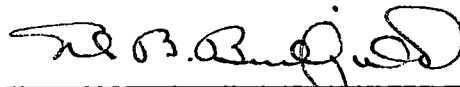


To the Graduate Council:

We are submitting herewith a dissertation written by Melissa Lynn Norwood entitled "Comparison of hearing aids and assistive listening devices with omni-directional and directional microphones at personal communication distances." We have examined the final copy of this dissertation for form and content and recommend that it be accepted in partial fulfillment of the requirements for the degree of Doctor of Philosophy, with a major in Speech and Hearing Science.

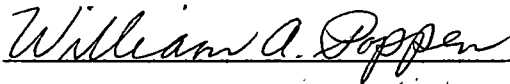
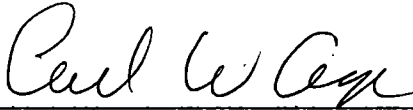


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We have read this dissertation
and recommend its acceptance:



Accepted for the Council:



Associate Vice Chancellor and
Dean of the Graduate School

**Comparison of Hearing Aids and Assistive Listening Devices
with Omni-directional and Directional Microphones
at Personal Communication Distances**

A Dissertation
Presented for the
Doctorate of Philosophy
Degree
The University of Tennessee, Knoxville

Melissa Lynn Norwood

May, 2000

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DEDICATION

This project is dedicated to my mother, Jane.

She has shown me that through perseverance and determination one can accomplish any goal. No matter how great the task or how far reaching the star, not shying away in the face of adversity, allows one to see their own strength.

Thank you for your endless support and unconditional love.

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To Shannon Frugia, you have been my sounding board, my computer expert and friend. Thank you for allowing me to sit in your office and relieve my frustrations and for helping out when the computer freaked out AGAIN! May the chocolate jar NEVER be empty!

To my committee members, I am truly grateful for your efforts to condense this project. The original project would still be a work in progress. The amount of data collected in the current condensed version was, at times, overwhelming, but I would always stop and think of what it could have been. Your efforts did not go unnoticed.

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YOU ARE "MY BEST FRIEND."

Look out world, Lynn is done!

ABSTRACT

This study was conducted to compare performance between binaural hearing aids and an ALD used in two modes (companion mode and interview mode). Performance was evaluated on the basis of word recognition at two distances and SNR at four distances. Additionally, this study was conducted to determine if directional microphones can be used to significantly improve performance with both hearing aids and an ALD using measures of word recognition and SNR. Word recognition performance measures indicated a rank ordering of the amplification devices with the poorest performance with the HA, improving with the ALDI and best when using the ALDC. The use of directional microphones improved word recognition for each device but the improvement was not sufficient to change the rank ordering of devices. As communication distance increased, the SNR decreased for three amplification devices. The measures of SNR found the least favorable SNR to be with the HA, improving with the ALDI and best with the ALDC. The use of directional microphones increased the SNR for the amplification devices over use of omni-directional microphones, however, the rank ordering did not change. Additionally, a critical SNR was established for the listeners in this study. Improving the SNR above the level of the SNR did not significantly improve word recognition performance regardless of amplification device or communication distance. These results provide clear evidence that the ALD can improve both word recognition performance and SNR more than HAs. While the use of directional microphones can be beneficial, the benefit of directional microphones in HA does not out-perform the benefit of directional

microphones in ALD. Additionally, the results indicate that the ALD can be used effectively in the interview mode at personal communication distances.

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CHAPTER ONE

INTRODUCTION

Listeners with sensorineural hearing impairment typically have greater difficulty comprehending speech in the presence of background noise, requiring as much as an 18-dB improvement in the signal-to-noise (SNR), to achieve word recognition scores equivalent to normal hearing listeners (Killion, 1997). Hearing aids are used to compensate for the hearing loss, however, they often do not improve the listener's ability to understand in background noise. There are several techniques that have been used to enhance performance in noise: noise suppression circuitry, directional microphones, and assistive listening devices (ALDs) using remote microphones. These techniques have been used to supplement omni-directional hearing aids as the standard form of amplification.

TECHNIQUES TO ENHANCE PERFORMANCE IN NOISE

Noise Suppression Circuitry. Noise suppression circuitry is an algorithm used to alter the frequency response within the amplifier of the hearing aid. This circuitry reduces output level primarily for low-frequency sounds. When used with omni-directional microphones in noisy environments the circuitry provides virtually no improvement in word understanding ability; however, when used with directional hearing aids there are small improvements (< 2%) in speech understanding (Valente, 1995). For this reason, some hearing aid manufacturers have elected to exclude this circuitry when using omni-directional microphones whereas with directional microphones the circuitry is used

automatically.

Directional Microphones. Enhanced microphone technology has enabled the development of hearing aids to improve the SNR reaching the listener's ear, by decreasing the amount of sound from sources to the side and behind the hearing aid user. This improvement can be expressed in terms of front-to-back ratio or directivity index. The front-to-back ratio is the difference (in dB) in the frequency response curves of the hearing aid measured at 0° and 180° azimuth. A limitation of this measure is that it only indicates the suppression of noise from directly behind the listener. A second measure, the directivity index is a single number ratio, based on the sensitivity at 0° azimuth compared to the sensitivity at every angle around the listener for a single frequency. The larger the directivity index, the greater the predicted ease for listening in difficult environments. A third measure is an articulation index-weighted directivity index based on the same measurement procedure used in the directivity index. The articulation-index provides different weights to the contribution each individual frequency band provides to the overall intelligibility of speech. In the articulation index-weighted directivity index procedure, the directivity index is weighted according to the weights provided by the articulation index. Because of the articulation index weighting, the more high frequencies used by the directional microphone, the greater the articulation index-weighted directivity index. As with the directivity index, the greater the articulation index, the better speech intelligibility in noise is predicted to be (Valente, 1998).

There are several types of microphone enhancement paradigms. The original design for the directional microphone was a single microphone with two sound inlet ports (front and back). In this system, an acoustic delay element in the rear port caused cancellations of signals arriving at the microphone when the sound source is towards the rear. This delay, sensed by the microphone as a difference in air pressure, caused an attenuation of the signals arriving at the back port. This reduction was typically limited to frequencies between 200 - 1500 Hz and provided an advantage of 2- to 4-dB for the directional microphone over an omni-directional microphone (Valente, 1998). The available data indicate that directional microphones can provide significant improvement in word understanding ability at unfavorable front-to-back ratios. Although the directional microphones improved word recognition, this improvement was substantially reduced as the reverberation times increased (Hawkins & Yacullo, 1984; Madison & Hawkins, 1983; Mueller & Johnson, 1979; Nielsen, 1974).

