Speech Recognition with Linear and Non-linear Amplification in the Presence of Industrial Noise

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ABSTRACT

An abstract of the thesis of Marcia Ann Olson for the Master of Science in Speech Communication: Speech and Hearing Science presented July 10, 1996.

Title: Speech Recognition with Linear and Non-Linear Amplification in the Presence of Industrial Noise.

In order to help reduce hearing loss, the Occupational Safety and Health Administration regulates noise levels in work environments. However, hearing aids are the primary rehabilitative service provided for individuals with an occupational hearing loss. Very little is being done to monitor hearing aid use in the work environment. Noise which may be safe to an unaided ear can amplified to levels that are damaging to the ear when a hearing aid is being worn. However, it is necessary for some individuals to wear amplification in these noisy environments for safety reasons. As a consequence it is important that these individuals be able to understand speech in the presence of industrial noise while wearing amplification.

The purpose of this study was to determine if there is a significant difference in speech intelligibility between linear hearing aids and different types of non-linear hearing aids when they are used in the presence of industrial noise.

Twenty-four normal hearing subjects were selected for this study. Each subject was ask to identify words in four CID W-22 lists which had been recorded through a linear hearing aid and two different non-linear hearing aids.

Test results showed significantly better word recognition for the linear in quiet condition over all other conditions. Significantly higher scores were obtained for the TILL condition than were obtained for the Linear in noise and the BILL condition. These preliminary results suggest that an individual wearing amplification in a noisy work environment would benefit with a TILL circuit. The TILL circuit would provide better speech intelligibility in this type of environment. Therefore, providing a safer work environment for the hearing aid user.

SPEECH RECOGNITION

WITH LINEAR AND NON-LINEAR AMPLIFICATION

IN THE PRESENCE OF INDUSTRIAL NOISE

by

MARCIA ANN OLSON

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CHAPTER I

INTRODUCTION

According to the National Institute of Occupational Safety and Health (NIOSH), occupational hearing loss is one of the ten most important preventable injuries (Lee-Feldstein, 1993). In order to prevent hearing loss, the Occupational Safety and Health Administration (OSHA, 1983) regulates the noise levels in the workplace. However, little attention is paid to the fact that many hearing impaired individuals are wearing hearing aids in the presence of industrial noise. Riko, McShane, Hyde, & Alerti (1990), report that the primary rehabilitative service provided to those individuals with an occupational hearing loss is the use of hearing aids. The study also reported that of the hearing loss claims they reviewed, 33.4% of the claimants owned a hearing aid. The problem is that even when occupational noise is safe to the unaided ear, it can be raised to increasingly high levels when a hearing aid is worn (Dolan & Maurer, 1996).

The output of a hearing aid can be limited in order to avoid levels which would exceed an individual's loudness discomfort level or levels which can be damaging. Two ways in which this can be accomplished is saturation limiting and compression limiting (Teder, 1993). Saturation output is also known as the maximum power output (MPO) of the hearing aid. The maximum power output of a hearing aid is the highest level of sound pressure to which the hearing aid is capable of amplifying. When the maximum output level is reached saturation limiting occurs when the amplifier is no longer capable of amplifying the sound beyond this level or the receiver is no longer capable of transducing a higher level signal (Pollack, 1975).

Often an individual's hearing aid is operating at a level of saturation (Van Tasell, 1993). In fact, the hearing aid user's own voice can cause the hearing aid to be saturated if the saturation sound-pressure level is not set high enough. When a linear hearing aid is worn in a noisy work environment it could also be saturating. When an individual's hearing aid has reached saturation, peak clipping occurs. This clipping of the waveform peaks results in harmonic distortion and intermodulation distortion (Skinner, 1988).

The distortion may create an unacceptable speech signal for the user and it degrades the amplified sound (Skinner, 1988; Teder, 1993; Preves & Newton, 1989; Moore, 1987, Van Tasell & Crain, 1992). The harmonic distortion may reduce the syllabic characteristics of speech (Teder, 1993). If the sound quality is poor enough the hearing aid may be rejected by the user.

Compression limiting, the second method of output limiting, occurs when the gain of the hearing aid is automatically reduced at a given input or output level (Teder, 1993). Compression limiting reduces or eliminates saturation which occurs in a linear hearing aid (Teder, 1993). The compression amplifier was initially developed to avoid sounds reaching the listener's loudness discomfort level (LDL). Since the initial development of the compression circuit, it has been considered a circuit which may allow individuals with a limited dynamic range to improve his or her speech intelligibility. However, Teder states "either method can cause major changes in the amplified speech, with negative consequences to intelligibility" (1993, p. 42).

Base Increase at Low Levels (BILL), Treble Increase at Low Levels (TILL), and the Programmable Increase at Low Levels (PILL) are other methods of limiting the output of the hearing aid. The BILL, TILL, and PILL are all considered automatic signal processing (ASP) circuits (Killion, Stabb, & Preves, 1990).

The BILL circuit was designed for individuals who are frequently exposed to noisy environments. The BILL circuit provides more bass response for lower input levels and automatically reduces the low frequency amplification for high input levels. This reduction in low frequency amplification as a result of high input levels helps reduce "overload distortion" (Killion, et al., 1990).

The TILL circuit was designed for individuals with a high frequency hearing loss. The TILL circuit provides more gain at the high frequencies for low level inputs and automatically decreases gain for high level input signals.

Lastly, the PILL circuit is a programmable level dependent frequency response which can be programmed so that the bass or the treble response decreases with increasing input level. Therefore, the hearing aid is being programmed to have responses similar to the BILL, or TILL circuit. The PILL circuit for this reason is the most versatile circuit (Killion, et al., 1990). Some ASP instruments are multi-band, where the frequency range is divided into two or more bands. Typically the gain of the multi-band hearing aid can be adjusted for each band individually, allowing for finer resolution of the frequency response of the hearing aid. (Preves, 1993).

