well as from the caseloads of the Washington University School of Medicine Adult Audiology Clinic. All participants were between the ages of 19-30 years old (mean age = 23). The participants were excluded if they were not native speakers of English and/or if they reported

having recent middle ear pathology, otologic surgery, and/or neurologic pathology. The participants were not excluded on the basis of race or gender. All participants will be given informed consent in accordance with the guidelines of the Institutional Review Board (IRB) of both Washington University in St. Louis, Missouri and University of Pittsburgh in Pittsburgh, Pennsylvania. In addition to the aforementioned age and case history criteria, these participants all had audiometric thresholds less than 15 dB from 250 Hz through 8 kHz in both ears and word recognition scores of the Northwestern University monosyllabic word lists (NU-6) within the 90% confidence limit based on their pure tone average (Dubno, Lee, Matthews, Mills, & Lam, 1995).

3.1.5 **Procedure**

All participants were screened after obtaining informed consent. Only after signing the consent form were they considered enrolled in the study. A self-report case history questionnaire (Appendix B) was administered to inquire into the participants' general health, history of hearing loss and/or middle ear disease, otologic surgery, and neurologic disorder. To determine auditory status, a standard audiometric test battery was completed. This included bilateral air and bone conduction threshold testing at the standard audiometric frequencies (ASHA, 1978), and word recognition testing with the Auditec recording of the Northwestern University Test #6 (NU-6) word lists (Tillman & Carhart, 1966). All of the equipment used to perform the tests of auditory function was calibrated according to the appropriate ANSI standards. All testing was conducted in test rooms that meet the ANSI standards for maximum background noise.

Once the participants were confirmed to have normal hearing and met all of the inclusionary criteria, they participated in the listening task. The listeners were seated in an

audiometric test booth and listened to the sentences presented diotically at 60 dB SPL through ER-2 insert earphones.

Prior to the test session, the presentations level of the conversationally spoken sentences were verified to be 60 dB SPL RMS. First, a calibration soundfile containing white noise with an average RMS equal to the average RMS of the sentence recordings was made and saved onto the computer as "original calibration noise". Using a Frye 7000 analyzer, real ear probe microphone measures were used to verify the output of the insert earphone in the participants' ears. Using the "original calibration noise" soundfile as the output signal of the earphones, the attenuator of the audiometer was adjusted so that the time weighted output of the ER-2 earphone peaked at 60 dB SPL on the Frye 7000 probe microphone system. The sentence stimuli were presented at the specified audiometer dial HL level to the participant. Calibration was performed before each participant's test session to ensure equal presentation level across subjects.

Once the stimuli calibration was complete, the sentences in the pre-experiment were presented to each participant using SuperLab 4.0 (Cedrus Corporation) software on a computer connected to a Grason-Stadler 61 audiometer. The 200 conversationally recorded sentences were presented randomly to each participant. The participants entered their responses in an Excel worksheet on a computer in front of them. The participants were instructed to "Listen carefully to the following sentences. Type the last word of each sentence in the appropriate blank on the worksheet. There are 200 sentences total and will be divided in groups of five with a 10 second pause between groups to give you time to compete your answers. You will be given a break after 25 sentences." The pre-experiment procedures took approximately 1-1.5 hours to complete.

3.1.6 **Data Analysis**

Once all of the data from the pre-experiment were collected, the data for each listener were converted into percent correct final-word identification. The data were then averaged across listeners and compared to the average score obtained by Kalikow et al. (1977).

3.2 MAIN-EXPERIMENT

3.2.1 **Stimuli**

The purpose of the main-experiment was to examine whether spectrally asynchronous delay affects the intelligibility of conversationally spoken sentences for hearing-impaired listeners. The stimuli used in the main-experiment were the set of 200 conversationally spoken sentences that was found to be intelligible by the group of normal-hearing listeners from the pre-experiment. The stimuli were modified so that the acoustic properties mimic the asynchronously delayed pattern of open-fit hearing aids. Figure 2-12 shows that the energy above 2 kHz is delayed in an open-fit hearing aid. Armed with this information, a MATLAB code was written to create four conditions in which the frequencies above 2 kHz were delayed by a specific value (0, 4, 8, or 32 ms) relative to the frequencies below the 2 kHz cut-off.

3.2.2 **Simulated spectrally asynchronous delay**

The steps for creating the asynchronously delayed sound files are depicted in Figure 3-1. Each sentence was filtered into a high-pass energy and a low-pass energy band using a digital finite impulse response (FIR) filter with an order of 50. Appendix C shows the magnitude and the phase response of the FIR filters. A FIR filter was chosen because it does not distort the phase of the signal. An order of 50 allows for stop-band attenuation with minimal ripple. The high pass band was then amplified by 18 dB according to the 1/3 gain rule for a moderate high frequency hearing loss of 55 dB. The 1/3 gain rule is used for many hearing aid prescriptive gain targets. Following the flat 18 dB amplification of the high frequencies, a delay value of 0, 4, 8, or 32 ms was applied. Zero ms represented the condition with no added asynchronous delay. Four and 8 ms are the conditions that are similar to the delay values of DSP hearing aids, whereas 32 ms is the condition in which a perceptual consequence of asynchronous delay is expected (Stone & Moore, 2003).

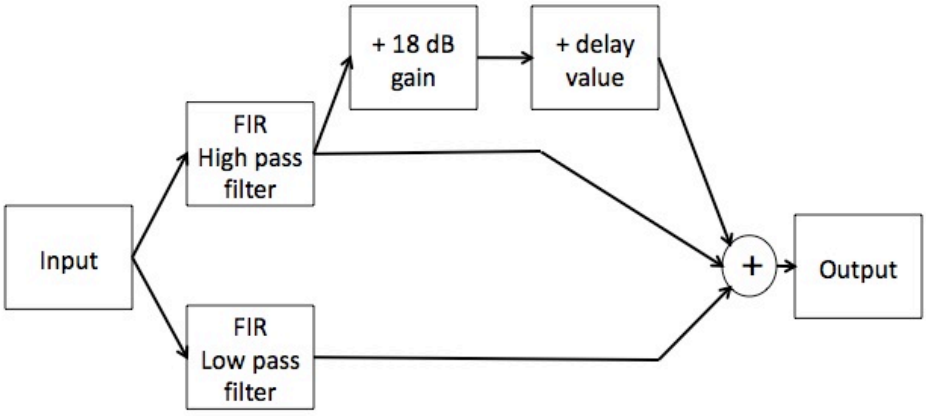


Figure 3-1: Diagram of the acoustic modifications to each of the sentence stimuli. The + 18 dB gain and delay pathway represents the DSP hearing aid pathway.

The amplified delayed high passed sound file was added to both the unamplified non-delayed high-pass sound file and the low-pass sound file to create the final output of a sound file. This output is similar to the acoustic pattern of sound at the eardrum of an open-fit hearing aid wearer. The addition of the unamplified non-delayed FIR high-pass filter and the FIR low-pass filter equals the input signal, thus representing the "natural pathway" of sound to the eardrum. The amplified delayed FIR high-pass band represents the output of the hearing aid. This MATLAB code allowed for manipulation of the high frequency band delay values while keeping the "natural pathway" and the amplification value constant for all sentence stimuli. Table 3-1 outlines the prepared stimuli.

Table 3-1: Gain and delay values for each of the 200 r-SPIN Low Predictability Sentences

	200 r-SPIN Low Predictability Sentences	200 r-SPIN Low Predictability Sentences	200 r-SPIN Low Predictability Sentences	200 r-SPIN Low Predictability Sentences
Gain value > 2 kHz	+18 dB	+18 dB	+18 dB	+18 dB
Delay value > 2 kHz	0 ms	4 ms	8 ms	32 ms

Figure 3-2 displays a series of spectrograms for the conversationally spoken recording of "Mr. Smith thinks about the CAP." The top spectrogram (a) displays the original recording while (b) displays the same sound file after a low-pass 50-order FIR filter at 2 kHz. The third spectrogram (c) displays the output after a high pass 50-order FIR filter at 2 kHz. Figure 3-2 (d) displays the same spectrogram as (c) only a delay of 32 ms and a gain of +18 dB was applied; this spectrogram represents the "hearing aid" pathway of an open fit device. The bottommost figure (e) represents the final sound file that was used as stimuli for the study. It is the

combination of spectrograms (b + c + d) and represents the sound that arrives at the eardrum via the natural sound pathway and the hearing aid pathway. The yellow line that bisects all of the spectrograms marks the onset of the high frequency band in the original recording. The 32 ms delay onset of the high frequency band relative to the original recording is seen in figure (d). Note that when the non-delayed (c) and delayed high frequency bands (d) are added together, the gaps between syllable and words, particularly between "the" and the final word "CAP", are overlaid and reduced (e).

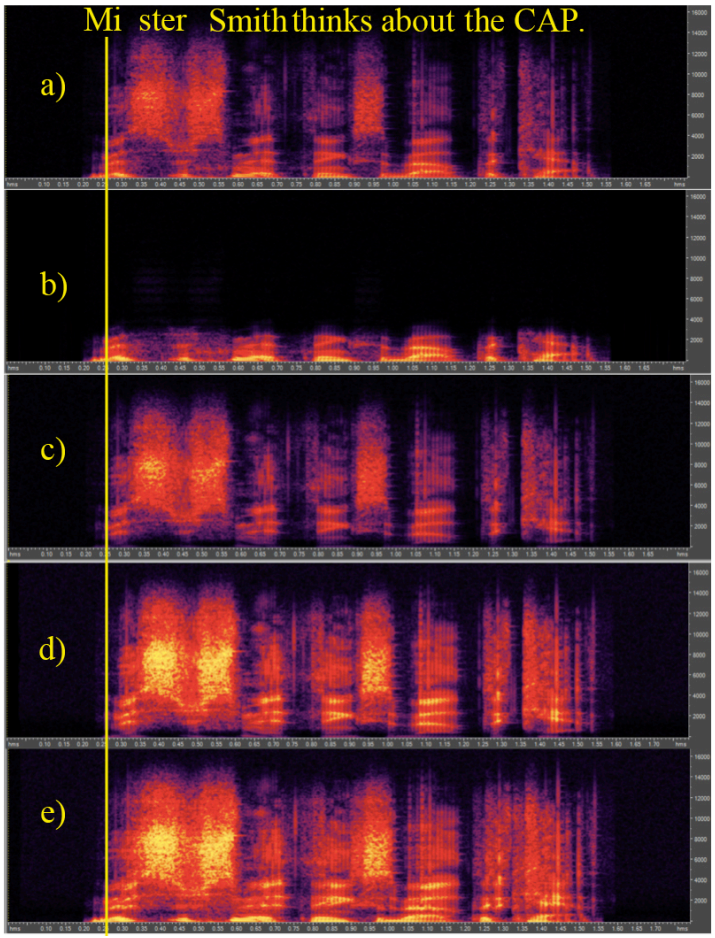


Figure 3-2: A series of spectrograms depicting the generation of the final stimuli for the conversationally spoken sentence, "Mr. Smith thinks about the CAP". a) The original conversational recording b) the original recording with a LP FIR filter at 2 kHz applied c) the original recording with a HP FIR filter at 2 kHz applied d) the same sound file as (c) only with a 32 ms onset delay and a 18 dB gain applied, representing the hearing aid pathway of sound and d) the final sound file that served as the stimuli which is the combination of (b + c + d), representing the sound arriving at the ear drum with the combination of the natural pathway and the hearing aid pathway. Note the reduction of temporal gaps between syllables in (e).

In addition to the reduction of the gaps between words and syllables, the spectrally asynchronous delay also caused a jitter or echo of spectral cues such as formant transitions. Figure 3-3 displays the series of spectrograms for the sentence "I can't consider the PLEA." Just as the previous figure (a) is the original file, (b) is the LP energy, (c) is the HP energy, (d) is the HP energy with 18 dB gain and 32 ms delay, and (e) is the final stimuli constructed by adding (b + c + d). The formant transition in the final word "PLEA" in Figure 3-3 (e) is somewhat blurred with a 32 ms spectrally asynchronous delay.