An improvement on the directional microphone was the incorporation of two matched omni-directional microphones in a front and back design, known as dual microphones. These microphones incorporate an electronic time-delay before their outputs are summed and applied to the pre-amplifier. The dual-microphone design provides a greater front-to-back ratio for signals arriving from the rear of the listener and extends the frequency response out to 2000 Hz as compared to an upper limit of 1500 Hz imposed by the previous directional microphones (Valente, 1998). The use of directional microphones and dual-microphones in hearing aids has been shown to improve SNR by as

much as 8 dB and improve word recognition performance by near 30% when compared to performance with an omni-directional microphone (Lurquin & Rafhay, 1996; Valente et al, 1995; Voss, 1997). Voss (1997) found an improvement of 29% in speech intelligibility at -15 dB SNR between hearing aids with an omni-directional microphone and hearing aids using a directional microphone when used in conjunction with a noise suppression algorithm for listening in background noise. The dual microphones have been incorporated into both hearing aids and ALDs.

The multiple-microphone array is an experimental design that can incorporate up to seventeen microphones, in either a broadside or end-fire configuration. The broadside configuration uses five microphones spaced 5.8 cm apart along the forehead of the user. The output from each microphone is weighted then summed. In the end-fire configuration, five microphones are spaced 2.0 cm apart along the side of the user's head. The output of the second to fifth microphones is time-delayed relative to the output of the first microphone. The output of all five microphones is then summed. Results for both broadside and end-fire configuration microphones indicate an improvement in directivity index ranging from 5 dB to 11.5 dB, depending on the microphone configuration (Soede, Berkhout, & Bilsen, 1993).

Another type of multiple-microphone technology is a beamforming microphone. The beamforming microphone enables the microphone in one hearing aid to communicate with the microphone in a second hearing aid. In this paradigm, the outputs of the

microphones are digitized and analyzed by Fast Fourier Transformation which is used to detect interaural signal differences, in phase and intensity between the hearing aids (Valente, 1998). Subjective quality rating data for beamforming microphones indicates a reduction in the perceived difficulty of listening in some situations (Schweitzer, Terry & Grim, 1996), however, this has not been quantified. This technology has not been incorporated into BTE hearing aids or ALDs with FM transmitters.

Assistive Listening Devices. Assistive listening devices are amplification devices designed to improve the SNR, by placing the microphone of the amplification system near the talker. The microphone and the receiver of these devices are coupled by a hard wire or by an infrared or FM transmission (Lewis, Feigin, Karasek, & Stelmachowicz, 1991). A limitation of the hard wire system is mobility restriction, in that the mobility of both the talker and the listener are restricted. The transmitter/receiver coupling system, which uses either frequency-modulated radio signals or infrared signals, does not restrict mobility in the same manner as the hard-wire system. When utilizing a FM system, the talker and listener can move freely about with little regard for distance up to 300 feet outdoors. The infrared system restricts movement, because the listener must be in the line of sight of the transmitter, and this system can not be used outdoors (Kaplan, 1987; Tyler & Schum, 1995). While movement is less restricted with infrared and FM systems, these systems are limited to input from one remote microphone. The use of a single remote microphone can be beneficial when there is input from one talker, such as in a classroom or church. In these settings, the remote microphone can be located in front of the talker or worn by the

talker as a lavalier microphone. What has been described in the literature is a general class of use in which the listener does not control the remote microphone, instead it is placed at a fixed position in front of the talker or worn by the talker. Throughout the present study, the term "companion mode" will be used to refer to the condition in which the remote microphone is under the control of the talker. When input is provided by more than one talker, such as in a conference room or on a theater stage, one or more fixed microphones can be used (fixed remote mode). In the fixed remote mode, unlike the companion mode, the enhancement ability of the transmitter/receiver system is limited due to the distance of the remote microphone from the talker (Bankoski & Ross, 1984; Brackett & Mandell, 1983; Kaplan, 1987; Nabelek & Donahue, 1986; Vaughn, Lightfoot & Teter, 1988). When using an ALD, it has been shown in several studies that there is an improvement in word recognition performance of up to 88% over performance with personal hearing aids in situations with varying amounts background noise (Hawkins, 1984; Nabelek & Donahue, 1986; Updike, 1994). This improvement is achieved by removing the microphone from the noise thereby improving the SNR.

When the ALD is used in the companion mode, the distance between the talker and the listener is typically beyond a personal communication distance, greater than 3 to 5 meters. Personal communication occurs when the distance between the talker and listener is less than 3 m. When the distances between the talker and the listener is greater than 3 meters, the communication is no longer interactive and is more informational, such as in classrooms, churches, conference rooms, or theaters. This can be described as one-way

communication, where the listener does not readily communicate with the talker. In the companion mode, the talker must take possession of the remote microphone, thereby indirectly accepting partial responsibility for the communication process. In contrast, hearing aids are designed for personal communication distances with complete responsibility lying within the hearing aid wearer.

One study evaluated communication using an ALD at a personal communication distance. Jerger, Chmiel, Florin, Pirozzolo & Wilson (1996) compared a monaural digitally programmable hearing aid and a monaural body-worn ALD, personal FM system, with a lavalier microphone in the companion mode for elderly listeners with a mild-to-moderate sensorineural hearing impairment. The performance of each system was evaluated with measures of speech perception and patient-perceived handicap. Speech perception was evaluated using monaural omni-directional amplification with the stimulus speaker located at one meter and the noise speaker mounted on the ceiling above the subject's head. The stimulus words were presented at 60, 70 and 80 dB SPL. When evaluating the ALD, the remote microphone was placed 6 inches from the stimulus loudspeaker. A constant environmental SNR of 8 dB near the listener's ear was maintained for all conditions. Word recognition in noise was 22% better when using the ALD, compared to performance with the hearing aid alone for new users and was 35% better for previous hearing aid users. Despite the difference in word recognition ability, the two devices were perceived to have similar improvement in overall patient-perceived benefit (5%). These authors concluded that the use of an assistive listening device can

effectively enhance word understanding ability and reduce patient-perceived handicap, however, the majority of the subjects evaluated in this study, chose not to purchase the "more effective" amplification system. The authors believed that one explanation for this perception might be the contrast in the ease of use between the two systems. The ALD system in Jerger's study used a large transmitter, a microphone (which is held by the talker) connected to the transmitter by a length of wire, a large receiver, and an ear-level transducer connected to the receiver by a length of wire, as compared to the one item of the personal hearing aid system.