Many researchers have focused on differences in speech intelligibility associated with compression hearing aids versus linear hearing aids (Walker, Byrne, & Dillon, 1984; Lippman, Braida, & Durlach, 1981; Dreschler, 1988a; Dreschler, Eberhardt, & Melk, 1984; Bustanamte & Braida, 1987; Moore, 1987). Various speech tests have been used to assess speech intelligibility. Yund, Simon, and Efron (1987), used the nonsense syllable test (NST), while Bustamante & Braida (1987), and Dreschler (1988b), used consonant-vowel-consonant words. The Bench and Bamford (BKB) sentence list was used by Moore, Glasberg, & Stone (1991), Moore, Johnson, Clark, & Pluvinage (1992), and Moore (1987) to evaluate the comparison of hearing aids for speech intelligibility.

Walker, et al. (1984) found that compression was beneficial to some individuals but not to all. They recommended that more research be done to determine what conditions result in maximum speech intelligibility performance for individuals when using compression amplification. Dreschler (1988a) found speech intelligibility to be significantly better for individuals when compression was used as opposed to linear amplification, as did Moore (1987) when comparing a two-channel compression hearing aid to a linear hearing aid. Nablek, (1983) noted an improvement in speech intelligibility with a compression hearing aid in the presence of noise. Dreschler, (1988b) compared compression to linear amplification in quiet using a 50 word list of nonsense consonant-vowel-consonant words, that were presented at four different intensity levels. Their study found compression amplification to be significantly better then the linear amplifier for all input levels. However, Dreschler et al., (1984) and Lippman et al., (1981) found no significant difference between compression amplification and linear amplification.

When comparing the performance of the hearing aids in the presence of noise, Dreschler, et al. (1984), and Moore (1987) used a random competing noise which represented the long term average speech spectrum. Other noise used was cafeteria babble (Lippman et al., 1981) and random noise (Walker, et al., 1984).

However, there have been no studies of how speech intelligibility is affected when a hearing aid is used in the presence of industrial noise. This issue is important because hearing impaired workers may need to understand speech while on the job. If workers are unable to understand speech it could lead to dangerous situations, depending on the work setting and circumstances.

The focus of this study was to determine if there is a significant difference in speech intelligibility between linear hearing aids and two different types of non-linear hearing aids when they are used in the presence of industrial noise. This issue will be addressed by comparing speech intelligibility scores obtained in quiet and in the presence of industrial noise using a Danavox Aura hearing aid, programmed to function as a linear circuit, a BILL circuit, and a TILL circuit.

CHAPTER II

REVIEW OF THE LITERATURE

Noise Exposure Associated with Hearing Aids

Many researchers have investigated the possibility of acoustic trauma resulting from hearing aid use. Conflicting theories have developed from various research studies. Some researchers believe powerful hearing aid can cause temporary threshold shifts (TTS) or even permanent threshold shifts (PTS), while others disagree and have found no significant acoustic trauma as a result of prolonged hearing aid use.

Roberts (1970) studied a case of a young boy wearing ear level amplification in one ear who had been diagnosed as having a progressive hereditary perceptive deafness. Nine months after being fit with the hearing aid a decrease in hearing sensitivity had occurred at 4000 Hz. Roberts describes this dip at 4000 Hz as the "classical C5 dip" for acoustic trauma. The unaided ear was used as the control ear. The hearing sensitivity in the unaided ear had also decreased by 10 dB, but the loss had occurred at all frequencies, not just at 4000 Hz. Ross & Truex (1965), as well as Jerger & Lewis (1975) also reviewed individual case studies which supported Roberts (1970) study. Ross & Truex (1965) recommend the use of automatic gain control (AGC) circuit to help limit the output of the hearing aid, which in turn would reduce the possibility of acoustic trauma.

Research completed by Macrae (1965, 1968, & 1991) also supports the theory that PTS can occur from hearing aid use. Macrae's subjects for his research consisted of children who had been fit monauraly. The unaided ear was used as the control, while any changes in the aided ear were analyzed. Macrae observed no change or no significant change in the hearing sensitivity of the unaided ear, but did notice significant deterioration of hearing in the aided ear. The PTS he observed in the aided ear was essentially flat across the frequencies with the use of power behind-the-ear hearing aids.

Humes & Bess (1981) suggested that the degree of TTS obtained from a short duration of a moderate intensity noise exposure may provide a means of predicting noise induced permanent threshold shift (NIPTS) for longer durations of exposure to higher intensity noise. This theory has prompted many researches to study TTS. If a relationship between the duration of the noise exposure, the intensity of the signal, and the amount of TTS as a result of the noise and the duration of the exposure can be established then some researchers believe this relationship will provide a means to predicting PTS. Humes and Bess (1981) describe a method of obtaining a TTS value used to predict possible PTS. This method consists of obtaining the TTS value two minutes after the noise exposure, this is known as TTS_2. It is known that TTS grows exponentially for eight to twelve hours after exposure. After you exceed this time period an asymptotic threshold shift (ATS) is met. This is when the values of the TTS remain the same. For instance if a TTS measurement is taken at twenty-four hours after exposure and forty-eight hours after exposure, the same TTS measure would be obtained. However, it has been found that it takes longer to recover to pre-exposure level when exposed to noise for forty-eight hours then when exposed for twenty-four hours, even though the same TTS measurement was obtained for both. Humes and Bess (1981) report varying correlation between TTS_2 measures and histological studies of damage to the cochlea resulting from noise exposure. A stronger correlation for estimated damage for higher frequencies has been obtained than for lower frequency predictions using TTS_2.