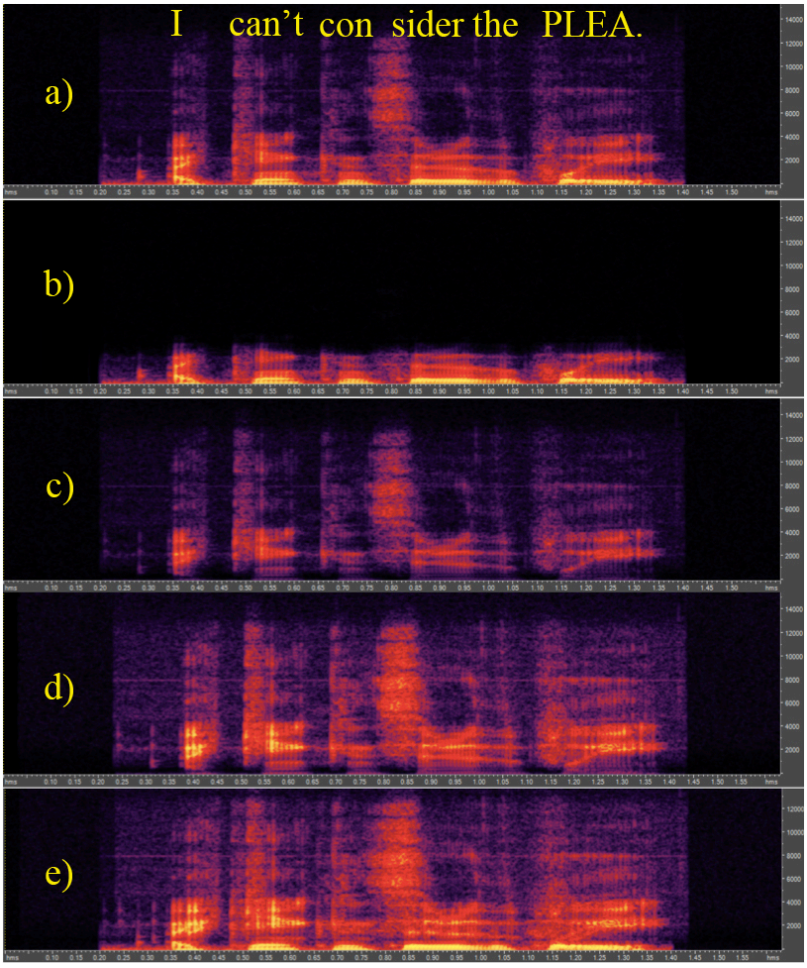


Figure 3-3: A series of spectrograms depicting the generation of the final stimuli for the conversationally spoken sentence, "I can't consider the PLEA". a) The original conversational recording b) the original recording with a LP FIR filter at 2 kHz applied c) the original recording with a HP FIR filter at 2 kHz applied d) the same sound file as (c) only with a 32 ms onset delay and a 18 dB gain applied, representing the hearing aid pathway of sound and d) the final sound file that served as the stimuli which is the combination of (b + c + d), representing the sound arriving at the ear drum with the combination of the natural pathway and the hearing aid pathway. Note the blurring of the formant transitions of "PLEA" in (e).

Spectrally asynchronous delays affect the voice-onset-time (VOT) of initial voiceless plosives. Figure 3-4 shows the 32 ms asynchronously delayed stimulus construction of the sentence, "We're speaking about the TOLL." In the original recording (a), "TOLL" has a VOT of 20 ms, however when a 32 ms asynchronous delay is applied the VOT is -12 ms (e). Figure 3-5 shows the same sentence construction with a 8 ms asynchronous delay (e), making the VOT of "TOLL" 12 ms.

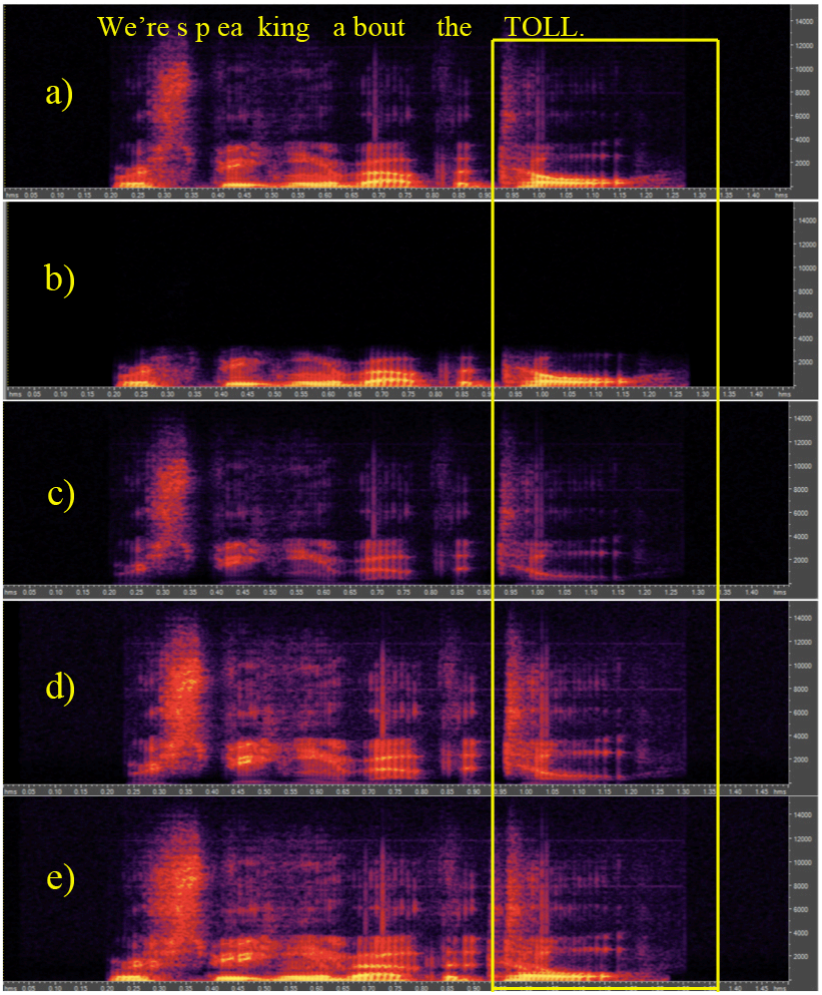


Figure 3-4: A series of spectrograms depicting the generation of the final stimuli for the conversationally spoken sentence, "We're speaking about the TOLL". a) The original conversational recording b) the original recording with a LP FIR filter at 2 kHz applied c) the original recording with a HP FIR filter at 2 kHz applied d) the same sound file as (c) only with a 32 ms onset delay and a 18 dB gain applied, representing the hearing aid pathway of sound and d) the final sound file that served as the stimuli which is the combination of (b + c + d), representing the sound arriving at the ear drum with the combination of the natural pathway and the hearing aid pathway. Note the shorter VOT of "TOLL" in (e). The VOT in (a) is 20 ms while the VOT in (e) is -12 ms.

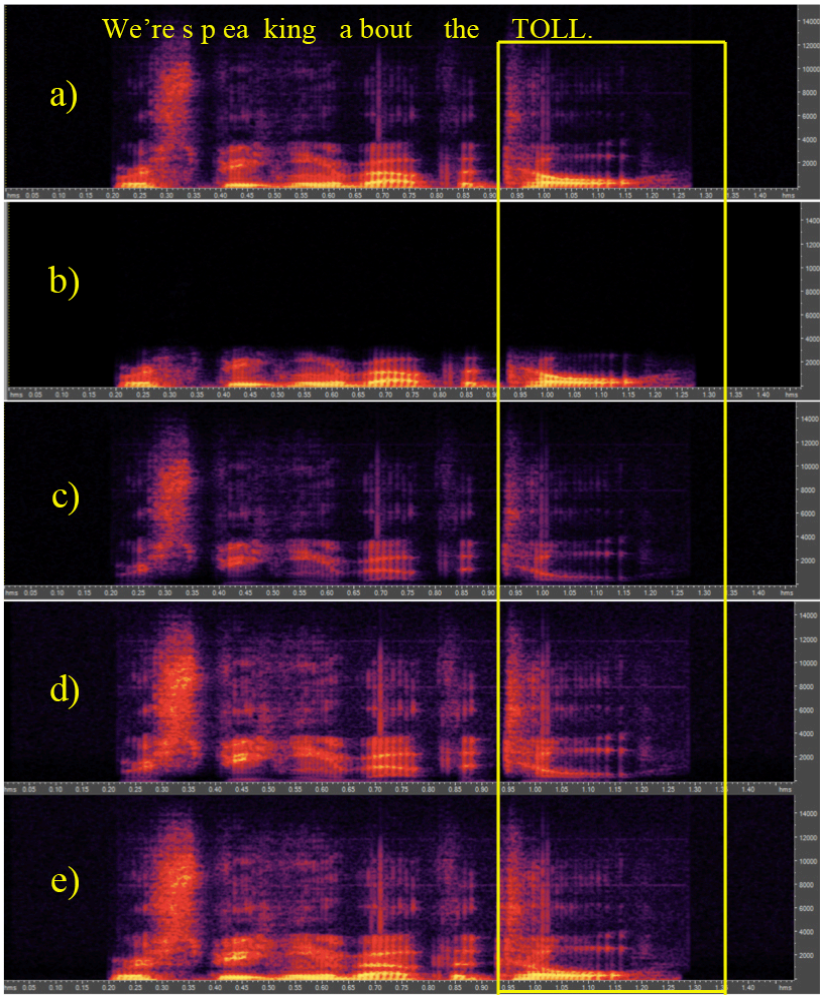


Figure 3-5: A series of spectrograms depicting the generation of the final stimuli for the conversationally spoken sentence, "We're speaking about the TOLL". a) The original conversational recording b) the original recording with a LP FIR filter at 2 kHz applied c) the original recording with a HP FIR filter at 2 kHz applied d) the same sound file as (c) only with a 8 ms onset delay and a 18 dB gain applied, representing the hearing aid pathway of sound and d) the final sound file that served as the stimuli which is the combination of (b + c + d), representing the sound arriving at the ear drum with the combination of the natural pathway and the hearing aid pathway. Note the shorter VOT of "TOLL" in (e). The VOT in (a) is 20 ms while the VOT in (e) is 12 ms

Figure 3-6 and Figure 3-7 depict the change in VOT for the key words "TANKS" and "STAMP". With an 8 ms spectrally asynchronous delay the spectral burst of the /t/ within each word occurred at the onset of the vowel vocalization (compare (a) and (e) in both figures). Also, for "STAMP" in Figure 3-7, the temporal gap between the /s/ and the /t/ was shortened.