An ALD can be incorporated with a behind-the-ear hearing aid, by using direct-audio boots attached to the bottom of the hearing aids. The direct-audio boots house the receiver component for the ALD, while a separate remote microphone contains both the microphone and the FM transmitter. When the FM system is used in conjunction with the hearing aids, the listener is able to receive auditory information as they normally would from their hearing aids and the remote microphone is placed near the talker increases the level of speech produced by the talker, thereby improving the SNR. In this mode of operation, the same signal is delivered to each aid and the signal is diotic rather than dichotic. A hand-held remote microphone (5" X 1 1/4" X 3/4") device (Phonak, HandyMic) was designed to be used with the direct-audio boots (Phonak, MicroLINK). This device may provide significant improvements in performance for listeners who require very favorable SNRs. The remote microphone can be hand-held by the talker (companion mode) or hand-held by the listener (interview mode). In the interview mode,

the listener can use the remote microphone much like a reporter, where the listener points the remote microphone in the direction of the talker that he wishes to hear. The interview mode permits two-way communication, in which both the talker and the hearing-impaired listener engage in communication with the advantage that the listener controls the microphone rather than the talker as in the companion mode. The ALD can be used with either omni-directional or directional microphones. The ALD also has an additional highly focused directional microphone. When the remote microphone is held by the listener, the microphone can be switched from omni-directional to a narrower focused directional microphone, allowing the listener to focus on more of the conversation taking place in the direction of the microphone. If the remote microphone is switched to highly-focused, the focus becomes narrower and only the sounds directly in line with the remote microphone will be received.

RATIONALE FOR THE STUDY AND RESEARCH QUESTIONS

The purpose of this study was to compare the performance of devices that improve SNR: binaural BTE hearing aids and the same hearing aids coupled to an ALD. Comparisons were made under four listening conditions that were considered to be representative of conditions that a listener might encounter in everyday life. The performance of hearing aids and an ALD with a remote microphone were compared on measures of SNR and word recognition in background noise. The present study was similar to the study done by Jerger et al (1996), in which performance with hearing aids was compared to performance with an ALD, however, the present study was conducted at personal communication distances that ranged from 1 - 2.5 m. In contrast to the Jerger

study, binaural BTEs were used in this study rather than monaural hearing aids as would be expected for a listener with sufficient hearing loss to consider the need to use an ALD. In addition, the effects of directional microphones combined with noise suppression circuitry was evaluated for both hearing aids and ALDs.

In the Jerger et al (1996) study in the remote microphone was worn by the talker. In the present study, performance with the ALD was evaluated in two modes, listener held remote microphone -- interview mode (ALDI) and talker-held remote microphone -- companion mode (ALDC).

Previous research has shown that hearing aids function well at distances below 1 m and poorly over a distance of 3 m, whereas ALDs in the companion mode perform efficiently at distances greater than 3 m, but have not been tested at distances less than 3 m (Bankoski & Ross, 1984; Dirks & Wilson, 1969b; Hawkins, 1984; Hawkins & Yacullo, 1984; Nabelek & Donahue, 1986; Nabelek, Letowski, & Donahue, 1986). ALD performance in the interview mode has not been evaluated.

In the present study, performance of binaural hearing aids and ALDs were evaluated using measures of word recognition ability at talker-listener distances of 1 and 2 meters. The acoustical performance of these devices was evaluated in terms of the ability of the devices to alter signal and noise levels near the tympanic membrane. The following questions were addressed:

1. Is there a significant difference in word recognition with binaural hearing aids versus an ALD used in either the interview or companion mode?

2. Is performance, measured as word recognition ability, related to signal level, noise level, or signal-to-noise ratio for each of the devices?
3. Does performance, measured as word recognition ability, differ for directional microphones and omni-directional microphones?

CHAPTER TWO

REVIEW OF THE LITERATURE

DEFINITION OF TERMS

Binaural hearing refers to hearing with two ears. The maximum advantage of binaural hearing can be expected to operate only when differences of phase, amplitude, and/or timing exists. Diotic listening refers to a situation where the identical signal is applied to both ears, such as through Y-cord listening. Y-cord listening is where one microphone feeds one amplifier and two receivers, such as through a stethoscope. Dichotic listening involves both ears listening to different signals. Binaural listening is dichotic, involving two ears or two completely separate hearing aid transmission systems. The term monophonic is typically used to describe a listening situation employing one sound source, whereas stereophonic describes a listening situation employing two or more sound sources, such as loudspeakers (Harris, 1979; Levitt & Voroba, 1974; Tobias, 1972). In the present study, both types of amplification utilized can be classified as binaural, however, listening via hearing aids can be termed dichotic and stereophonic listening, whereas listening through the assistive listening device is diotic and monophonic.

WORD RECOGNITION PERFORMANCE

Several recent studies have presented data on listener's performance on word recognition tasks in quiet and in the presence of background noise (Norwood-Chapman, 1997; Souza & Turner, 1994; Sperry, Wiley, & Chial, 1997; Wilson, Oyler & Sumrall,

1995; Wilson, Zizz, Shanks, & Causey, 1990). Based on the data from Norwood-Chapman, 1997, Wilson et al, 1995, and Wilson et al 1990, word recognition performance in quiet ranged from 20% at 30 dB SPL to 80% at 45 dB SPL and near 100% at levels 55 dB SPL and above for listeners with normal hearing. The difference between the 20% and 80% correct recognition points, used to estimate the slope of the psychometric functions, indicated a slope of 4.5%/dB. Performance by listeners with mild to moderate hearing impairment was similar to the listeners with normal hearing, however, the 80% point was not obtained until 63 dB SPL and near 100% until 90 dB SPL. Levels below 55 dB SPL were not tested with the hearing-impaired listeners, therefore estimates of 20% correct recognition and the slope of the psychometric function could not be obtained.

When listening in the presence of background noise, performance by normal hearing listeners ranges from 20% at -3 dB SNR to 80% at 7-8 dB SNR. The slope of the psychometric functions varied from 4.6%/dB to 5.4%/dB (Norwood-Chapman, 1997; Sperry et al, 1997). For listeners with hearing impairment, performance ranged from 20% at 2 dB SNR to 80% at 12 dB SNR with a slope of 5.5%/dB (Norwood-Chapman, 1997). These results indicate that the slope of the word recognition in multi-talker babble function for hearing impaired listeners was nearly identical to that of the normal hearing listeners, however, the function is shifted by 5-dB. The results indicate that the subjects with hearing impairment require a 5-dB greater SNR than the normal hearing listeners to reach the same performance level. Although, this information is not known for listeners with

moderate to severe hearing loss, the same shift in required SNR for equal performance is anticipated.

Souza and Turner (1994) evaluated word recognition under four types of background noise, including a twelve speaker multi-talker babble, in normal hearing, young hearing-impaired and elderly hearing-impaired subjects. The use of high-pass noise for all test stimuli was included to compensate for differences in high-frequency audiometric thresholds between the young hearing-impaired and the elderly hearing-impaired subjects. These investigators found that word recognition performance in high-pass masking noise alone for all subjects was near 90% or better, however, the addition of multi-talker babble decreased word recognition scores to 55% for normal hearing listeners and 35% for hearing-impaired listeners at 5-dB SNR.