A second method of measuring TTS has been developed. This measurement is referred to as the time integral of TTS (ITTS). This method of measuring TTS is more accurate then TTS_2 measurements. ITTS calculations require that multiple threshold measurements be taken through the growth and recovery pattern. Measurements are taken until complete recovery has occurred. Studies have shown a direct or linear correlation between noise dose and ITTS. This would in turn mean that ITTS can be predicted by any given noise dose. This direct correlation has also been seen when measuring ITTS resulting from hearing aid use. It was found that the output of the hearing aid and ITTS is directly related, the greater the output of the hearing aid the greater the amount of ITTS (Humes & Bess, 1981).

Macrae (1968) found that TTS occurred in children with sensorineural hearing loss who had been fit with high gain amplification. In all four children there was a significant decrease in hearing sensitivity. The shift in the threshold recovered over time. It was observed in three of the children that the rate of recovery was slower than rates typically obtained in normal-hearing children who experienced the same amount of TTS. Humes (1978) and

Harford & Markle (1955) also observed TTS after hearing aid use. Humes (1978) found measurements taken after noise exposure resulted in TTS of greater then 40 dB, which occurred across the frequency range. Hearing sensitivity recovered to a normal level with in 24 hours of the exposure. It was determined that the amount of TTS was influenced the greatest by the sound pressure level of the hearing aid, the duration of the exposure, and the frequency of the test signal. The children in Harford & Markle's (1955) study showed thresholds returning to the pre exposure levels after the use of the hearings was discontinued.

Contrary to the findings that hearing aids can cause PTS or TTS, Ross & Lerman (1967), Bellfleur & Vandyke (1968), Markides (1976) and Naunton (1957) reported no significant change or shift in thresholds as a result of hearing aid use. Ross and Lerman (1967) limited the maximum output of the hearing aids in their study, which may have limited the possibility of hearing loss. Bellefleur & Vandyke (1968) stated that absence of PTS or TTS in some children who wear powerful hearing aids may be a result of setting the volume of the hearing aid at a lower level. This may lower the output to levels that are not dangerous.

Jerger and Lewis (1975) criticize the use of children for these research studies because it is more likely that children will have fluctuating hearing losses which can affect the results. There is also the possibility of over amplification from auditory trainers, which could be the cause of the acoustic trauma not the hearing aid.

Use of Hearing Aids In Industrial Settings

Dolan, Maurer, & O'Connor, (1992) compared the output of three hearing aids to determine if the aided 8 hour time weighted average (TWA) exceeded OSHA's maximum of 90 dBA. It was determined that moderate to high gain hearing aids worn in noise averaging 75 to 80 dBA resulted in a TWA which exceeded the OSHA guidelines of 90 dBA. These results would suggest OSHA guidelines are not appropriate for hearing aid users. A study completed later by Dolan and Maurer (1994) showed that the use of a multiband hearing aids could result in a reduction of the worker's average daily noise exposure to 90 dBA or lower. It was also found that single-channel compression, fixed and adaptive high-pass filters reduced the average noise exposure, but in most cases not as effectively as the multiband hearing aid.

Hearing Aids

Automatic signal processing (ASP) is a method of limiting the output of a hearing aid. According to Skinner (1988), the major reason for using automatic signal processing is to enable the hearing aid user both to hear individuals speaking to them from a distance and to listen to their own voice at a comfortable level. According to Mare, Dreschler, and Verschuure (1992) the range of perceivable sound by individuals with a sensory-neural hearing loss is reduced. This reduction is a result of increased auditory threshold when in combination with a normal discomfort level. Linear hearing aids can alter the effects of the sensitivity loss, but may also cause parts of the speech signal to be too loud and reach discomfort levels. The compression circuit, which reduces the dynamic range, is an alternative to the linear circuit to alleviate the discomfort problem.

Linear hearing aids provide a constant 1:1 ratio between the input of the signal and the output of the hearing aid. This relationship stays constant until the maximum output of the hearing aid is reached. When the input has exceeded the maximum output, the hearing aid begins to saturate (Katz, 1994; Pollack, 1975). Katz (1994) defines saturation as the point when the input continues to increase, but does not result in an increase in the output. At this point peak clipping occurs. Pollack (1975) reports that speech intelligibility is not substantially decreased by peak clipping but the sound quality is decreased. Kuk (1996) reports that the distortion that occurs from saturation does not significantly affect the speech signal until the total harmonic distortion has reached more than 20%.

When saturation occurs in the presence of background noise, the higher amplitude peaks of speech are "clipped". The clipping of the speech peaks causes a reduction in the speech signal however, it does not alter the noise. Therefore, the signal to noise ratio is decreased when peak clipping occurs thus making speech understanding more difficult (Kuk, 1996).

Traditional automatic signal processing (ASP) either provides more amplification at low level input signal or reduces the gain of a high level input signal, without changing the frequency spectrum of the signal. A more advanced ASP circuit know as level dependent frequency response (LDFR) circuits has been developed in the last decade (Killion et. al., 1990). Killion et. al. (1990), developed a classification system for the different types of LDFR ASP circuits in relation to how the circuit responds to low level inputs. The two main categories of ASP circuits are the fixed frequency response (FFR) and level dependent frequency response (LDFR). The Base Increase at Low Levels (BILL), and the Treble Increase as Low Levels (TILL) circuit are two types of LDFR circuits. The BILL circuit was designed for individuals who are frequently exposed to noisy environments. It provides less amplification of low frequencies in high intensity input signals or it provides more amplification in the low frequencies in low intensity input signals (Killion, et al., 1990; Katz, 1994).