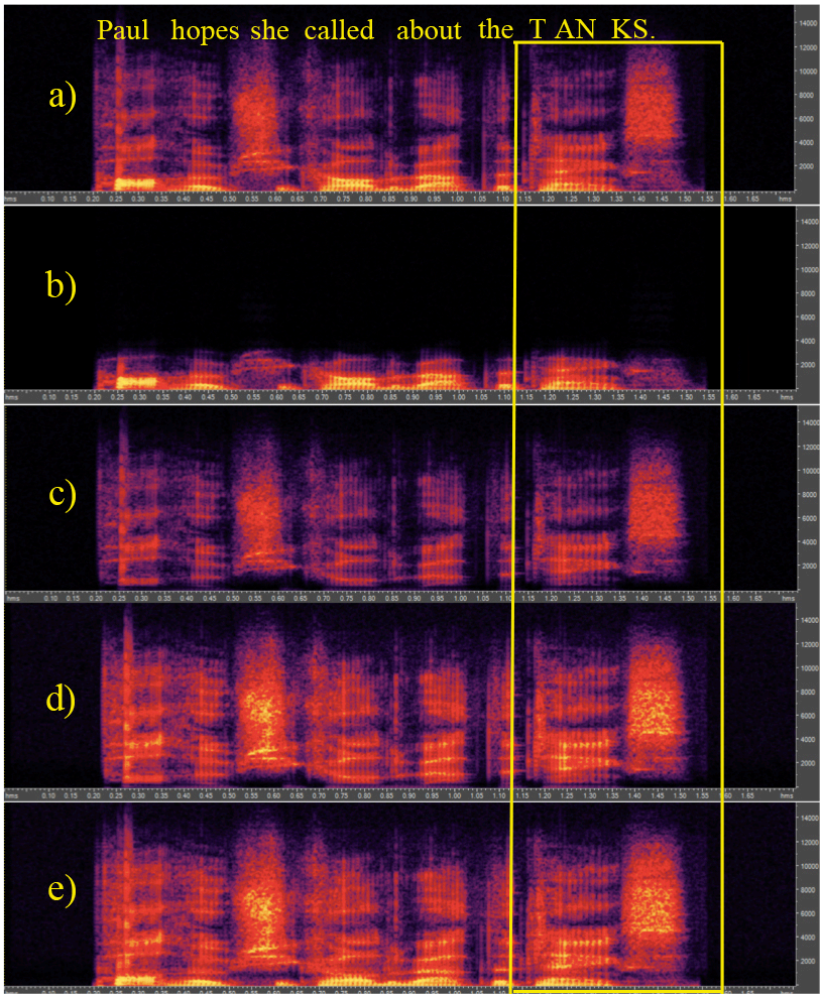


Figure 3-6: A series of spectrograms depicting the generation of the final stimuli for the conversationally spoken sentence, "Paul hopes she called about the TANKS". a) The original conversational recording b) the original recording with a LP FIR filter at 2 kHz applied c) the original recording with a HP FIR filter at 2 kHz applied d) the same sound file as (c) only with a 8 ms onset delay and a 18 dB gain applied, representing the hearing aid pathway of sound and d) the final sound file that served as the stimuli which is the combination of (b + c + d), representing the sound arriving at the ear drum with the combination of the natural pathway and the hearing aid pathway. Note the shorter VOT of "TANKS" in (e).

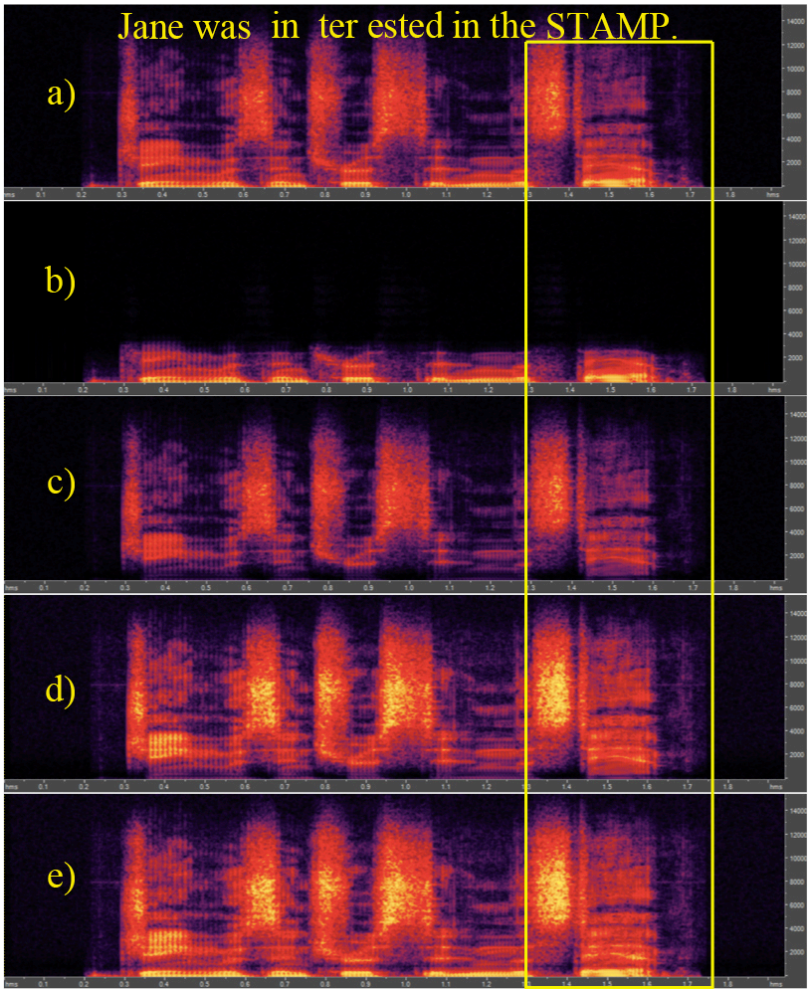


Figure 3-7: A series of spectrograms depicting the generation of the final stimuli for the conversationally spoken sentence, "Jane was interested in the STAMP". a) The original conversational recording b) the original recording with a LP FIR filter at 2 kHz applied c) the original recording with a HP FIR filter at 2 kHz applied d) the same sound file as (c) only with a 8 ms onset delay and a 18 dB gain applied, representing the hearing aid pathway of sound and d) the final sound file that served as the stimuli which is the combination of (b + c + d), representing the sound arriving at the ear drum with the combination of the natural pathway and the hearing aid pathway. Note the shorter VOT and shorter gap between the /s/ and the /t/ of "STAMP" in (e).

The spectrally asynchronous delays modified the original recordings so that the temporal gaps between syllables and words were shorter, the VOT of voiceless plosives were shortened, and the formant transitions were jittered and weakened. Stimuli with spectral asynchronous delay values of 0 ms (control), 4 ms, 8 ms, and 32 ms were created to examine their impact on speech intelligibility of both normal-hearing and hearing-impaired listeners.

3.2.3 **Subjects**

Based on a moderate effect size, alpha set to 0.05, it was calculated that 12-32 participants are necessary to achieve a power level of 0.8. As the design of this experiment is a repeated measures ANOVA, the degree of correlation between measures has a significant impact on power. Because the same participants were tested under each of the 4 asynchronous delay conditions (0 ms, 4 ms, 8 ms, and 32 ms), some degree of correlation is expected between measures. Table 3-2 displays the relationship between the correlation between measures and sample size necessary to achieve a power of 0.08. For this study, a correlation of 0.4 was chosen, as it is not the weakest or strongest correlation. The correlation between measures will likely be higher than 0.4 as it would be expected to have consistent ranking of participants across conditions. Positing a correlation of 0.4 errs on the side of caution in case the data display a weaker correlation than expected. Therefore 25 participants in each group were enrolled in the main experiment examining the effects of asynchronous delay on the intelligibility of conversational speech.

Table 3-2: Power analysis for Main Experiment

Moderate effect size, alpha = 0.05, power = 0.8	
Value of Correlation	Required Sample Size
0.2	32
0.3	29
0.4	25
0.5	20
0.6	17
0.7	12

The participants were recruited from the community of the St. Louis area, as well as from the caseloads of the Washington University School of Medicine Adult Audiology Clinic. All

participants were between the ages of 21-65 years old (Average age = 57 years, sd = 7.2). Age-related decline in auditory temporal processing has been shown to begin in the sixth decade of life (CHABA, 1988). Dubno, Dirks, and Morgan (1982) found that a group of subjects over the age of 65 with normal hearing required a higher articulation index than a group of younger normal hearing subjects to achieve 50% recognition of the LP r-SPIN sentences in noise. Therefore, the age of 65 served as the age cut-off for recruitment. The participants were excluded if they were not native speakers of English and/or if they reported having recent middle ear pathology, otologic surgery, and/or neurologic pathology. They also were excluded if they were current hearing aid users to control for hearing aid experience. The participants were not excluded on the basis of race or gender. All participants were given informed consent in accordance with the guidelines of the Institutional Review Board (IRB) of both Washington University in St. Louis, Missouri and University of Pittsburgh in Pittsburgh, Pennsylvania. Appendix D contains the individual demographic data for all participants of the main experiment.

The hearing-impaired participants for the main experiment presented high frequency hearing loss consistent with candidates for open-fit amplification. Figure 3-8 displays the average audiometric data for the hearing-impaired participants. All of the participants had speech reception thresholds that were within 8 dB of their 3-frequency pure tone average (500, 1000, & 2000 Hz). The word recognition score of each participant was within the 90% confidence interval according to his or her pure tone average (Dubno, et al., 1995).

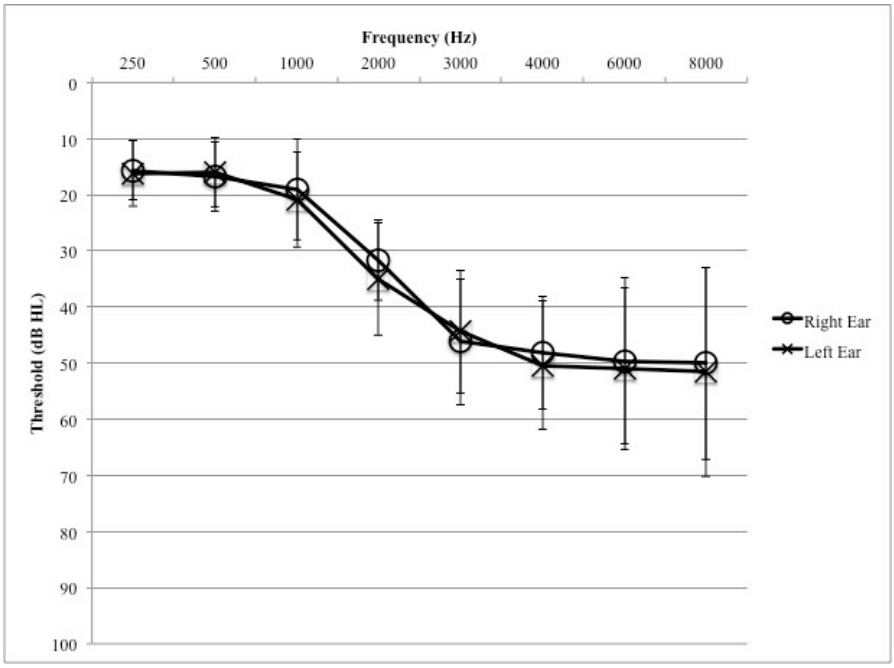


Figure 3-8: Average audiometric data of the 25 hearing-impaired participants

The rational behind allowing more severe hearing losses past 4 kHz results from two arguments. The first is the fact that the hearing loss of individuals with age-related high frequency sensorineural hearing loss tends to continue a downward slope past 4 kHz. Including these individuals will allow the data to generalize to this clinical population. Secondly, most open-fit hearing aids on the market tend to roll-off amplification past 3-4 kHz. This is due in part to the limited bandwidth of the hearing aids. With the advent of receiver in the canal (RIC) open-fit devices, manufacturers claim that the bandwidth of amplification extends out to 6-8 kHz. However test box measurements obtained by researchers at the University of Pittsburgh have found that most RIC hearing aids do not have usable gain past 3-4.7 kHz. Table 3-3 compares the measured bandwidth as defined as useable gain for a flat 50 dB hearing loss of both receiver in the hearing aid with open fit tubing and receiver in the canal hearing aids by several manufacturers with the reported bandwidth of the aids by their manufacturer. Regardless of

whether the hearing aid is able to amplify sounds past 4 kHz, most clinical audiologists recommend open-fit devices for these sloping losses. Therefore, for the sake of generalizing to a larger clinical population, more hearing loss will be allowed past 6 kHz.