Word recognition in noise is evaluated most completely by using a range of SNRs for conditions in which the competing message has no effect on word recognition to the conditions in which the competing message completely overwhelms the stimulus. Several researchers have proposed the use of evaluating word recognition under a variety of SNRs (Beattie, 1989; Cooper & Cutts, 1971; Dirks et al, 1982). Investigators have used a constant stimulus level and varied the level of the competing message (Dirks et al, 1982; Jerger & Hayes, 1976; Plomp, 1986) or varied the stimulus level and used a constant level for the competing message (Beattie, 1989; Cooper & Cutts, 1971; Dirks et al, 1982; Norwood-Chapman, 1997; Sperry et al, 1997; Wilson et al, 1990). A third method used

has been to maintain the SNR while varying the sound pressure level of both the stimulus and the competing message (Beattie, 1989; Hall, 1983; Jerger & Hayes, 1977; Schwartz, Surr, Montgomery, Prosek & Walden, 1979). The preferred procedure is dependent on the examiner's purpose. In the present study, the signal and noise levels remain constant, however, the SNR is varied by the distance from the microphone of the amplification system to the stimulus and noise loudspeakers, which is dependent on the type of amplification system being utilized. See Appendix B for a detailed description of signal and noise levels by distance and amplification system. Calibration of the signal levels is based on the average level of the carrier phrase, so that the calibration tone VU meter reading is equal to 0-dB. The noise level calibration is set so that the average speech level of the multi-talker babble has a VU meter reading of 0-dB. This same calibration procedure was used for the Wilson et al, 1990, Sperry et al, 1997, and Norwood-Chapman, 1997 data.

Word Recognition with Reduced Cues. Word recognition performance is not only affected by background noise, but also by reverberation of sounds in the environment. In a room, sound propagates from the source until it is absorbed by materials and air in the room. As the direct sound travels to the listener's ear, the remaining sound is reflected off walls, tables, other people in the room and other material within the room. According to Nabelek & Nabelek (1993), the early reflections in a reverberant room reach the listener's ear as early as ten milliseconds after the initial onset of the signal. Early reflections affect the quality of the signal as they blend with direct sound. Multiple reflections follow with

various intensities and delays, some of which are closely spaced falling almost or completely on top of each other and are perceived as noise. Reverberation time (T) is the amount of time required for the sound pressure level of the signal to decrease 60-dB after the sound source ceases (ANSI - S1.1, Revised 1976). As reverberation time increases, word recognition decreases (Gelfand & Hochberg, 1976; Moncur & Dirks, 1967). The amount of absorption within a room also effects reverberation. The more absorption, the shorter the reverberation times. Anechoic chambers, which contain extraordinary amounts of absorptive materials, have reverberation times of zero. Small rooms and offices have reverberation times between 0.4 sec and 0.8 sec whereas large classrooms, auditoriums, and churches have reverberation times up to 1.5 sec (Nabelek & Nabelek, 1993). According to Nabelek (1980) normal-hearing listeners can tolerate reverberation times of up to 1.2 sec without adverse effects on word recognition, however, acceptable reverberation times are significantly lower for hearing-impaired listeners. In the present study, the experimental measures are conducted in a sound-treated booth with near zero reverberation times.

Many classroom studies involving the parameters of noise and reverberation have been conducted with children. In one such study, the effects of room acoustics on monosyllabic word recognition abilities in children were studied by Finitzo-Hieber and Tillman (1978). The listening situations selected were designed to simulate acoustical conditions found in educational environments. Twelve different reverberation and noise combinations were studied for normal hearing and hearing-impaired listeners. These

investigators found that as the reverberation times increased, the word recognition of both subject groups was significantly degraded, with a greater degradation in word recognition abilities in the hearing-impaired group.

Crum (1974) examined how well normally-hearing adults performed on the monosyllabic word intelligibility tasks in various reverberant conditions. Four reverberation times (0, 0.4, 0.8 and 1.2 sec) were evaluated at three-speaker-to-listener distances (6, 12 and 24 feet). Crum found that the longest reverberation times decreased intelligibility in the normally-hearing adults, however, the speech understanding abilities were not significantly affected by the shorter reverberation times at any distance. In an effort to overcome the difficulties encountered due to reverberation, group amplification systems have been recommended (Nabelek & Robinson, 1982).

In addition to reverberation, the distance between the speaker and the listener may effect word recognition performance. Peutz (1971) evaluated word recognition at various distances from the sound source using different volume levels and reverberation times, while keeping the level of the speech constant. This investigator found that up to 1 meter from the sound source, word recognition performance is quite good, however, performance declines until a certain distance is reached. At this "critical distance" word recognition performance remains constant, despite additional changes in distance or reverberation times.

HEARING AIDS

Personal hearing aids are comprised of four main components: (1) microphone, (2) amplifier, (3) receiver, and (4) power supply. Each component is responsible for one particular aspect of sound transmission from unamplified to amplified signal. The location of each of these components within the hearing aid is dependent on the style of hearing aid. Sound transmission can be affected by differences in the location of the microphone on the hearing aid. For behind-the-ear (BTE) amplification, the microphone is located on the superior portion of the hearing aid, behind the pinna (Hull, 1993). The sound pressure level (SPL) measured at a variety of points around the head can be affected by diffraction of sound of the head and by resonances created within the external ear. The largest change that occurs to sound as it progresses from the free field to the BTE microphone location (above the ear), is 3-dB at 2000 Hz, due to head diffraction effects. Larger changes are seen with in-the-ear style hearing aids, with SPL increases up to 12-dB for the frequency region 3000 to 6000 Hz (Kuhn, 1979; Sullivan, 1989). The importance of this finding is that more gain is required for behind-the-ear hearing aids to achieve the same desired gain as other style hearing aids, due to the microphone location within the hearing aid.

Additional increases in SPL occurring at the hearing aid microphone may be dependent on the direction of the sound reference to the hearing aid microphone. For sounds originating from the same side as the hearing aid microphone, an increase of 5- to 7-dB occurs through the middle frequencies, however, a decrease occurs in the higher

frequencies due to the head baffle effect. Conversely, for sounds originating from the opposite side of the head as the hearing aid microphone, a significant reduction in the middle and high frequencies, reaching 10-dB at 4000 Hz occurs due to the head shadow effect (Dirks & Wilson, 1969; Hawkins, Foley, Stelmachowicz & New, 1992).

Fitting Procedures. Regardless of the type or style of amplification system being utilized, an assessment of the amplification system in regards to the listener's hearing loss must occur. Since the invention of the first hearing aid, techniques for hearing aid evaluation have been studied. Curran (1988) stated that no single hearing aid evaluation procedure can withstand close scrutiny because limitations to each method can be identified. Mueller, Hawkins, and Northern (1992) proposed four general guidelines for evaluating hearing aids: (1) the measurement should permit rapid re-evaluation each time the hearing aid is adjusted electro-acoustically; (2) the measurement should be sensitive enough to note small changes in hearing aid characteristics created when acoustical modifications are made in acoustic coupling system; (3) high objectivity and good reliability of the measurement; and (4) an estimate of the validity of the hearing aid selected. Two approaches, comparative or prescriptive, are typically used for evaluating the hearing aid (Davies & Mueller, 1987). In the present study, the prescriptive approach will be used to fit the hearing aids to the individual hearing loss characteristics. Additionally, the comparative approach will be used during the one-month field trial for each amplification system.