The TILL circuit was designed for individuals with a high frequency hearing loss. The TILL circuit provides more amplification of high frequencies for low level signals and automatically decreases gain for high level input signals. According to Killion, et al. (1990), the reduction of gain for high level input signals helps reduce audible distortion. By providing better high frequency amplification, which extends in to the higher frequency range, (i.e. amplification being provided up to approximately 6000 Hz) the speech recognition of weak high-frequency consonant speech cues should improve. This higher frequency amplification should improve speech intelligibility an individual with a high frequency hearing loss. Therefore, the assumption of the TILL circuit is that the improvement in overall speech intelligibility will carry over to noise environments and also improve the individual's overall speech recognition in noise. Overall the TILL circuit makes no attempt to reduce the noise (Schum, 1996).

The K-Amp (Killion amplifier) is a specific type of TILL circuit. The K-AMP circuit provides high frequency emphasis for low intensity signals (Teder, 1993; Preves, 1992; Hickson, 1994). It was originally designed for individuals who have difficulty understanding low level spoken speech and individuals who have difficulty in the presence of high environmental noise. The K-Amp has been found to perform better than noise reduction circuits in high environmental noise as reported by Preves (1992); however the author does not specify which noise suppression circuits. This boost of high frequencies in the presence of low levels allows consonants to be amplified to a level more closely approximating the vowels (Preves, 1992).

The BILL circuit is designed to reduce the background noise, by reducing the low frequencies. Through the reduction of the lower frequency noise the BILL circuit also minimizes the effects of the upward spread of masking (Schum, 1996).

Many studies have been completed comparing the effects of different types of amplification circuits and individual's speech intelligibility abilities. Dempsey (1987) completed a study that compared a BILL circuit to a linear circuit. Testing was completed at four different S/N ratios (quiet, +5, 0, -5). Performance for the BILL circuit resulted significantly better speech intelligibility scores for all conditions. Similar results were found by Ono, Kanzaki, & Mizoi (1983). Their results showed an increase of up to a 15% improvement in speech recognition for 50 out of 53 of the subjects who had a sensorineural hearing loss, when the ASP circuit, BILL. However, Tyler &

Kuk (1989) found did not find that the noise suppression circuit was more beneficial in the recognition of consonants in the presence of speech babble.

The Danavox Aura programmable hearing aid is a 3-channel dynamic range compression circuit, with each of the three channels having independent input compression. The three channels divide the input into independent frequency bands. By varying compression threshold in each band the Danavox Aura can be programmed to operate as a TILL or BILL hearing aid.

Word Recognition Tests

One of the purposes of speech audiometry is to enable the audiologist to assess how a hearing loss affects the individuals' daily communication capabilities (Hannley, 1986; Goetzinger, 1972; & Jerger, 1973). The first speech intelligibility tests developed were lists of 50 monosyllabic words (Hannley, 1986). Several different speech lists have since been developed, including the PAL PB-50 which was developed by the Harvard Psychoacoustic laboratory, the Central Institute for the deaf (CID) W-22 word list developed by Hirsh et al., (1952) and the NU-6 word list developed by Northwestern University (Hannley, 1986; & Jerger, 1973). When developing the test materials, the following criteria were met. First, the word lists were to have a phonemic distribution similar to the English language (i.e. phonemically balanced). In addition, the words needed to be monosyllabic, familiar, and equal in difficulty among the different lists (Hannley, 1986; Jerger, 1973). Speech intelligibility testing is usually administered at 25 to 40 dB above the speech reception threshold or at the most comfortable level

(MCL) of speech. The range of presentation levels from 25-40 dB SL represents a range of speech from soft spoken speech to a normal conversation level (Hannley, 1986; Pollack, 1975).

By the 1940's the techniques for the development and administration of speech audiometry materials had been established and only small basic changes have been made since that time (Jerger, 1973). Hirsh et al. (1952) developed the Central Institute for the Deaf Auditory test W-22 as a result of their dissatisfaction with the PAL PB-50 test material. Hirsh et. al. (1952) felt the Pal PB-50's words lists were not phonemically balanced and that the words list contained several rare words. They also felt that no suitable recording of the test PAL PB-50 had been recorded which was commercially available.

Speech discrimination testing has also been applied to assess the usefulness of various types of amplification (Jerger, 1973). Harting & Newhart (1936) compared the relationship between "electroacoustic performance characteristics of carbon-type hearing aids and speech understanding." Carhart (1946) developed the first program comparing hearing aids using speech audiometry to help in the selection of a hearing aid for an individual.

Effects of output limitation on speech discrimination

Plomp (1994) reported that individuals with a hearing deficit on the average require a signal to noise ratio (S/N) of 3 to 6 dB higher than normal hearing individuals to adequately understand speech. This would partially explain why hearing impaired individuals have greater speech recognition difficulties in the presence of competing noise. A study completed by

Duquesnoy & Plomp (1983) showed a 10 dB increase in speech reception thresholds when noise is increased by 10 dB. It was concluded that hearing aids amplify speech adequately in quiet but not in the presence of noise. To address the need for better amplification in order to increase speech recognition in the presence of noise, compression hearing aids were designed. It was theorized that a person will understand speech better when amplified speech is audible through out the entire frequency range. Wide-band automatic gain control (AGC) compression and frequency dependent AGC compression circuits have been used to help improve speech recognition (Plomp, 1994).

Plomp (1994) points out some limitations to wide-band AGC compression and frequency dependent AGC circuits. When wide-band AGC is used it causes as input signal to be maintained at an "optimal" level. However, the listener may no longer perceive normal loudness variations in speech. For instance soft sounds are made louder so they are more audible and, on the other hand, louder sounds are compressed so that they are within a listener's comfortable levels. An alternative to wide-band compression is frequency dependent AGC. The down fall of frequency dependent AGC when adjusted to optimize speech recognition using long time constraints is that a time delay occurs between the time the initial input signal is heard and the compression circuit is activated. Therefore, an initial soft sound may not be loud enough to be heard or else a loud sound initially is uncomfortably loud, until the compression circuit is activated. Both of these conditions may hinder the individual's ability to understand the speech signal because it is too soft and not audible or too loud and uncomfortable. On the other hand, if the time constraints are shortened to eliminate the time delay issue then the speech spectrum is constantly altered which in turn reduces speech intelligibility.