Table 3-3: Comparison between the bandwidths of open-fit hearing aids (receiver in the hearing aid and receiver in the canal) as reported by its manufacturer and as measured by an independent lab at the University of Pittsburgh

	Hearing Aid	Manufacturer's reported bandwidth	Measured bandwidth as defined as useable gain for a flat 50 dB hearing loss by Hearing aid lab at the University of Pittsburgh
Receiver in the hearing aid (over the ear)	A	200-6400 Hz	~200-4400 Hz
	B	100-7200 Hz	~200-3800 Hz
	C	100-5200 Hz	~200-3700 Hz
	D	100-5800 Hz	~200-3700 Hz
	E	200-7700 Hz	~200-3200 Hz
	F	100-5600 Hz	~200-3200 Hz
	G	100-7000 Hz	~200-3200 Hz
	H	100-7150 Hz	~200-3000 Hz
	I	200-5000 Hz	~200-2500 Hz
Receiver in the canal	J	200-7600 Hz	~200-4200 Hz
	K	140-6000 Hz	~200-4200 Hz
	L	160-6000 Hz	~200-4200 Hz
	M	100-7000 Hz	~200-3700 Hz
	N	100-8400 Hz	~200-3600 Hz
	O	200-7350 Hz	~200-3200 Hz
	P	100-7900 Hz	~200-3000 Hz
	Q	Not reported	~200-3000 Hz

A second group of 25 normal-hearing listeners (average age = 54 years, sd = 11) were recruited for the main experiment in order to test whether normal-hearing listeners are affected by the introduction of asynchronous delay. The data from this group were not compared to the hearing-impaired group, but rather serve to answer the question of whether asynchronous delays have an effect on speech intelligibility of conversational speech. These participants were

excluded if they were not native speakers of English and/or if they reported having recent middle ear pathology, otologic surgery, and/or neurologic pathology. The participants were not excluded on the basis of race or gender. These participants had audiometric thresholds less than 20 dB from 250 Hz through 8 kHz in both ears and word recognition scores on the Northwestern University monosyllabic word lists (NU-6) were within the 90% confidence limit based on their pure tone average (Dubno, Lee, Matthews, Mills, & Lam, 1995).

3.2.4 **Procedure**

All participants were screened after obtaining informed consent. Only after signing the consent form were they considered enrolled in the study. A self-report case history questionnaire (Appendix B) was administered to inquire into the participant's general health, history of hearing loss and/or middle ear disease, otologic surgery, and neurologic disorder. To determine auditory status, a standard audiometric test battery was completed. This included bilateral air and bone conduction threshold testing at the standard audiometric frequencies (ASHA, 1978), NS word recognition testing with the Auditec recording of the Northwestern University Test #6 (NU-6) word lists (Tillman & Carhart, 1966). All of the equipment used to perform the tests of auditory function was calibrated according to the appropriate ANSI standards. All testing was conducted in test rooms that meet the ANSI standards for maximum background noise.

After the audiometric data were obtained, each participant's hearing thresholds in dB HL were converted to dB SPL. First, the dB HL value was added to the calibrated ANSI S3.6 (1996) reference earphone sound pressure levels (RETSPL) to obtain the conversion from dB HL to dB SPL. This value reflects the dB SPL level when played in the average ear canal volume. Table 3-4 shows the ANSI S3.6 RETSPL values for ER-3A earphones. To obtain the SPL thresholds

specific to the participant's ear canal volume, the measured Real Ear to Coupler Difference (RECD) was added to the dB SPL. RECD is the difference in SPL between the average ear canal volume assumed by a 2cc coupler and the probe microphone measurement of SPL in the actual ear canal of a participant.

Table 3-4: ANSI S3.6 (1996) RETSPL for ER-3A earphone

Frequency (Hz)	dB SPL in HA-2 (with rigid tube)
250	14
500	5.5
1000	0
1500	2
2000	3
3000	3.5
4000	5.5
6000	2
8000	0

The RECD was measured by connecting the audiometric ER-3A earphone (the earphone used during audiometric testing) to a HA-2 coupler in Frye 7000 Hearing Aid Analyzer. After plugging the input jack of the ER-3A earphones into the Frye 7000, the system generated a frequency sweep signal to capture the output of the ER-3A earphone across frequencies. Next, the Frye 7000's probe microphone was inserted in the participant's ear canal. The ER-3A earphone was inserted into the participant's ear over the probe microphone, taking care that the probe tip was past the depth of the insert earphone. Again, the system generated a pure-tone sweep while the probe microphone measured the output of the earphone in the ear. The difference in dB between these two measures is the RECD. Table 3-5 shows how the RECD is added to a participant's thresholds in dB HL and the RETSPL value to obtain that participant's

thresholds in dB SPL. Calculation of each individual's hearing threshold in dB SPL was the first step in ensuring audibility of the stimuli used for the experiment.

Table 3-5: An example the calculation of a participant's hearing threshold in dB SPL

Frequency	250	500	1000	2000	3000	4000	6000	8000
Hearing threshold in dB HL	5	5	20	35	40	45	50	65
+ RETSPL	14	5.5	0	3	3.5	5.5	2	0
+ RECD =	-1	-2	1	5	6	10	7	6
Hearing threshold in dB SPL	18	8.5	21	43	49.5	60.5	59	71

The presentation level of the stimuli was determined in the following manner. First, prior to the presence of any participant, the "original calibration noise" soundfile was modified with the MATLAB algorithm that applied 18 dB of gain above 2 kHz. This file was saved as "gain calibration noise" on the computer. It was used to verify the audibility of high frequency signals for each hearing-impaired participant in the main-experiment. Also, the audiometer attenuator was adjusted so that the time-weighted output of the ER-2 earphone peaked at 60 dB SPL on probe microphone measurements for the "original calibration noise" as described in the pre-experiment.

During calibration for the hearing-impaired listeners, a Frye Fonix 7000 probe microphone was threaded into the participant's ear canal. The participant's hearing thresholds in dB SPL were entered in the Fonix system. The real ear system was set up so that it was measuring an SPL-o-gram with the stimulus turned off. An ER-2 earphone was inserted into the participant's ear over the probe microphone. For the hearing-impaired listeners, the "gain

calibration file" was played at the audiometer attenuator setting determined earlier. The Fonix 7000 displayed the output of the ER-2 earphone playing the "gain calibration file". The output measure was examined, verifying that the output at all frequencies from 250-4 kHz closely matched NAL-R target for 60 dB input. The audiometer attenuator was adjusted in 2 dB steps until audibility was achieved. Figure 3-9 displays the average NAL-R target for 60 dB input and the average output of the earphone as measured by the probe microphone. The final attenuator setting was the presentation level for the listening task. The probe microphone was then removed from the hearing-impaired participant's ear, and the same calibration measurements were made on the opposite ear. Once the presentation levels were determined, the probe microphone was removed from the participant's ear, and the experimental listening task began.

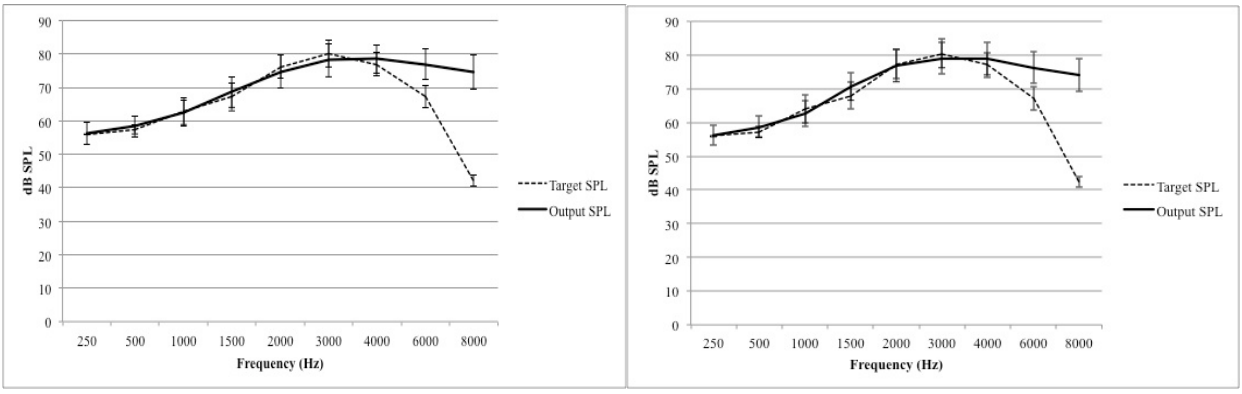


Figure 3-9: Measured Real Ear SPL Output in response to the "gain calibration filr" and NAL-R target for 60 dB input for the (a) right ear and (b) left ear

Eight-hundred sentences were presented randomly to each participant using SuperLab 4.0 (Cedrus Corporation) software on a computer connected to a Grason-Stadler 61 audiometer. The 800 processed sentences represent the 4 conditions of simulated high frequency delay. Each set of 200 conversational sentences had asynchronous delay values of 0 ms (control condition), 4 ms, 8 ms, and 32 ms. These 800 sentences were presented randomly to 25 hearing-impaired participants and 25 normal-hearing listeners, thereby reducing a condition presentation order

effect. The possibility of a learning effect from repeating the same 200 sentences for each of the four conditions is slim as Kalikow and colleagues (1977) found negligible learning effects for the low predictability sentences. Regardless, the data were analyzed to ensure that there was not a learning effect from sentence repetition.

Each participant was seated in a sound treated room with ER-1 insert earphones placed in each ear. A microphone mounted in front of the participant recorded his or her responses. The participant was instructed to "Listen carefully to the following sentences. Repeat each sentence immediately after you hear it. Please speak clearly into the microphone. The listening task is divided into 16 blocks of 50 sentences. You will be given a break between each block. If you need a break sooner for any reason, please speak up and let me know." The participants were asked to repeat the entire sentence to prevent listeners from using all of their attention resources on the final key word.

3.2.5 **Data Analysis**

The recorded data from each participant in the main-experiment were presented to a normal-hearing listener to judge the last word (key word) being spoken in each of the 800 sentences. The judge was given the answer keys containing the 800 correct key words in the order of presentation for each participant. The judge marked whether the participant got each key word correct or incorrect. If the participant was incorrect, the judge also wrote down the word that he/she heard the participant say. The judge was blinded to the conditions, thus removing tester bias. The percent correct key word identification and the identification errors were extracted from the recordings in this manner for all of the sentences spoken by each participant.

4.0 RESULTS

4.1 PRE-EXPERIMENT: CLEAR-SPEECH VS. CONVERSATIONALLY SPOKEN SPEECH

The conversationally spoken, clearly spoken, and the original recording of each of the 200 r-SPIN sentences were analyzed in Adobe Audition to calculate speaking rate in words per minute (wpm) and syllable per second (syl/s), vowel duration of the final key word, and the voiceless voice onset time (VOT) of the key word. All data were analyzed with a paired t-test for significant differences and the descriptive data are listed in Table 4-1

Table 4-1: Acoustic differences between Clear, Conversational (same male speaker) and Original recordings (different male speaker) of the R-SPIN LP sentences (Bilger et al., 1984).

	Clear		Conversational		Original	
	mean	SD	mean	SD	mean	SD
Words per min	144	23.5	270	38.9	190	23.2
Syllable per sec	3.05	0.46	5.67	0.72	4.03	0.47
Vowel duration (ms)	242	80	180	92	230	61
Voiceless VOT (ms)	95	12	33	16	97	16

As displayed in Table 4-1, the speaking rate of the conversational recordings was significantly faster than both the clear recordings and original recordings ($p<0.01$). The

conversational recordings were nearly twice as fast as the clear speech recordings (270 wpm vs. 144 wpm). Also, the mean voiceless VOT for the conversational recordings was significantly shorter than both the clear and original recordings ($p<0.01$) being roughly one third of that found for the clear recordings (95 ms vs. 33 ms). The mean voiceless VOT for the clear and original recordings did not significantly differ ($p<0.01$). Figure 4-1 displays a spectrogram demonstrating the difference in VOT of the word "tanks" between clear (top; 62 ms) and conversational (bottom; 27 ms) speech. As with many of the conversationally spoken words with a voiceless plosive in the word-initial position, the burst and aspiration of the initial consonant /t/ in tank was reduced.