During a prescriptive fitting, electroacoustical measurements are made using a computerized probe-microphone measurement system. These measurements utilize a miniaturized, self-calibrating microphone with a soft silicone probe-tube extension for making measurements at the tympanic membrane. The acoustic measurements performed in the ear canal, without and with amplification devices can be used to obtain information about the influences of the amplification device, the impedance characteristics of the ear anatomy, the natural resonance of the ear canal, and the spectrum of sound energy reaching the eardrum. These measurements can also provide immediate feedback in regards to how a sound is transmitted from the source, through the amplification device and ear canal, to the tympanic membrane, allowing the system user to better understand the individual effects of the amplification device (Mueller, Hawkins & Northern, 1992).

The National Acoustics Laboratories (NAL) developed a prescriptive fitting formula based on pure-tone threshold measures with the rationale that the prescriptive gain should amplify the long-term speech spectrum so that it is comfortably and equally loud across all frequencies. The speech signal is shaped so that each frequency band contributes equally to its loudness. To determine the desired gain, threshold values are subtracted from most comfortable level values, and gain is increased by 4.6 dB for each 10 dB of hearing loss. Additionally two correction factors are used, the first for loudness differences across frequency with the 60-phon line as a reference and secondly, shaping based on the long-term speech spectrum (Byrne & Dillon, 1986). There is also a correction factor for hearing aid style, due to the difference in hearing aid microphone

location between behind-the-ear hearing aids and in-the-ear hearing aids. In a study of subjective evaluation, Byrne and Cotton (1988) compared the NAL prescribed gain procedure with alternate frequency responses represented of other fitting strategies. These investigators found that subjects preferred the NAL responses in terms of speech intelligibility and pleasantness of sound quality.

Although prescriptive fitting rationales were developed for use with linear hearing aid circuits, these fitting formula have been used with non-linear amplification provided that several issues are addressed prior to the fitting hearing aids with non-linear circuitry. First, the prescriptive target gain used for non-linear hearing aids may vary as a function of the input signal and whether the insertion response matches the target gain depends on the input signal selected. Secondly, for hearing aids that use input-controlled gain, the configuration of the insertion response can vary as a function of the input level and the compression kneepoint (Preves, Beck, Burnett, & Teder, 1989). Mueller, Hawkins and Northern (1992) recommend evaluating non-linear hearing aids with two input levels, the first below the threshold kneepoint and secondly with the hearing aid in compression.

Microphone Directionality. A directional microphone is designed to be more sensitive to sounds arriving from the direction the listener is looking and decreasingly sensitive to sounds as the angle increases to either side of the head and with minimum sensitivity for sounds arriving from behind the listener. However, the pattern of directionality may be affected by head reflection and reverberation (Weiss & Neumann,

1993). Figure 1 illustrates the polar sensitivity patterns for omni-directional and directional microphones in free field and when placed on KEMAR. In free field, a noticeable difference, near 20-dB at 180° azimuth, in the polar sensitivity pattern occurs between the omni-directional and directional microphones, however, this does not take head and torso effects into account. The polar pattern when placed on KEMAR shows a decrease in the omni-directional as well as the directional microphone, but the pattern is further reduced for the directional microphone when sounds are produced from the back of the listener.

Several investigators have evaluated the effects of an omni-directional microphone and a directional microphone with monaural and binaural hearing aids (Buerkli-Halevy, 1986; Hawkins & Yacullo, 1984; Madison & Hawkins, 1983; Weiss & Neuman, 1993). Studebaker, Cox and Formby (1980) found that the use of directional microphones could reduce the noise level by up to 20-dB in rooms with little or no reverberation, yet in highly reverberant rooms, the use of the directional microphone did not produce a significant reduction in the noise level.

Hawkins and Yacullo (1984) evaluated the word recognition performance in multi-talker babble for monaural and binaural hearing aids using omni-directional and directional microphones in three levels of reverberation. These investigators found a significant binaural advantage (2-3 dB SNR for 50% criterion performance) that was independent of microphone type or reverberation level. Secondly, these investigators found a significant

directional microphone advantage (3-4 dB improvement in SNR) that was independent of hearing aid arrangement (monaural or binaural), however, it was dependent on the reverberation level. The longer the reverberation time the less directionality advantage seen, indicating that directional microphones operate optimally in room with short reverberation times. These investigators suggest that to best control the acoustic environment, the listener should minimize the effects of reverberation by reducing the speaker to listener distance, thereby increasing the SNR. Similar results were found in a recent study by Valente, Fabry, and Potts (1996). In this study, the traditional directional microphone with front and back facing ports was replaced with two omni-directional microphones - one facing forward and one facing backwards. These investigators found a 7-8 dB improvement in SNR when using a directional microphone over an omni-directional microphone. One explanation for the nearly double improvement in SNR findings from that of Hawkins and Yacullo was the difference in amplification between the front microphone and the back microphone within the hearing aid, also referred to as the front-to-back ratio (FBR). In the Hawkins study, the FBR was 8, 13, 12, 10, and 2 dB at 500, 1000, 2000, 3000, and 4000 Hz, respectively, whereas in the Valente et al study, the FBR was 27, 20, 20, 20 and 12 dB. The directional microphone used in Valente's study provides more attenuation of signals arriving from the rear than the microphone in Hawkins study. Despite the difference in the two microphones, the end result is that the use of directional microphones can improve SNR.

Voss (1997) evaluated multi-microphone hearing aids when used in the omni-directional and directional modes. In this study, the hearing aids were programmed for three listening modes: (1) quiet - omni-directional, (2) noise - omni-directional, and (3) noise - directional. Unlike the Hawkins et al study which presented the speech at 0° azimuth and noise at 180° azimuth, Voss used speech at 0° azimuth and noise from four speaker each at a 45° azimuth circling the listener. He found a 15% improvement in word recognition using the noise algorithm with the directional microphone over the quiet program with the omni-directional microphone at a -10 dB SNR. At -15 dB SNR, a difference of 29% improvement in word recognition was observed when using the noise algorithm and directional microphone over the quiet algorithm and omni-directional microphone. When using the noise algorithm and comparing the differences between directional and omni-directional microphones alone, the word recognition scores are less defined, 9% at -10 dB SNR and 22% at -15 dB SNR. The subjective evaluation accompanying this study indicated that although the noise algorithm and omni-directional microphone “was comfortable”, the noise algorithm used in conjunction with the directional microphone provided the subjects with a clearer sound, “making speech more intelligible in noise”.

Kuk (1996) compared subjective performance ratings for listeners when allowed to switch from one mode of directionality to the other. He found that subjects preferred using the omni-directional microphone in quiet listening situations, such as television or radio, and preferred the directional microphone in noisy situations, such as restaurants or

in group situations. Additionally, he found a preference for omni-directional microphone use with various non-speech sounds, such as music or warning signal, indicating a need for using different types of microphones in different types of listening situations.