Dreschler (1988a), compared the effects of speech discrimination of sixteen high school hearing impaired students using a compression hearing aid. There were two conditions: with the aid set to maximum compression and no compression (linear). Thirteen of the subjects had a sensorineural hearing loss and three had a conductive hearing loss. Fifty nonsense consonant-vowel-consonant words were used as the test stimuli. The four different word lists were recorded at four different levels ranging from 55 dBA to 85 dBA measured as the dB input of the hearing aid at the microphone. The speech material was amplified by the hearing aid and recorded at each of the four presentation levels. The recorded material was presented to the subject just above his or her 50% intelligibility threshold. The author found that speech intelligibility scores were significantly better in the compression condition than when the hearing aid was not compressing.

A study by Lippman, Braida, and Durlach (1981) looked at the differences between a computer simulated multichannel amplitude compression hearing aid and a linear hearing aid. Five subjects were used for this study. Each had a moderate to severe sensorineural hearing loss. The hearing aid conditions were as follows: Compression hearing aid 1 (C1) provided a reduced compression ratio and less high frequency emphasis; compression hearing aid 2 (C2) restored normal loudness contours for pure tones; linear hearing aid 1 (L1) had a response appropriate for a flat hearing loss; linear hearing aid 2

(L2), and linear hearing aids 3 (L3), and 4 (L4) provided varying amount of high frequency emphasis. The first experiment Lippman et al. conducted looked at the differences between the hearing aids when they were adjusted to restore normal loudness contours for pure tones. Nonsense sentences and consonant-vowel-consonant nonsense syllables words were used as the test stimuli. The test materials were recorded by both a male and female speaker and presented in quiet and in the presence of cafeteria noise. The noise was presented at a signal-to-noise ratio (s/n) of 10 dB. It was found that the linear amplification resulted in better intelligibility scores then either of the compression circuits. It was concluded from this experiment that compression does not result in better speech discrimination than linear amplification when speech levels vary only a small amount.

The second experiment conducted compared L1, L2, and C1 hearing aids. The speech material used was the CID W-22 word lists presented in quiet, the SPIN test presented in quiet and noise, and the Harvard Sentence test presented in quiet. The authors found that compression was not significantly better then linear amplification in quiet or noise. In all of the test performed the L2 hearing aid performed better then the L1 hearing aid. In quiet the scores were fairly equal for L2 and C1, but in noise the C1 hearing aid resulted in mean scores that were 6 points worse than the L2 hearing aid.

The third and last experiment conducted compared the effects of speech intelligibility presented at reduced input levels. The hearing aids used for this experiment were C1 and the subjects' best linear hearing aid according to scores obtained in experiment one. The speech material was presented at

0 dB, 8 dB, 16 dB, and 24 dB below the most comfortable listening level (MCL) for each subject. The compression circuit and liner circuit were equivalent when speech was presented at the MCL. However, it was found that compression resulted in much better scores than linear amplification at lower levels.

Crain and Yund (1995) completed a similar study to Lippman, Braida and Durlach (1981), which compared two types of multichannel compression in order to evaluate vowel and stop consonant discrimination. A flat and a shaped multichannel compression (MCC) circuit were compared. The flat compression system consisted of the same compression ratio in each channel, while the shaped compression was adjusted accordingly to the subject's area of audibility in each channel.

Crain and Yund (1995) scored recognition of vowels in quiet and vowel-consonant-vowel combination recognition in quiet. The overall results for both experiments concluded that multichannel compression can result in degraded vowel and voiced stop consonants only in extreme compression conditions. With the shaped compression condition, it was concluded that the number of channels did not effects discrimination. It was found in the flat compression conditions, however that discrimination abilities decreased as the number of channels increased. Overall the results of this study are not consistent with the theory that hearing aids which operate with a MCC circuit with greater then two channels degrades the speech signal which decrease speech recognition.

CHAPTER III

METHODS

The Linear, BILL, and TILL circuits were compared in the presence of industrial noise and in quiet. CID W-22 word lists were used to assess speech intelligibility for the four conditions.

Subjects

Twenty-four volunteer subjects participated in this study. The subjects were asked by the researcher to participate in the study. If they agreed to participate, then a screening process was begun. To qualify for the study, the subjects had to have pure tone thresholds of 20 dB HL (ANSI S3.6, 1989) or less, for audiometric test frequencies from 250 Hz through 8000 Hz. All subjects had negative histories of auditory pathologies.

Instrumentation

The Stimulus Tape

For each test condition, a tape recording of hearing aid output with speech and/or noise as input was produced through the following procedures. Industrial noise was presented via a TEAC DA-P20 digital audio tape(DAT) recorder and the Central Institute for the Deaf (CID) W-22 word list was presented via a Sony CD player model CDP-C505. The DAT recorder was connected to the left channel of a NAD 2240DE amplifier and the CD player was connected to the right channel of the amplifier. The output of the amplifier was led to the input of two separate Leader LAT-45 attenuators. The output of each attenuator was led to two separate Realistic Nova 15 loudspeakers. The loudspeakers were placed inside a sound treated room, and the speakers were placed at a 45 degree angle and a 135 degree angel to KEMAR's ear[1]. One loudspeaker presented the industrial noise at a time weighted average of 80 dBA while the other loud speaker presented the speech signal at an average of 70 dB SPL. It was ensured that the industrial noise was a time weighted average of 85 dBA by monitoring the noise at the level of the microphone of the hearing aid using a Larson Davis 821 integrated sound level meter. KEMAR was placed so that his ear was 30 inches from the speakers and centered between the speakers. The speech and industrial noise was amplified by a programmable Danavox Aura 143u hearing aid which was placed on the ear of KEMAR and coupled via a silicone earmold to a Zwislocki coupler. The output of the hearing aid was transduced by an ACO Pacific 1/2 inch pressure microphone which was connected to an ACC Pacific preamplifier and ACO Pacific Acoustical Interface. Output of the power supply was amplified by a Rane PE17 parametric equalizer and routed to a Proton 740 audio cassette recorder. A Danavox Aura 143u programmable hearing aid was used in this study.