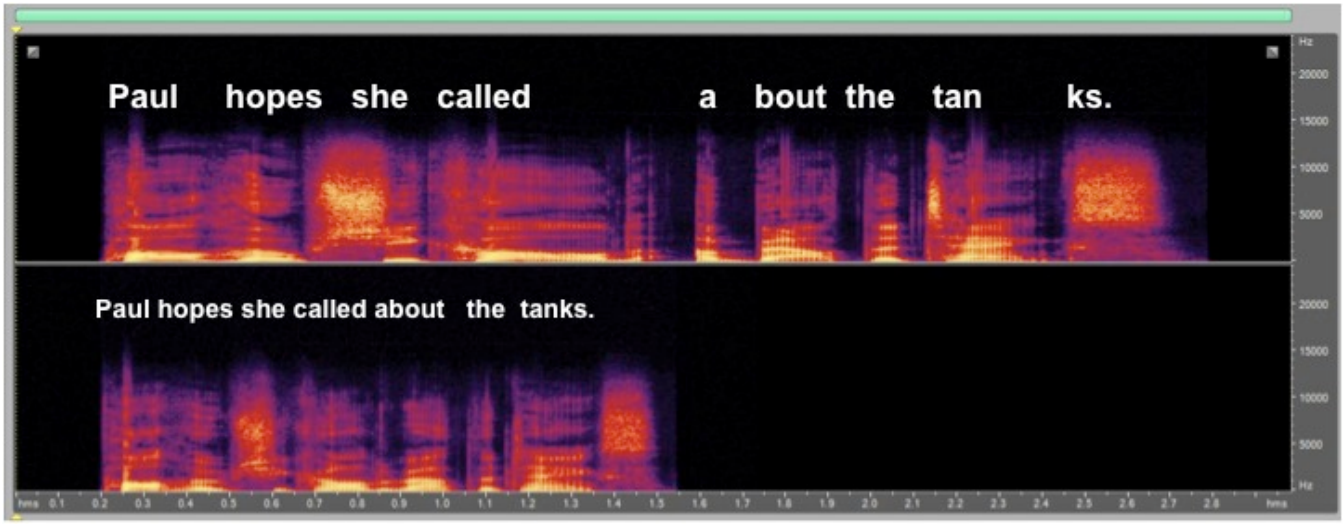


Figure 4-1: Spectrogram demonstrating the difference in VOT of the word "tanks" between clear (top; 62 ms) and conversational (bottom; 27 ms) speech.

For plosives in word-final position, the conversational recordings exhibited weaker bursts and releases as compared to the clear speech recording. Figure 4-2 and Figure 4-3 show spectrograms for two sentences with plosives in the word-final position of the final key word. In Figure 4-2, the burst in the /p/ of the word "sheep" in the conversational recording is weaker and

the aspiration is shorter than its clear speech counterpart. However, in Figure 4-3, the release of the /p/ in “sap” is omitted in the conversationally spoken version of the sentence.

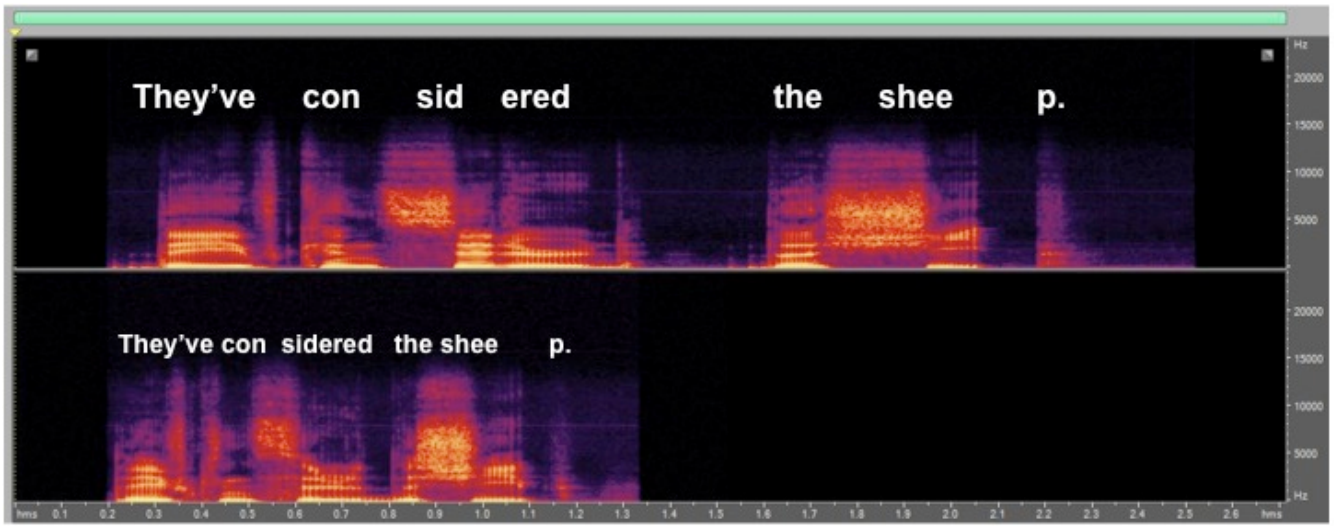


Figure 4-2: Spectrogram demonstrating the intensity and durational differences in the release of the final plosive /p/ of the word “sheep” between clear (top) and conversational speech (bottom).

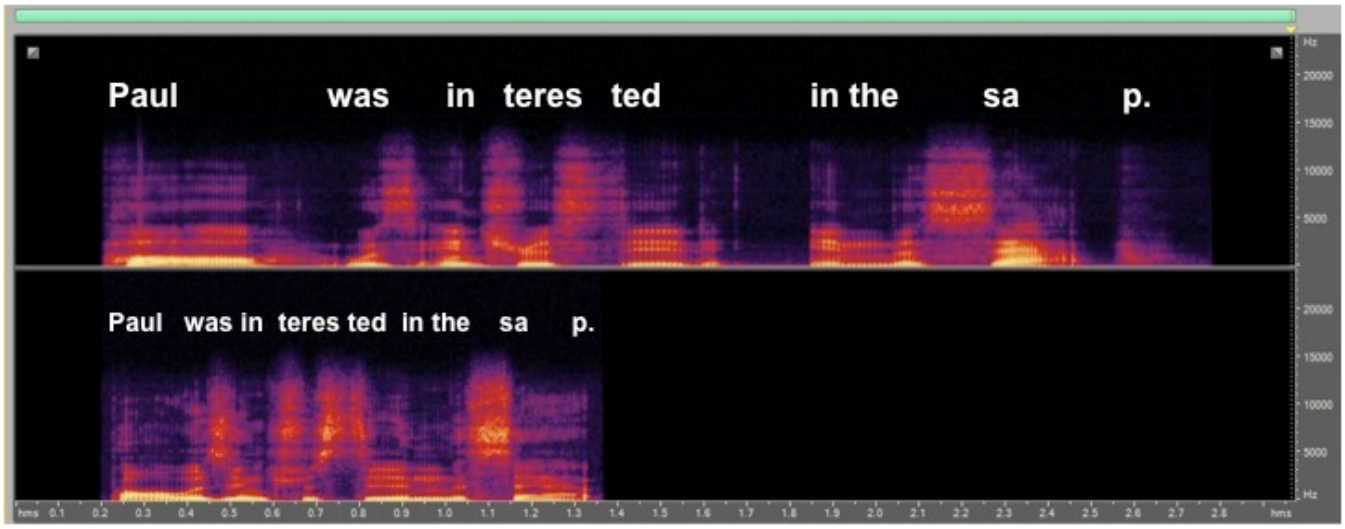


Figure 4-3: Spectrogram demonstrating the intensity and durational differences in the release of the final plosive /p/ of the word “sap” between clear (top) and conversational speech (bottom). Notice the omission of the final plosive release.

Vowel duration was significantly shorter in the conversationally recorded speech than both the clear speech and the original recordings ($p < 0.01$). The spectrogram in Figure 4-4 demonstrates the vowel duration difference of the key word "sand" between the clear and conversational recordings. Formant transitions were shorter as a consequence of the conversational speech's faster articulation rate and shorter vowel durations as shown in Figure 4-5.

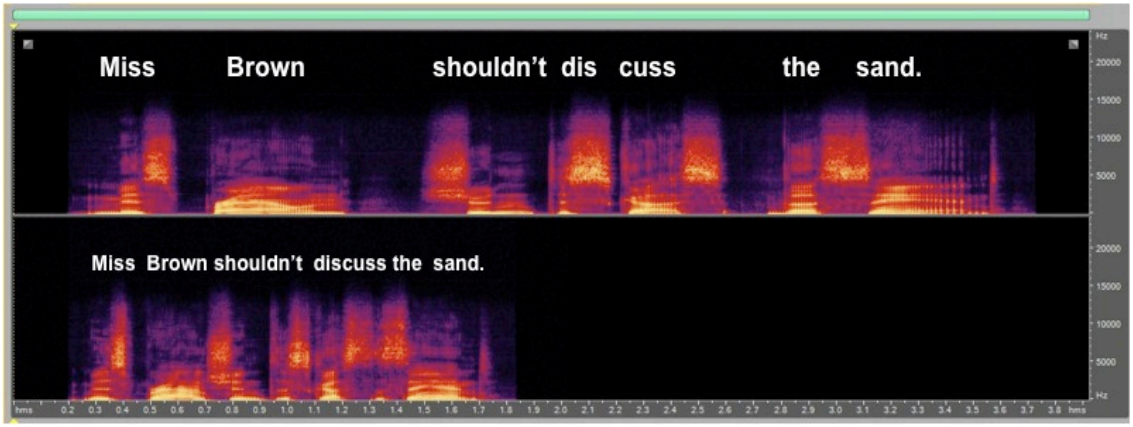


Figure 4-4: Spectrogram demonstrating the vowel duration difference of the word "sand" between clear (top, 379 ms) and conversational speech (bottom, 200 ms)

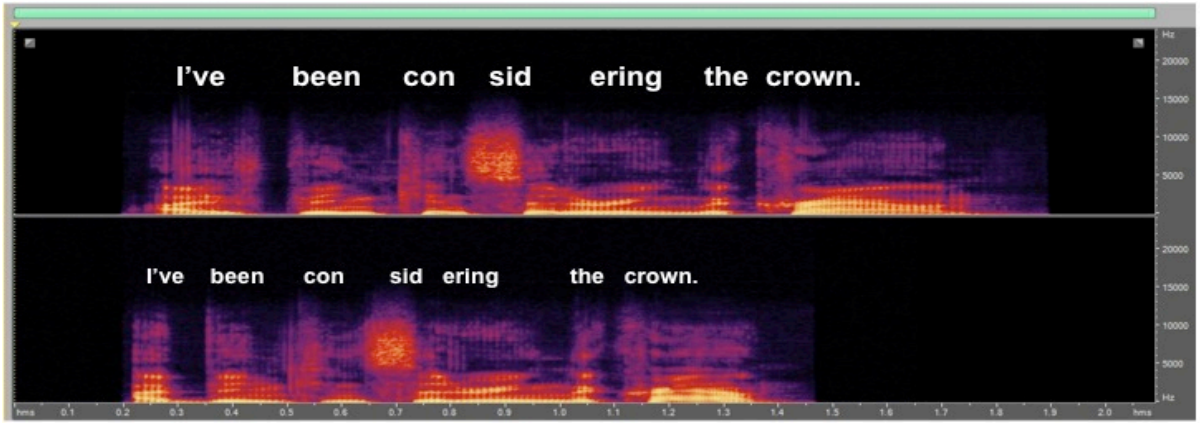


Figure 4-5: Spectrogram demonstrating the difference in the formant transitions for the /r/ in crown between clear (top) and conversational (bottom) speech

4.2 PRE-EXPERIMENT: INTELLIGIBILITY OF CONVERSATIONAL R-SPIN RECORDINGS

The data from the 15 young normal-hearing participants of the Pre-Experiment were plotted as percent correct final key-word identification. The average score for final key-word identification was 98% correct with a standard deviation of 1.33. Therefore, the conversationally spoken recorded r-SPIN sentences were acoustically different from clear speech and the original r-SPIN recording yet intelligible by normal-hearing listeners.