Comparison of Performance with Monaural versus Binaural Hearing Aids.

There have been numerous studies that have compared performance with one hearing aid versus two hearing aids (Bronkhorst & Plomp, 1989; Dirks & Wilson, 1969; Hawkins, 1984; Heyes & Ferris, 1975; Nabelek & Pickett, 1974; Naidoo & Hawkins, 1997; Tillman, Carhart, & Olsen, 1970; Zelnick, 1970). Several advantages of binaural listening include increased localization skills and improved word recognition in noise.

Heyes & Ferris (1975), evaluated the ability to localize to a sound source with monaural and binaural ear-level and a body-style hearing aids. They found that, at low signal levels, subjects wearing binaural ear-level hearing aids showed a superior ability to determine the location of the sound source over the monaural hearing aid condition. At high signal levels, localization ability with monaural hearing aids approximates that of binaural hearing aids. Dermody & Bryne (1975) reported that the difference between binaural and monaural aided localization is critically dependent on the level of the stimulus signal, for example at low intensity levels (10 dB SL) the difference in loudness ranged from 5 to 8-dB, whereas at high intensity levels (40 dB SL) the difference in loudness ranged from 10 to 15 dB. One explanation offered is that at high signal levels the interaural time difference is more important than interaural intensity difference for

localization. In the present study, subjects are fitted with binaural hearing aids and a binaural ALD that utilizes diotic listening. Although localization effects are not being directly evaluated by the experimental measures, this is evaluated by the subjective performance.

One arrangement that has proven to be successful in demonstrating the binaural advantage, with respect to word recognition, is one in which the speech stimulus is presented from one loudspeaker and a competing noise is presented from a separate loudspeaker (Bryne, 1980). This arrangement situates the listener in the center of a sound-treated room seated several feet from the each of the two loudspeakers, which are placed at 45 degrees to either side of the listener's head. When comparing monaural and binaural listening, six different test conditions are possible, including measures of near-ear and far-ear conditions. A comparison of performance under each of these six conditions can provide the researcher with information regarding binaural squelch effect, and head shadow effect, as well as an effect of monaural versus binaural performance. The binaural 'squelch' effect is evident if performance with binaural hearing aids is superior to performance under the most favorable monaural condition. The degree of 'squelch' can be calculated by subtracting the monaural near-ear score from the binaural near-ear score. An estimate of the head shadow effect can be obtained by subtracting the speech score obtained under the less favorable monaural far-ear condition from the score obtained for the monaural near-ear condition. The binaural 'squelch' and the absence of the head

shadow effect have been used to explain better performance with binaural hearing aids as compared to monaural performance (Libby, 1980).

Dirks and Wilson (1969b) evaluated word recognition in noise with monaural and binaural ear-level hearing aids for listeners with normal hearing and listeners with mild to moderate hearing loss. Word recognition ability was obtained from tape recordings of the speech stimuli and the competing message stimuli via monaural and binaural hearing aids positioned on an artificial head. These investigators varied the SNR from -30 dB to 10 dB with (1) speech and noise at 0° azimuth and (2) with speech at 90° azimuth and noise at 0° azimuth. No significant binaural advantage was found when speech and noise are produced from the same loudspeaker. The mean SNR difference for 50% criterion performance was less than 1-dB between monaural listening and binaural listening. A significant binaural advantage was found for performance with monaural and binaural hearing aids at all SNRs when speech and noise are produced by different loudspeakers. The greatest difference was obtained between the monaural far ear scores and monaural near ear scores, where at 50% criterion performance, normal hearing listeners averaged scores of 5.5-dB poorer and listeners with hearing impairment averaged 6-dB poorer in the far ear condition. Dirks and Wilson concluded that individuals with sensorineural hearing loss utilize interaural time differences between the speech stimuli and the competing message stimuli for a binaural advantage. These investigators further concluded that the binaural unaided advantage, as obtained in previous studies, can be extended to aided conditions.

Hearing aid performance has been shown to decrease in the presence of noise and reverberation. Finitzo-Hieber and Tillman (1978) evaluated hearing-impaired children without and with a moderate gain hearing aid. The results indicated that hearing-impaired children experienced greater degradation in speech understanding with their hearing aids, a decrease of 38% when the reverberation increased from 0 to 1.2 seconds and a decrease of 33% at 6-dB SNR, as compared to without their hearing aids.

Nabelek and Pickett (1974) evaluated the performance of normal hearing and moderately-severe hearing impaired subjects under monaural aided and binaural aided listening conditions in quiet, in the presence of background noise and in reverberation. These investigators found that on average in quiet and at a SNR of 10 dB, the hearing-impaired subjects performed 4.5% better with a reverberation time of 0.3 seconds and 3.6% better with reverberation time of 0.6 seconds under binaural listening than in the monaural conditions. Word recognition performance was significantly reduced as SNR levels became more negative (5 and 0-dB) for hearing-impaired listeners, however, these SNR levels did not have a significant effect on normal-hearing subjects. These investigators concluded that binaural hearing aids can improve ability to understand speech in quiet and in noise.

In a study by Hawkins (1984), the effects of hearing aid use in background noise with children was evaluated. Hawkins found that hearing aid performance in noisy

environments was best at SNRs greater than 10 dB. In this study, Hawkins also studied the effects of noise on speech recognition with monaural and binaural hearing aids. The results indicated that monosyllabic word recognition scores were 20% better with binaural hearing aids at 15 dB SNR, however, at the lower SNR (6 dB), word recognition performance for the binaural hearing aids decreased dramatically. Additionally, the use of omni-directional and directional microphones in the monaural and binaural hearing aid conditions were evaluated. Word recognition was significantly better 2.8% and 3.5% for the binaural directional hearing aids than with binaural omni-directional and monaural omni-directional hearing aids, respectively. No significant difference (0.9% difference) was found when comparing the monaural directional and binaural directional hearing aids. This investigator concluded that word recognition performance is best with binaural hearing aids with directional microphones.

In the previously mentioned study, Tillman et al (1970) found that noise adversely affected speech intelligibility in subjects with and without hearing loss. These investigators found that when hearing-impaired subjects were evaluated without and with their hearing aids and in the presence of background noise, all subject groups performed better on the speech task without amplification. Speech understanding scores were the lowest when hearing aids were used in the noisiest condition (6 dB S/N), indicating the adverse effects of background noise were compounded by hearing aids.