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[1] Knowles Electronics Mannequin for Auditory Research (KEMAR) is a representation of the average adult head and torso. KEMAR's pinna and ear canal approximate a human's natural resonance and impedance properties. A Zwislocki coupler is placed inside KEMAR located at the ear canal. The Zwislocki coupler was also designed to approximate the resonant characteristics of a human ear.

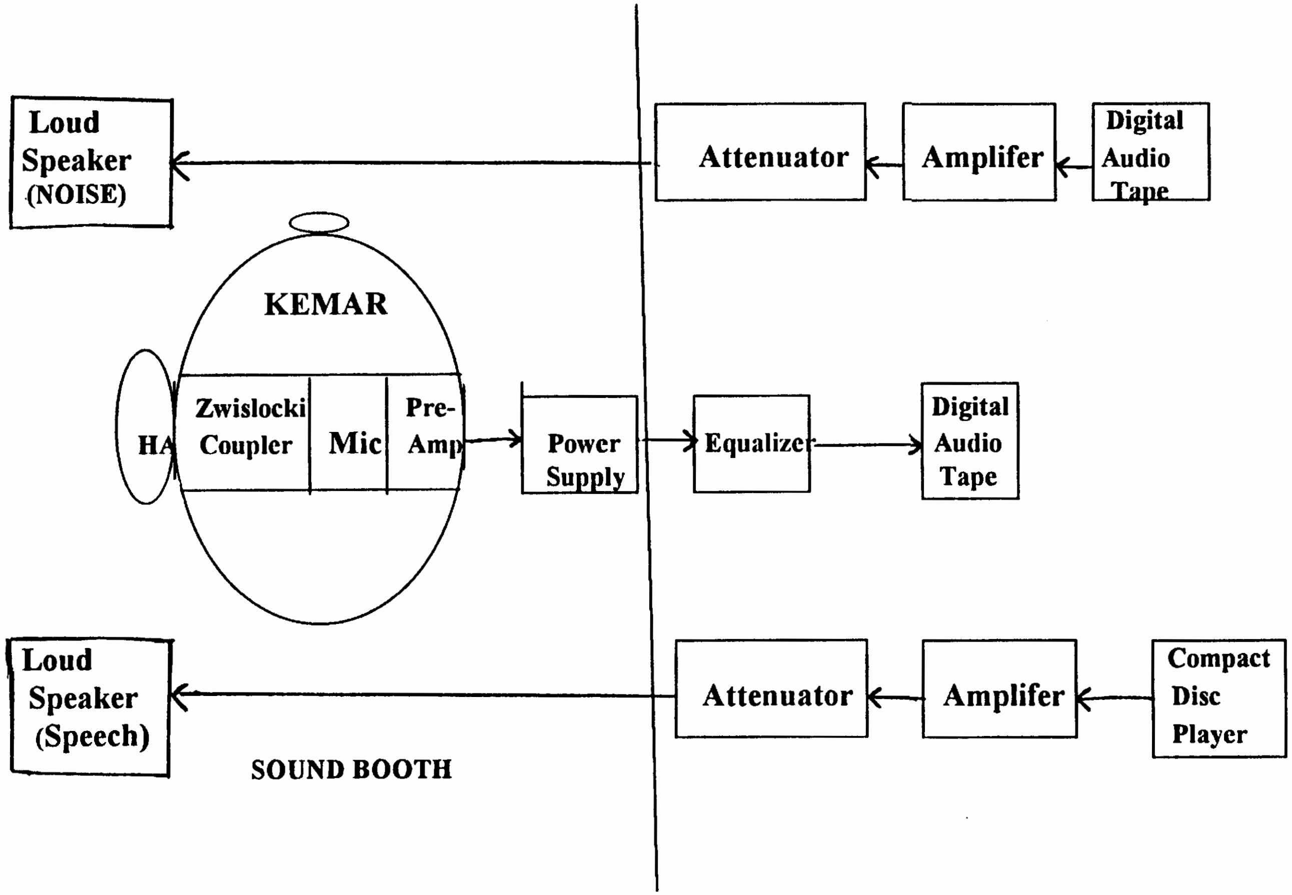


Fig. 1 Block diagram of instrumentation used to record digital master tape.

The hearing aid was programmed to function like a linear circuit, a BILL circuit, and a TILL circuit (Appendix A). A hypothetical hearing loss was used to program the gain of the hearing aid (Table I). The output of the aid for each of the three programs was recorded onto a cassette tape which was presented to each subject

Procedures

The industrial noise used for this study was recorded at the Beaver Heat Treating Company in Milwaukee. The measurements were recorded using a TEAC model DA-P20 digital audio tape recorder and an 812 Larson-Davis sound level meter.

The speech stimuli used for this experiment was the Central Institute for the Deaf W-22 word lists. This test consists of four list of fifty monosyllabic word lists. The CID W-22 speech test was recorded onto a compact disc by the Hearing and Speech Sciences Laboratory at Brigham Young University.