4.3 MAIN EXPERIMENT: EFFECT OF PRESENTATION ORDER

The data from both the 25 normal hearing and the 25 hearing-impaired participants were plotted separately as percent correct key word identification as a function of presentation order for each set of 200 sentences. The data sets were each analyzed with one-way ANOVA to see if there is a significant main effect of presentation order. Due to the previous reports of negligible learning effect for the low predictability sentences and the randomized experimental design significant results were not anticipated. For the normal-hearing group, the results of the ANOVA found a significant main effect for presentation order ($F(3, 72) = 8.695$, $p < 0.05$). Post hoc analysis using a Bonferroni correction factor revealed that only the score from the first presentation was significantly different at $\alpha < 0.05$ from the following three presentations. A Bonferroni was applied in order to control for a type I error (incorrectly identifying an order effect when there is in fact no order effect). Figure 4-6 displays the normal-hearing group mean score and standard deviation for each presentation order. The average score for presentation order ranged from

95.86 at the first presentation to 97.12 at the third presentation. Although the difference in the percent correct score between the first presentation and the subsequent presentations of each sentence is minimal (roughly a 1.25 percentage point difference), the difference was found to be significant at $p < 0.05$ due to the small variance of the data. However, given the low predictability nature of key word in each sentences, the fact in that all of the sentences and experimental conditions were randomly presented to each listener in one block, and the high accuracy in identification (ceiling effect) of key words in sentence by normal hearing listeners, it was decided that the learning effect was negligible.

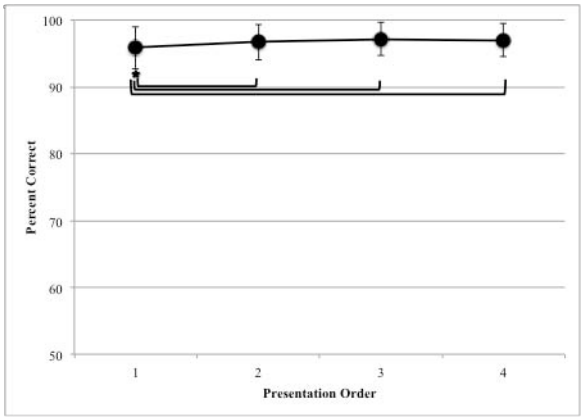


Figure 4-6: The average percent correct key word identification of the 25 normal hearing participants plotted as a function of presentation order (combined across all delay conditions). Only the first presentation was significantly different from the other presentations at $\alpha < 0.05$.

The results of the ANOVA for the hearing-impaired group also revealed a significant main effect for presentation order ($F(3, 72) = 5.103$, $p < 0.05$). Post hoc analysis using a Bonferroni correction factor revealed that only the score between the first and last presentation was significantly different at $\alpha < 0.05$. Figure 4-7 displays the hearing-impaired group mean score and standard deviation for each presentation order. The average score for presentation order ranged from 87.26 at the first presentation to 89.16 at the final presentation. Again, given

the previous research and the consideration that all of the experimental conditions were presented in random order, it was decided that there was negligible learning effect among the hearing-impaired participants.

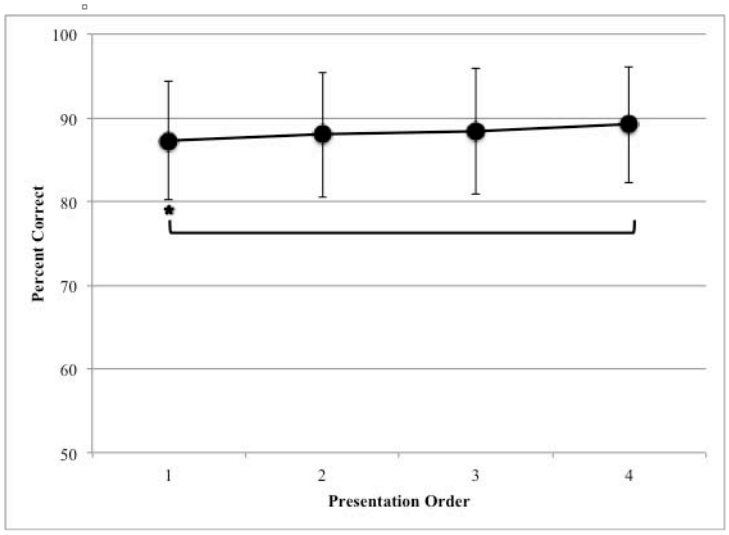


Figure 4-7: The average percent correct key word identification of the 25 hearing-impaired participants plotted as a function of presentation order (combined across all delay conditions). Only the difference between the first and last presentation was significantly different at $\alpha < 0.05$.

4.4 MAIN EXPERIMENT: EFFECT OF ASYNCHRONOUS DELAY FOR NORMAL-HEARING LISTENERS

The data from the group of 25 normal-hearing listeners were plotted as percent correct key word identification as a function of the four conditions (0 ms, 5 ms, 10, ms, and 30 ms). Repeated measures ANOVA (subjects by delay condition) was performed to test for the main effect of asynchronous delay condition. In a repeated measures design, the chance of making a type I error (erroneously declaring a difference) increases with each comparison. In order to control the overall type I error rate, it was necessary to decrease the value of alpha for each of the four

comparisons. However as the type I error rate is minimized, the chances of making a type II error increases. It is important to examine the severity of each error type prior to making the decision to control the type I error with the use of a Bonferroni to adjust the level of alpha for each comparison. Table 4-2 lists the possible true outcomes of the study and the errors associated with each outcome. The severity of the error is listed in the final column. The severity of the type II error (incorrectly declaring no difference between varying amounts of asynchronous delay) is much higher than that of the type I error. In order to tightly control the type II error rate, the Bonferroni correction factor was not applied.

Table 4-2: Analysis of the severity of the error for the research outcome

Possible True outcomes	Consequence	Potential Errors		Erroneous conclusion	Severity of error
		Type I	Type II		
% correct on SPIN test without delay = % correct on SPIN test with delay	Suggest that asynchronous delays up to 30 ms have no effect on hearing impaired listener's ability to understand conversational speech	Incorrectly declared a difference between SPIN scores	Cannot make type II error when the true outcome is no difference between scores	Type I error: Asynchronous delay has an effect on hearing impaired listeners speech perception and needs to be controlled in DSP circuits.	Severity is **low** because it does not matter whether delays are controlled or not.
% correct on SPIN test without delay ≠ % correct on SPIN test with delay	Suggest that asynchronous delays affect SPIN scores and the amount of delay needs to be tightly controlled in DSP circuits.	Cannot make a type I error when the true outcome reports a difference between scores.	Incorrectly declared no difference between groups.	Type II error: Asynchronous delay up to 30 ms has no effect on hearing impaired listeners speech perception.	Severity is **high** because asynchronous delays do need to be controlled. The delay has a negative effect on listeners' speech perception.

Figure 4-8 displays the results of the effect of delay condition on key word identification by normal-hearing listeners. The results of the ANOVA for the normal hearing group revealed a significant main effect for delay condition ($F(3, 72) = 2.852$, $p = 0.043$). Post hoc analysis revealed that only the score between the 4 ms and the 32 ms condition was significantly different

at $\alpha < 0.05$. The average score ranged between 96.32 (32 ms condition) and 97.14 (4 ms condition). However, this slight difference was found to be statistically significant due to the low variability among scores.

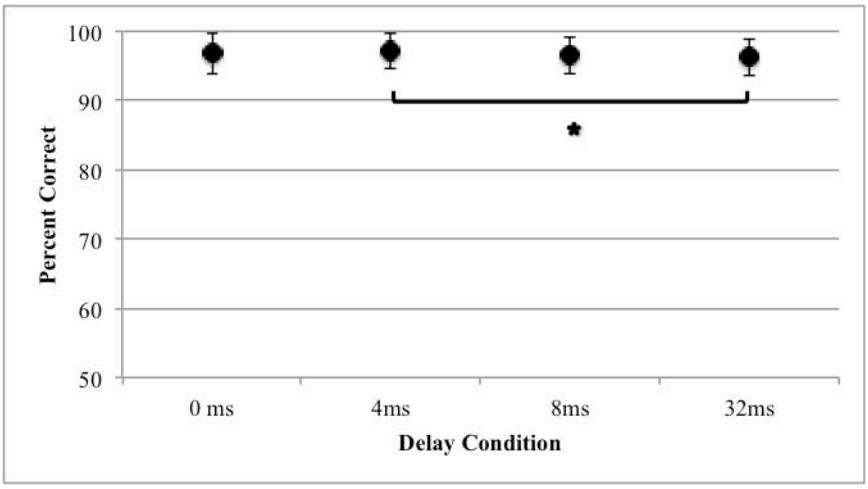


Figure 4-8: The average percent key word identification by normal-hearing listeners as a function of spectrally asynchronous delay. Only the difference between the 4 ms and the 32 ms condition was found to be significant at $\alpha < 0.05$.

4.5 MAIN EXPERIMENT: EFFECT OF ASYNCHRONOUS DELAY FOR HEARING-IMPAIRED LISTENERS

The data from the group of 25 hearing-impaired listeners were plotted as percent correct key word identification as a function of the four conditions (0 ms, 4 ms, 8, ms, and 32 ms). Repeated measures ANOVA (subjects by delay condition) was performed to test for the main effect of asynchronous delay condition. As with the normal-hearing data, a Bonferroni correction factor was not applied in order to control the type II error rate. Figure 4-9 displays the effect of each delay condition on key word identification for the group of hearing-impaired listeners. The

average values were 89.74%, 88.66%, 88.36%, and 85.36% for the 0, 4, 8, and 32 ms conditions respectively. Appendix E displays the individual data for the hearing-impaired listeners with the individuals with a greater than 5% decrease in score with increasing delay values highlighted. A main effect for the asynchronous delay condition was found ($F(3, 72) = 19.788$, $p < 0.05$). A post-hoc analysis revealed that only the difference between the 4 ms and the 8 ms condition was not significant; all other differences were significant at $\alpha < 0.05$.

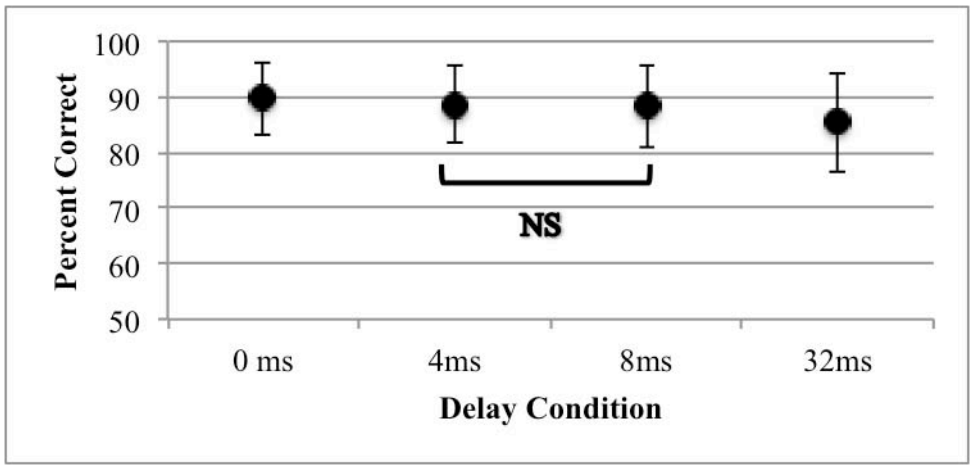


Figure 4-9: The effect of delay condition on key-word identification for hearing-impaired listeners. All differences between conditions with the exception of the difference between 4 ms and 8 ms were found to be significant at $\alpha < 0.05$.

5.0 DISCUSSION

The results of the Pre-Experiment determined that the conversationally recorded r-SPIN low predictability sentences were acoustically different from both clear speech and the original commercially available r-SPIN recordings (Bilger et al., 1984, Cosmos Distributing, Inc.), yet intelligible by normal-hearing listeners. These conversationally spoken recordings were used to better represent "real world" speech acoustics and to reflect interactions between signal processing algorithms and the rapidly changing speech acoustics.

The Main experiment used these 200 conversationally recorded r-SPIN sentences repeated for each asynchronous delay condition (0 ms, 4 ms, 8 ms, and 32 ms). Although all 800 sentences were presented in random order to each participant and previous research reported no learning effect of the low predictability r-SPIN sentences (Kalikow et al., 1977), the data were analyzed to determine if there was an effect of presentation order. Although there was a statistically significant difference for both the normal-hearing and hearing-impaired group of listeners for which the key word identification of the first presentation was slightly poorer than the last presentation, the learning effect was deemed negligible due to the randomized experimental design, the small effect size, and previous research finding minimal learning effect with low-predictability sentences.