ASSISTIVE LISTENING DEVICES

Assistive listening devices (ALDs) are amplification systems which are designed to improve the SNR, in many cases by 15-dB or more. Enhancement of the SNR is accomplished by placing the microphone of the amplification system near the sound source. The signal is transmitted from the microphone to a receiver worn by the listener, thereby improving the SNR (Lewis, Feign, Karasek, & Stelmachowicz, 1991). This signal can be transmitted through wires, frequency modulated radio waves, audio-induction coupling or infrared light waves. One of the most commonly used assistive listening device is the frequency modulated (FM) system. With this system, the signal is transmitted via low frequency radio waves from the remote microphone to the receiver. FM systems have in the past been recommended for use only in educational settings and only for children with severe to profound hearing losses, however, these systems are now being used by listeners with varying degrees of hearing loss, listeners of all ages and in noneducational settings, such as cinemas, theaters, museums and other public buildings (Bankoski & Ross, 1984; Brackett & Mandell, 1983; Kaplan, 1987; Nabelek & Donahue, 1986; Vaughn, Lightfoot & Teter, 1988). Most personal ALDs are capable of delivering the signal to the listener in two ways: (1) directly to the ear through the use of ear level transducers (earbuds, headsets, or button-type receivers) or (2) by coupling the ALD receiver to a personal hearing aid. Several methods exist for coupling an ALD and a personal hearing aid including direct audio input, a neck loop, or silhouette inductor.

Fitting Procedures. Evaluation of assistive listening devices, specifically FM systems, are similar to those used with hearing aids. Currently, there is not a standard electroacoustic measurement procedure for evaluating the basic performance of FM systems, therefore, most manufacturers follow the procedures set forth by ANSI S3.22 (1987) for electroacoustical measurements of hearing aids. Although the basic principle of amplification, is the same between the hearing aids and ALDs, to make sounds louder, there are electroacoustical differences that must be taken into consideration prior to evaluating the FM system. The first is the input level used to measure electro-acoustical fit of the system. Since the remote microphone of the FM system is approximately 6 to 8 inches from the speakers mouth, the resulting input to the FM system is 80 to 85 dB SPL, instead of the 60 dB SPL typically used for measuring hearing aid performance (Hawkins, 1984). There may also be changes in the frequency response of a hearing aid when coupled to a FM system. There is not one coupling method that preserves the hearing aid frequency response (Hawkins & Schum, 1985), however, one study found that the use of direct audio-input as a means of coupling a FM system to a hearing aid may produce the most similar frequency response (Hawkins & Van Tasell, 1982). With the introduction of pen-sized remote microphones, the fitting procedures for ALDs may need to be altered to incorporate the interview mode in the evaluation as well as the companion mode.

FM Microphone Placement. Placing the FM microphone/transmitter close to the loudspeaker to simulate the distance of the microphone from the speaker's mouth has been recommended for more accurate fitting evaluations (Flexer et al, 1987; Hawkins, 1984;

Picard & LeFrancois, 1986; Ross & Giolas, 1971). Although this is an option, an additional calibration to ensure that the expected input level occurs at a particular distance must be made. Lewis, Feign, Karasek, and Stelmachowicz (1991) report that the distance and the height of the remote microphone in front of the loudspeaker can alter the level of the expected output. Measurements were made for a signal adjusted to be 60 dB SPL at a distance of 1 meter and a height of 42 inches. The output was measured at distances of 6 and 8 inches in front of the loudspeaker and at heights of 24, 30, 36 and 42 inches from the floor to the center diaphragm of the loudspeaker. The output measured rarely met expected level and varied across frequencies, 250 to 8000 Hz. The output at a height of 30 inches deviated the least from expected values, whereas the output at 42 inches, where the initial measurement was made, deviated the most.

Word Recognition. Nabelek, Donahue & Letowski (1986) compared speech perception in quiet and in noise for three types of assistive listening devices in a medium-size classroom. These devices were evaluated using (1) young, normal-hearing adults, (2) middle-age mild-to-moderately hearing-impaired adults, (3) moderately hearing-impaired adults utilizing hearing aids, and (4) elderly mildly hearing-impaired subjects. The three types of ALDs evaluated were (1) audio induction loop, (2) FM system and (3) infrared (IR) light system. Additionally, these results were compared with speech perception using the public address (PA) system. These researchers found (1) no significant differences in speech perception for the three listening systems; (2) a difference of 3% between performance using the FM system and the IR system for both hearing-

impaired subjects, and hearing aid users. These results are statistically significant, however, the authors reported that these should not be considered clinically significant.

In a similar study, Nabelek, & Donahue (1986), evaluated the speech perception performance of ALDs in an auditorium. The results of this study indicated that a small, (3%), statistically significant difference is shown between FM systems and IR systems, however, a large (26%) difference is reported between the ALDs and the PA system, for all listeners. These authors conclude that the use of an assistive listening device provides improved speech intelligibility for all listeners.

EVALUATION OF HEARING AID AND ALD PERFORMANCE

The two most commonly used amplification options in classroom setting including personal hearing aids and FM systems. Hawkins (1984) compared these two options by dividing the types of amplification into different categories. Hearing aids were studied in monaural versus binaural conditions with omni-directional and directional microphones. The FM system was evaluated separately and when coupled to each hearing aid category. Scores on the word recognition task indicated that subjects performed significantly better with the FM system used alone than with hearing aids alone or when coupled to the FM system. This investigator found that on average the FM-only condition provided a 15.3 dB advantage in SNR over the four hearing aid conditions. This advantage was only slightly less than the difference in the intensity of speech at the location of the teacher's microphone (84 dB SPL) and the child's hearing aid microphone at 2 meters (65 dB SPL).

Moving the microphone from the child's ear to the teacher's neck level, increased the SNR by 19 dB. This researcher also found that when the environmental microphone of the hearing aid is turned on, the FM advantage over the hearing aids drops from 32% to 12% at 6 dB SNR. Hawkins speculated that if less optimal classroom locations had been chosen (e.g. locations further from the speaker) the FM system alone would have shown to be even more beneficial.

In a study by Urdike (1994), a FM auditory training system was compared with Contralateral Routing of Signal (CROS) hearing aids and the personal hearing aids worn by six unilaterally hearing-impaired young children. This investigator found that with the FM system, children's scores improved significantly, as much as 20% to 88% in noise, over that of their unaided performance. When the performance in the FM system was compared with the other amplification systems, similar results were obtained in quiet and in noise with scores ranging from 0% in noise with personal amplification system to 100% in noise with the FM auditory trainer. The investigator also noted that the children with the greatest hearing loss were able to derive the most benefit from the FM auditory trainer.

Ross and Giolas (1971) evaluated the use of an FM system under normal classroom conditions. Thirteen children with hearing-impairments and eighteen age-matched normally hearing children were tested under three separate conditions: (1) the child's usual listening condition (unaided or aided); (2) binaural situation with an FM

receiver using only the environmental microphone of the receiver; and (3) a binaural listening condition using input from both the FM receiver and the wireless microphone/FM transmitter. These investigators found that the hearing aid only condition resulted in poorer word recognition scores (20.9%) as compared to no amplification (32%), yet, when FM systems were used in conjunction with the hearing aids, performance was significantly better (27% increase) for all subjects, including the normally-hearing subjects.