Each subject was seated in a sound-treated room. The subject was read the instruction for completing the speech testing (Appendix B). The recorded speech and noise were delivered to the listener by means of a Proton 740 cassette deck. Output of the cassette deck was connected to the left channel of a Grason Stadler 16 Audiometer and transduced through a TDH-39 headphone. Each subject was asked to write down each response. After the completion of the testing, the responses of the subject were marked as either right or wrong by the researcher. The speech recognition score was then determined by tallying the number of correct responses in comparison to the total number of possible words.

Table I

Hypothetical Hearing Loss

Frequency (Hz)	dB HL
250	35
500	35
1000	30
2000	50
3000	70
4000	75
6000	75
8000	85

Four different conditions were tested using the CID W-22 speech recognition test. To minimize order effect each of the four word lists were counter balanced with each of the four hearing aid conditions. The order of the hearing aid conditions was also counter balanced across subjects (see appendix C for specific order).

CHAPTER IV

RESULTS

Data Analysis

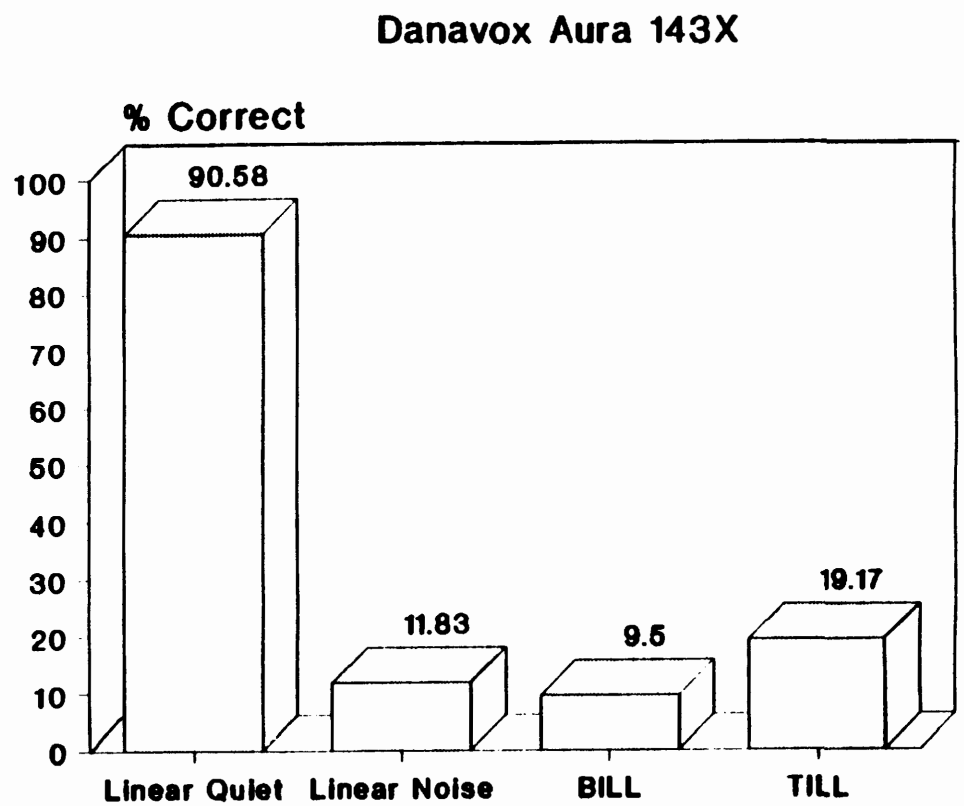
Speech intelligibility scores of twenty-four normal hearing subjects were obtained to determine if a particular hearing aid circuit would result in better speech understanding in the presence of industrial noise. The individual scores for the CID W-22 word lists are given in Table II. The bar graph in Figure 1 represents the mean percentage of correct responses for twenty-four subjects for each test condition: linear in quiet; linear in noise; TILL in noise and BILL in noise. The mean score for the Linear circuit in quiet was 90.6% with a range of 78-96% with a standard deviation of 4.1. Scores were drastically lower when the speech was presented in industrial noise. The mean score for the linear circuit when presented in the presence of industrial noise was 11.8%, with a range of 2-28% and a standard deviation of 6.2. The BILL circuit resulted in a mean score of 9.5% with a range of 0-18% and a standard deviation of 5.5. The last circuit, TILL, resulted in a mean score of 19.2% with a range of 8-40% and a stand deviation of 8.2. Thus, when the speech was presented in industrial noise the subjects tended to score the highest with the TILL circuit.

The data were subjected to a one-way analysis of variance, the results of which are given in Table III. The test conditions main effect was significant ($P<.001$). Follow-up t-tests on paired samples revealed that the scores in the linear in quiet condition were significantly higher than in all other conditions ($P<.001$). Significantly higher scores were obtained for the TILL condition than were obtained for the Linear in noise ($P<.003$) and BILL condition ($P<.001$). The linear in noise was not significantly different form the BILL ($P<.129$), however.

TABLE II

Individual Intelligibility Scores

SUBJECT	LINEAR QUIET	LINEAR NOISE	BILL	TILL
1	86	16	12	20
2	92	12	10	40
3	94	8	4	24
4	94	16	16	28
5	86	2	0	20
6	78	2	4	16
7	90	12	4	8
8	88	16	2	16
9	92	16	4	14
10	88	28	18	14
11	94	22	6	18
12	94	20	12	16
13	84	10	10	12
14	92	12	6	8
15	92	8	18	12
16	90	16	8	18
17	90	10	4	16
18	96	10	6	8
19	92	6	12	22
20	96	8	14	34
21	90	4	10	22
22	94	10	18	22
23	92	12	16	34
24	90	8	14	18



N = 24

Fig. 2 Bar Graph of Percent correct scores.