5.1 NORMAL-HEARING LISTENERS PERCEPTION OF SPECTRALLY ASYNCHRONOUS DELAYS

Normal-hearing listeners were essentially unaffected by spectrally asynchronous delays up to 32 ms. The only statistically significant difference found was between the 4 ms and the 32 ms delay condition, and the difference was roughly equal to an additional 1.5 correct key word identification out of 200 sentences or a less than a 1 percentage point change in score. It appears that normal-hearing listeners were either able to ignore the spectro-temporal distortion and/or use other acoustic cues to correctly identify the key word. For example, previous research has found that normal-hearing listeners give greater perceptual weight to rapid formant transitions than to other acoustic cues when identifying consonants (Lindholm et. al., 1988; Hedrick & Jesteadt, 1996; Hedrick & Younger, 2001, 2007). With the delays occurring above 2 kHz, the F1 formant transitions were unaltered, while most of the F2 and F3 formant transitions were left intact albeit delayed in time. Normal-hearing listeners may have been able to use these formant transitions and other redundant acoustic cues such as spectro-intensity cues to correctly identify key words despite the delay of signals above 2 kHz up to 32 ms. Another consideration would be that the normal-hearing listeners were able to suppress or ignore the acoustic delay above 2 kHz to take advantage of the "open-ear" or the unaltered non-hearing aid pathway of sound.

5.2 HEARING-IMPAIRED LISTENERS PERCEPTION OF SPECTRALLY ASYNCHRONOUS DELAYS

Hearing-impaired listeners performed well on the key word identification of the r-SPIN LP conversationally spoken sentences with an average score for the control condition (zero delay) of 89.74 with a standard deviation of 6.53. This score isn't surprising considering that the task was performed in quiet and that audibility was tightly controlled by the inclusion criteria (no greater than a moderate loss from 2-4 kHz) and through the application of appropriate gain to meet the NAL-NL1 prescribed SPL output target based upon each participant's hearing thresholds. Hearing-impaired listeners were adversely affected by a 32 ms delay to the spectrum above 2 kHz. These results confirm previous research that found that perceptual differences were found with frequency dependent delay greater than 24 ms (Stone and Moore, 2003). The difference between the control condition and the shorter delay conditions of 4 and 8 ms was also statistically significant, albeit it was only a slight degradation in performance (an average difference of only 1 percentage point between the conditions). It appears that once audibility is accounted for, hearing-impaired listeners are tolerant of spectrally asynchronous delays up to 8 ms, but then are adversely affected by 32 ms in quiet listening situations. Given that there is a negative consequence of spectrally asynchronous delay on the speech perception abilities of hearing-impaired listeners, hearing aid manufacturers should be conscious of the speed of hearing aid digital processing.

It is possible that the hearing-impaired listeners, like the normal-hearing listeners relied somewhat on the formant transitions that were held intact despite the delay conditions for correct key word identification. When the individual data are examined (Appendix E), it appears that some listeners are tolerant of the asynchronous delay similar to that of normal-hearing listeners.

When the hearing threshold data of these listeners are examined (Appendix D), it seems that the hearing-impaired listeners with milder losses are more tolerate of the delay conditions. As for the other listeners who were affected by the asynchronous delay, perhaps they rely on other speech perceptual cues. Studies have shown that hearing-impaired listeners rely less on formant transitions and more on spectral shape and temporal properties of speech for phonemic identification (Lindholm et al., 1988; Hedrick and Younger 2007). Listeners with mild-moderate sensorineural loss can perceive timing cues such as voice onset time (VOT) and spectro-temporal cues such as envelope onset asynchrony (EOA) similar to those with normal hearing (Johnson et al., 1984; Ortmann et al., 2010). If hearing-impaired listeners are relying on these particular spectro-temporal cues for phonemic and subsequently word identification, then their perceptual performance is more likely to be affected by spectrally asynchronous delays as these delays blur the onset of voiceless plosives. This hypothesis was supported by significantly poorer performance in the conditions with spectrally asynchronous delay by hearing-impaired listeners, but not by normal-hearing listeners.

Hedrick and Younger (2007) demonstrated that hearing-impaired listeners rely even less on formant transitions when performing in conditions with background noise or reverberation. It would be assumed that in background noise, these listeners would rely more on spectro-temporal cues such as VOT or EOA in these conditions. It would be of interest to measure the intelligibility of the conversational sentences with the same delay conditions in background noise and/or reverberation. It is hypothesized that because listeners are relying more on gap and spectro-temporal distinctions, they would be more susceptible to shorter asynchronous delay values.

APPENDIX A

REVISED SPEECH PERCEPTION IN NOISE (R-SPIN) LOW PREDICTABILITY SENTENCES

List 1
1. Miss White won't think about the CRACK.
2. He wouldn't think about the RAG.
3. The old man talked about the LUNGS.
4. I was considering the CROOK.
5. Bill might discuss the FOAM.
6. Nancy didn't discuss the SKIRT.
7. Bob has discussed the SPLASH
8. Ruth hopes he heard about the HIPS.
9. She wants to talk about the CREW.
10. They had a problem with the CLIFF.
11. You heard Jane called about the VAN.
12. We could consider the FEAST.
13. Bill heard we asked about the HOST.
14. I had not thought about the GROWL.
15. He should know about the HUT.
16. I'm glad you heard about the BEND.
17. You're talking about the POND.
18. Nancy had considered the SLEEVES.
19. He can't consider the CRIB.
20. Tom discussed the HAY.
21. She's glad Jane asked about the DRAIN.
22. Bill hopes Paul heard about the MIST.
23. We're speaking about the TOLL.
24. We spoke about the KNOB.
25. I've spoken about the PILE.

List 2
1. Miss Black thought about the LAP.
2. Miss Black would consider the BONE.
3. Bob could have known about the SPOON.
4. He wants to talk about the RISK.
5. He heard they called about the LANES.
6. She has known about the DRUG.
7. I want to speak about the CRASH.
8. I should have considered the MAP.
9. Ruth must have known about the PIE.
10. The man should discuss the OX.
11. They heard I called about the PET.
12. Bill cannot consider the DEN.
13. She hopes Jane called about the CALF.
14. Jane has a problem with the COIN.
15. Paul hopes she called about the TANKS.
16. The girl talked about the GIN.
17. Mary should think about the SWORD.
18. Ruth could have discussed the WITS.
19. You had a problem with a BLUSH.
20. We have discussed the STEAM.
21. Tom is considering the CLOCK.
22. You should not speak about the BRAIDS.
23. Peter should speak about the MUGS.
24. He has a problem with the OATH.
25. Tom won't consider the SILK.

List 3

1. Mr. White discussed the CRUISE.
2. Miss White thinks about the TEA.
3. He is thinking about the ROAR.
4. She's spoken about the BOMB.
5. You want to talk about the DITCH.
6. We're discussing the SHEETS.
7. Betty has considered the BARK.
8. Tom will discuss the SWAN.
9. You'd been considering the GEESE.
10. They were interested in the STRAP.
11. He could discuss the BREAD.
12. Jane hopes Ruth asked about the STRIPES.
13. Paul spoke about the PORK.
14. Mr. Smith thinks about the CAP.
15. We are speaking about the PRIZE.
16. Harry had thought about the LOGS.
17. Bob could consider the POLE.
18. Ruth has a problem with the JOINTS.
19. He is considering the THROAT.
20. We can't consider the WHEAT.
21. The man spoke about the CLUE.
22. David has discussed the DENT.
23. Bill heard Tom called about the COACH.
24. Jane has spoken about the CHEST.
25. Mr. White spoke about the FIRM.

List 4

1. Mary had considered the SPRAY.
2. The woman talked about the FROGS.
3. Miss Brown will speak about the GRIN.
4. Bill can't have considered the WHEELS.
5. Mr. Smith spoke about the AID.
6. He hears she asked about the DECK.
7. You want to think about the DIME.
8. You've considered the SEEDS.
9. Ruth's grandmother discussed the BROOM.
10. Miss Smith considered the SCARE.
11. Peter has considered the MAT.
12. The old man considered the KICK.
13. Paul could not consider the RIM.
14. I've been considering the CROWN.
15. We've spoken about the TRUCK.
16. Mary could not discuss the TACK.
17. Harry might consider the BEEF.
18. We're glad Bill heard about the ASH.
19. Nancy should consider the FIST.
20. They did not discuss the SCREEN.
21. The old man thinks about the MAST.
22. Paul wants to speak about the BUGS.
23. You're glad she called about the BOWL.
24. Miss Black could have discussed the ROPE.
25. I hope PAUL asked about the MATE.

List 5

1. Betty knew about the NAP
2. The girl should consider the FLAME.
3. They heard I asked about the BET.
4. Mary knows about the RUG.
5. He was interested in the HEDGE.
6. Jane did not speak about the SLICE.
7. Mr. Brown can't discuss the SLOT.
8. Paul can't discuss the WAX.
9. Miss Brown shouldn't discuss the SAND.
10. David might consider the FUN.
11. She wants to speak about the ANT.
12. He hasn't considered the DART.
13. We've been discussing the CRATES.
14. We've been thinking about the FAN.
15. Jane didn't think about the BROOK.
16. Betty can't consider the GRIEF.
17. Harry will consider the TRAIL.
18. Tom is talking about the FEE.
19. Tom had spoken about the PILL.
20. Tom has been discussing the BEADS.
21. Tom could have though about the SPORT.
22. Mary can't consider the TIDE.
23. He hopes Tom asked about the BAR.
24. We could discuss the DUST.
25. Paul hopes we heard about the LOOT.

List 6

1. You were considering the GANG.
2. The boy had considered the MINK.
3. He wants to know about the RIB.
4. She might have discussed the APE.
5. The old woman discussed the THIEF.
6. You were interested in the SCREAM.
7. We hear they asked about the SHED.
8. I haven't discussed the SPONGE.
9. Ruth will consider the HERD.
10. The old man discussed the DIVE.
11. The class should consider the FLOOD.
12. I'm talking about the BENCH.
13. Paul has discussed the LAMP.
14. You knew about the CLIP.
15. She might consider the POOL.
16. Bob was considering the CLERK.
17. The man knew about the SPY.
18. The class is discussing the WRIST.
19. They hoped he heard about the RENT.
20. Mr. White spoke about the JAIL.
21. Miss Brown might consider the COAST.
22. Bill didn't discuss the HEN.
23. The boy might consider the TRAP.
24. He should consider the ROAST.
25. Miss Brown spoke about the CAVE.

List 7

1. We're considering the BROW.
2. I am thinking about the KNIFE.
3. They've considered the SHEEP.
4. He's glad we heard about the SKUNK.
5. The girl should not discuss the GOWN.
6. Mr. Smith knew about the BAY.
7. We did not discuss the SHOCK.
8. Mr. Black has discussed the CARDS.
9. Mr. Black considered the FLEET.
10. We are considering the CHEERS.
11. Sue was interested in the BRUISE.
12. Miss. Smith couldn't discuss the ROW.
13. I am discussing the TASK.
14. Paul should know about the NET.
15. Miss Smith might consider the SHELL.
16. You cannot have discussed the GREASE.
17. I did not know about the CHUNKS.
18. I should have known about the GUM.
19. Mary hasn't discussed the BLADE.
20. Ruth has discussed the PEG.
21. We have not thought about the HINT.
22. The old man discussed the YELL.
23. They're glad we heard about the TRACK.
24. The boy can't talk about the THORNS.
25. Bill won't consider the BRAT.