Table IIII

Results of One-way Analysis of Variance

	SS	DF	MS	F	P
Circuit	108174.5	3	36058.2	1116.4	0.0
Error	2228.542	69	32.298		

CHAPTER V

DISCUSSION AND CONCLUSIONS

Many hearing aid users have difficulty distinguishing or understanding speech in the presence of background noise. Researchers have studied the effectiveness of various hearing aid circuits in the presence of background noise, such as cafeteria noise, babble noise and speech weighted noise (Yund et al., 1987; Bustamante & Braida, 1987; Dreschler 1988b; Moore et al., 1991; Moore et al.' 1992; and Moore, 1987) However, very little research has been completed regarding the improvement of a hearing aid user's ability to understand speech in the presence of industrial noise with regards to various types of circuits.

The purpose of this study was to determine if non-linear amplification would provide a significant difference in speech intelligibility when used in the presence of industrial noise. The four CID W-22 word lists were recorded in the presence of industrial noise. Each subject identified as many words as possible and the percentage of words correctly identified was calculated.

It was found that speech intelligibility was greatly reduced when the words were presented in background noise relative to when words were presented in quiet. No improvement was seen with the BILL circuit, which reduces low frequencies in the presence of a high level input. However, the TILL circuit, which reduces high frequencies in the presence of high level inputs, resulted in a statistically significant higher score than the BILL or Linear circuits in the noise condition.

It is possible that the TILL circuit provided greater speech understanding because the noise was primarily weighted in the higher frequency range. This would mean that the high frequency portion of the noise would not be amplified as greatly as it would be for either the BILL or Linear condition. This would result in less masking of the speech signal, which in turn would provide better speech intelligibility. In contrast the BILL circuit would reduce only low frequencies, and not reduce the higher frequency noise. This would result in a higher noise level than what was present in the TILL condition, thus causing the high frequency consonants to be masked more by the noise, thus reducing speech intelligibility. It is hypothesized that the TILL circuit provided better speech understanding than the linear circuit because the linear circuit uses peak clipping to reduce the output of the hearing aid in the presence of high level inputs. Peak clipping will cause distortion in the speech signal, thus making it less intelligible.

Implications

The results of this study have shown that the TILL circuit can significantly improve an individual's speech intelligibility in the presence of the industrial noise. This may allow individuals who wear hearing aids and work in these types of noise environments the ability to understand speech better. Along with improved speech understanding while wearing a hearing aid with a TILL circuit, this may also provide the employee a safer work environment because it also reduces noise exposure. The awareness of sounds and understanding of other co-workers to warn the hearing aid user of danger while working in noise could be better provided when wearing a TILL amplification hearing aid.

Limitations

The subjects used in this study were all normal hearing subjects. Further research should be completed using the same test conditions, but with hearing impaired subjects. It may or may not be found that hearing impaired subjects' speech intelligibility scores would vary from those obtained from normal hearing subjects. Individuals with a sensorineural hearing loss tend to have a reduction in speech discrimination. This may be a result of the reduced acuity of sound transmission through the auditory system, specifically in the cochlea and auditory nerve. The individual with a high frequency hearing loss is also not hearing high frequency consonants. The high frequency consonants provide speech cues which in turn improves speech discrimination or intelligibility. Therefore, a high frequency hearing loss may result in reduction of speech intelligibility in comparison to a normal hearing individual particularly when noise is introduced which tends to mask out even more of the speech cues than in a quiet listening situation.

It is also recommended that following parameters be addressed: the signal to noise ratio, the type of industrial noise used, and the speech test used to calculate speech intelligibility.

The signal to noise ratio may alter the outcome of speech intelligibility scores. This study was completed as a signal to noise ratio of -10 dB. This signal to noise ratio makes speech recognition very difficult. In normal listening situations, an individual will naturally increase the intensity of his or her voice to over come the noise. This increase in the intensity of the speech would result in a much lower signal to noise ratio possibly making speech more intelligible.

The speech signal used for this study was single syllable words not connected speech discourse. In the "natural" work environment the employee would converse with other co-workers using connected discourse, therefore the speech material used in this study is not an accurate replica of the normal interactions of individuals. Further research could be completed using a standardized speech intelligibility test which consists of complete sentences.

Finally, the industrial noise used in this study was recorded at a steel plant, therefore the results may not correlate to other noisy work environments. It would be recommended that further studies be completed using the same test conditions, but have the words be presented in the presence of other types of industrial noise.

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APPENDIX A

Hearing Aid Parameters

	LINEAR	BILL	TILL
MAX SSPL90	119.6 dB	117.5 dB	111.5 dB

Freq. Bands	Frequency Range, Hz	Compression Threshold, dB SPL	
		BILL	TILL
LOW	0-600	60	90
MID	600-3000	70	70
HIGH	3000-6000	90	60

APPENDIX B

Subject Instructions

You first will hear a noise in one ear, which will be present for the entire testing time. You will then hear a man's voice presenting single syllable words in the same ear that the noise is being presented to. You will hear nothing in the other ear.

You will be ask to identify four lists of words consisting of fifty single syllable words. You will be required to write down your response to each word. There will be a short pause in between each word in order to give you time to write the word down. If you are unsure of the word go ahead and guess. You will be notified between each of the four word lists when one word list has finished and when the next one will begin.

APPENDIX C

Counterbalancing of word lists and hearing aid conditions

Subjects (1-6)

Hearing Aid Condition	LQ	LN	B	T
Word list	1	2	3	4

Subjects (7-12)

Hearing Aid Condition	T	B	LN	LQ
Word list	2	1	4	3

Subjects (13-18)

Hearing Aid Conditions	B	T	LQ	LN
Word List	4	3	2	1

Subjects (19-24)

Hearing Aid Condition	LN	LQ	T	B
Word List	3	4	1	2