List 8

1. Bob heard Paul called about the STRIPS.
2. Paul has a problem with the BELT.
3. They knew about the FUR.
4. We're glad Ann asked about the FUDGE.
5. Jane was interested in the STAMP.
6. Miss White would consider the MOLD.
7. They want to know about the AIM.
8. The woman discussed the GRAIN.
9. You hope they asked about the VEST.
10. We should have considered the JUICE.
11. The woman considered the NOTCH.
12. The woman knew about the LID.
13. Jane wants to speak about the CHIP.
14. Bob should not consider the MICE.
15. Ruth hopes she called about the JUNK.
16. I can't consider the PLEA.
17. Paul was interested in the SAP.
18. He's glad you called about the JAR.
19. Miss Smith knows about the TUB.
20. The man could not discuss the MOUSE.
21. Ann was interested in the BREATH.
22. You're glad they heard about the SLAVE.
23. The man could consider the SPOOL.
24. Peter knows about the RAFT.
25. She hears Bob asked about the CORK.

APPENDIX B

CASE HISTORY FORM

Read the following questions and circle the appropriate answer:

1. How old are you? _____

2. Are you in good general health? Yes No
   If you answered no, please explain your medical conditions:

_____

_____

3. Do you feel that you have a hearing loss? Yes No
   If yes, do you feel that one ear is better than the other? Yes No
   If so, which ear is your better ear? Right Left

4. Are you a native speaker of English? Yes No

5. Have you had any recent ear infections, drainage, or pain in your ears? Yes No
   If you answered yes, please give the date of the ear infection and when and how it was resolved: _____

_____

6. Have you had any surgeries performed on your ears? Yes No
   If yes, please explain: _____

_____

7. Have you ever been diagnosed with any neurologic disorder (i.e., brain tumor, stroke, Parkinson's disease, etc.)? Yes No
   If yes, please explain: _____

_____

APPENDIX C
FIR FILTER REPSONSE

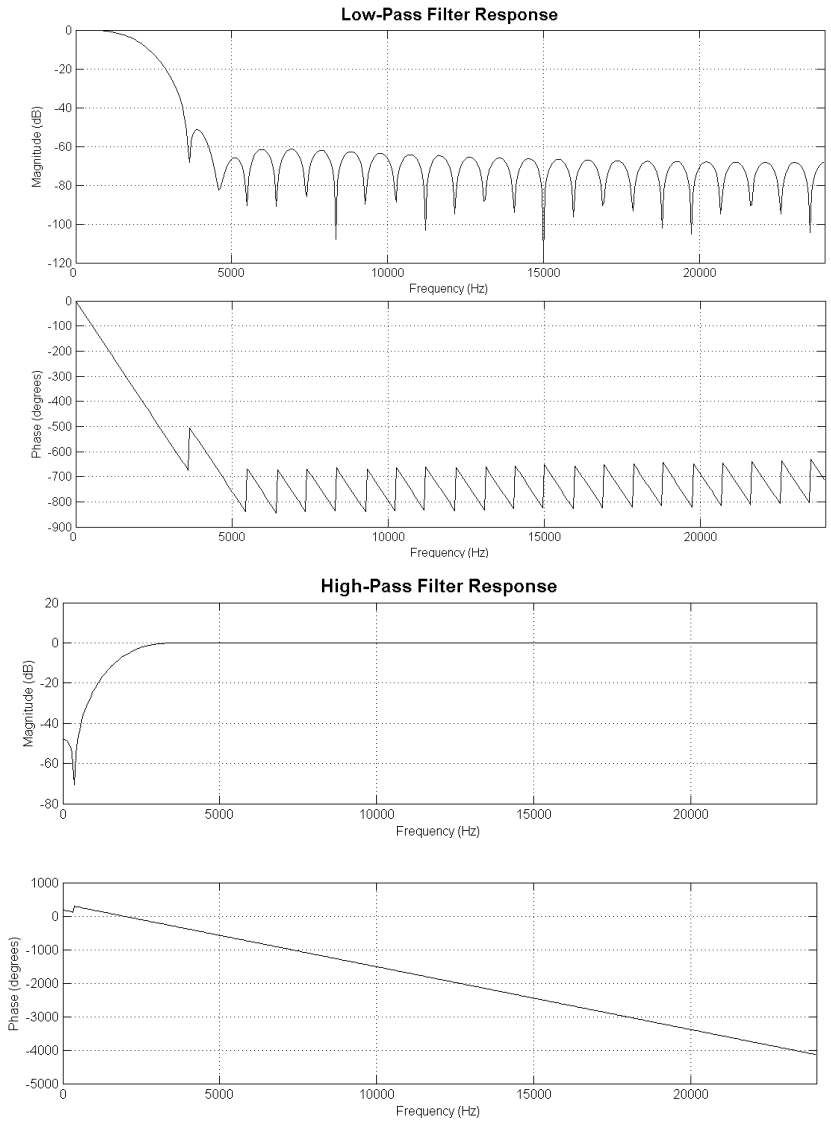


Figure 5-1: Graphs depict the magnitude and phase response for the FIR high pass (top) and low pass (bottom) filter used to create the delay conditions.

APPENDIX D

DEMOGRAPHIC INFORMATION FOR THE PARTIPANTS OF THE MAIN EXPERIMENT

Table 5-1: Main experiment normal-hearing listeners' demographics (age and hearing thresholds)

			Frequency in Hz							
	Age	Ear	250	500	1000	2000	3000	4000	6000	8000
NH1	21	RE	15	10	10	10	5	5	5	5
		LE	15	15	10	15	10	5	5	10
NH2	46	RE	15	15	10	5	15	10	10	0
		LE	10	5	10	5	15	10	10	0
NH3	61	RE	10	15	15	15	15	15	15	15
		LE	10	15	15	15	15	15	15	15
NH4	64	RE	20	15	5	5	10	10	20	20
		LE	15	10	10	15	20	20	10	15
NH5	61	RE	20	15	15	10	5	10	15	15
		LE	15	15	20	15	15	15	10	15
NH6	61	RE	10	10	15	10	5	20	20	20
		LE	10	10	10	15	5	20	20	15
NH7	65	RE	5	5	5	10	10	10	20	15
		LE	5	5	0	10	20	20	20	20
NH8	59	RE	10	10	15	15	20	20	20	25
		LE	10	10	20	15	15	20	20	20
NH9	65	RE	15	15	15	15	15	20	15	20
		LE	15	15	15	15	15	15	15	10
NH10	62	RE	10	15	15	10	15	10	10	10
		LE	15	15	15	15	10	15	15	10
NH11	63	RE	15	10	15	15	15	15	15	20
		LE	15	10	15	5	15	15	10	15
NH12	54	RE	15	10	10	10	5	5	10	10
		LE	10	10	10	10	5	10	10	5
NH13	55	RE	15	10	5	10	15	15	15	15
		LE	10	10	5	10	15	10	15	10
NH14	55	RE	10	10	5	0	10	10	10	15
		LE	10	5	5	0	5	10	10	15
NH15	54	RE	10	10	15	15	20	15	15	20
		LE	15	10	15	10	20	15	15	15
NH16	56	RE	15	10	10	10	5	5	10	10
		LE	15	15	10	5	5	5	10	15
NH17	61	RE	20	15	15	10	15	15	10	15
		LE	20	15	15	10	15	15	10	15
NH18	59	RE	15	15	15	10	15	10	15	15
		LE	15	15	10	15	15	10	15	10
NH19	56	RE	5	10	10	0	10	15	15	15
		LE	5	5	5	5	10	15	15	15
NH20	54	RE	15	10	5	5	5	15	10	10
		LE	15	10	5	0	5	10	15	15
NH21	42	RE	15	10	10	10	10	10	5	5
		LE	15	10	5	10	15	5	10	5
NH22	37	RE	10	10	5	5	5	5	0	0
		LE	10	10	5	5	0	5	0	0
NH23	28	RE	10	5	0	5	0	0	5	0
		LE	5	0	0	5	0	0	5	0
NH24	51	RE	10	10	10	10	10	15	15	15
		LE	10	10	10	15	10	15	10	5
NH25	39	RE	10	10	0	0	5	10	5	0
		LE	5	5	0	0	5	5	5	0

Table 5-2: Main experiment hearing-impaired listeners' demographics (age and hearing thresholds)

			Frequency in Hz							
	Age	Ear	250	500	1000	2000	3000	4000	6000	8000
HI1	61	RE	15	15	15	30	55	55	50	45
		LE	15	15	10	40	55	60	40	35
HI2	62	RE	15	20	20	35	50	40	25	30
		LE	20	15	20	50	45	50	30	40
HI3	64	RE	20	20	20	25	55	65	75	65
		LE	20	15	20	25	55	60	70	75
HI4	63	RE	20	15	25	25	30	35	30	35
		LE	20	20	20	20	25	30	35	40
HI5	43	RE	15	15	15	30	60	60	55	40
		LE	20	20	25	40	60	60	60	55
HI6	49	RE	5	5	15	25	30	30	40	55
		LE	5	5	15	30	40	55	55	60
HI7	65	RE	20	20	25	50	60	55	50	45
		LE	20	20	30	45	60	60	55	45
HI8	64	RE	15	20	15	30	55	35	25	15
		LE	20	20	20	45	50	40	35	25
HI9	61	RE	15	20	10	30	30	35	50	50
		LE	10	10	10	20	35	50	50	55
HI10	58	RE	15	10	10	35	65	60	50	50
		LE	10	5	5	45	55	60	60	40
HI11	44	RE	5	15	20	30	40	50	65	60
		LE	5	15	20	35	45	50	60	60
HI12	55	RE	15	20	20	25	35	40	45	50
		LE	10	15	20	25	30	35	40	45
HI13	65	RE	15	20	25	35	35	45	50	50
		LE	15	20	30	40	35	45	50	55
HI14	53	RE	20	20	25	30	30	40	35	35
		LE	20	15	20	25	30	30	30	30
HI15	60	RE	10	10	10	30	55	55	60	70
		LE	20	15	35	45	55	60	55	75
HI16	54	RE	30	25	35	40	50	55	70	70
		LE	30	25	35	35	45	60	65	65
HI17	61	RE	10	5	0	20	55	55	50	50
		LE	10	5	10	45	60	65	70	65
HI18	56	RE	15	20	25	35	35	45	35	30
		LE	15	15	20	35	30	40	40	30
HI19	65	RE	20	15	20	35	40	55	55	65
		LE	20	20	25	40	45	55	60	70
HI20	59	RE	20	20	10	25	55	50	45	60
		LE	15	15	15	25	35	35	35	50
HI21	44	RE	20	30	45	50	55	50	50	40
		LE	20	30	40	50	55	55	50	30
HI22	49	RE	10	10	15	30	55	65	75	75
		LE	10	10	15	15	50	70	70	80
HI23	51	RE	15	15	15	25	45	45	30	20
		LE	15	15	15	30	30	45	30	15
HI24	65	RE	15	25	25	35	35	35	50	60
		LE	20	25	25	30	40	35	50	65
HI25	65	RE	15	10	15	30	45	50	75	85
		LE	20	15	20	40	45	55	80	85

APPENDIX E

Table 5-3: individual data for hearing-impaired listeners percent correct key word identification as a function of asynchronous delay

	Delay Conditions			
	0 ms	4 ms	8 ms	32 ms
HI1	98	93.5	95.5	91.5
HI2	87.5	82.5	86	82
HI3	90.5	88.5	91.5	85.5
HI4	95.5	97	98	97
HI5	92.5	93.5	95	91.5
HI6	91.5	87.5	89.5	88.5
HI7	85.5	86.5	82.5	76
HI8	97	97	96	96.5
HI9	99	98.5	98	96
HI10	90.5	91	89	85.5
HI11	78	72	74	72
HI12	91	91	89	83.5
HI13	89	86	92.5	84
HI14	90.5	90.5	88.5	88.5
HI15	76.5	78.5	75.5	69.5
HI16	84	88	83	78
HI17	94.5	94	95	95
HI18	89	91	86.5	87
HI19	97.5	93	88.5	92.5
HI20	87	85	88.5	85.5
HI21	77	74.5	72	65.5
HI22	81.5	81	79	74
HI23	98.5	97	97.5	96.5
HI24	91.5	90.5	92	89
HI25	90.5	89	86.5	83.5

HIGHLIGHTED PARTICIPANT NUMBER INDICATES THAT DATA SLOPE AS A FUNCTION OF INCREASING DELAY.

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