CHAPTER THREE

METHODS

The performance of hearing aids and the same hearing aids coupled to an ALD with a remote microphone were compared on measures of SNR and word recognition in background noise. The subject qualification, types of hearing instruments used, the experimental procedures and the experimental protocol are discussed in detail below.

GENERAL DESCRIPTION OF SUBJECTS

Data were obtained from 20 hearing-impaired subjects (11 males and 9 females), aged 45- to 65-years (mean age = 53.6 years), who were experienced amplification users. Subjects initially selected from information obtained in a review of clinical files from the Lamar University, Audiology Clinic. Subjects had a bilateral flat or gradually sloping moderate to severe sensorineural hearing loss for frequencies 500 Hz to 6000 Hz with air-bone gaps no greater than 10-dB. Subjects were required to have word recognition scores greater than 50% at their most comfortable listening level.

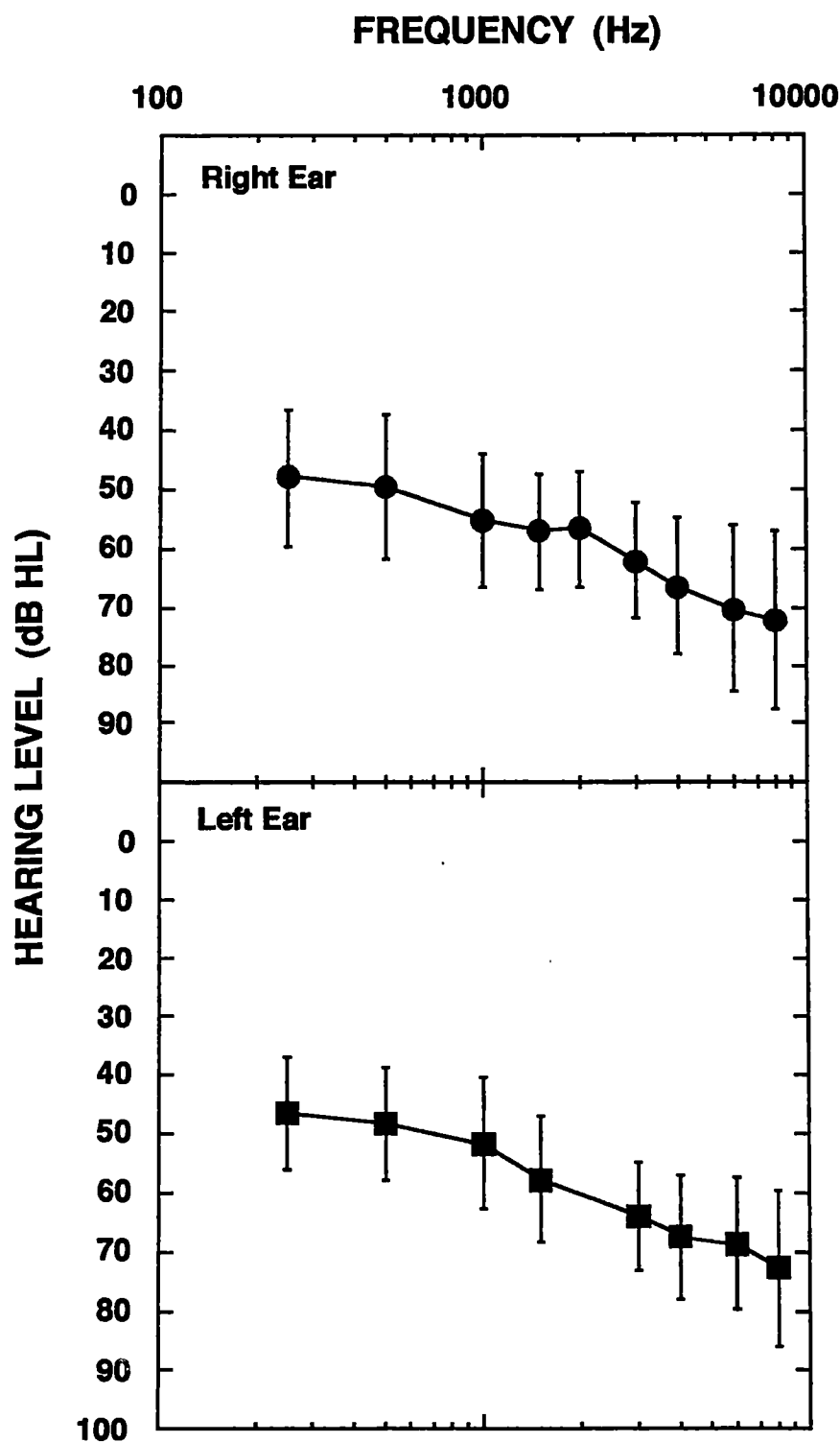
SUBJECT QUALIFICATION PROCEDURES AND CRITERIA

Otoscopic Inspection. An otoscopic inspection, using a hand-held otoscope (Welch-Allyn, Model 25020), was performed to assure that the ear canal was free from excessive wax and debris and that there was no evidence of external ear pathology.

Hearing Thresholds. Subjects were seated in a comfortable chair in a double-walled sound-treated booth (Industrial Acoustics Company, Model 1403) for the measurement of pure tone thresholds. Ambient noise levels were measured prior to and following experimental procedures to assure that noise levels did not exceed ANSI S3.42 - 1992. An audiometer (Grason-Stadler, Model 16) was used to deliver pure tone signals at the frequencies 0.25, 0.5, 1, 1.5, 2, 3, 4, 6, and 8 kHz to the subjects using earphones (Telephonics, TDH-50P) encased in supra-aural cushions (Model 51) and from a bone vibrator (Radioear, B-71) at the specified frequencies 0.5, 1, 2, and 4 kHz. Signal presentation level was varied in 5-dB steps using steady pure tones. The subjects responded by using a push button response indicator that was connected to the audiometer.

Subjects were required to have pure-tone thresholds in the moderate to severe sensorineural hearing loss range (40-90 dB HL) with air-bone gaps less than 10 dB at any frequency (500 Hz to 6000 Hz) bilaterally for inclusion in this study. The slope of the hearing loss was flat or gradually sloping, defined as no greater than 20 dB HL between octave-band frequencies. Subject threshold data are shown in Figure 1 with the top section representing the right ear and the bottom section representing the left ear. Mean pure-tone thresholds for frequencies 250 Hz to 8000 Hz ranged from 48.0 to 72.5 dB HL (SDs ranging from 9.8 to 15.26 dB) for the right ear and ranged from 46.5 to 72.75 dB HL (SDs ranging from 9.12 to 13.13 dB) for the left ear. Individual hearing threshold data are presented in Appendix A.

Figure 1. Mean audiometric thresholds for hearing-impaired subjects. The right ear is shown in the top portion and the left ear is represented in the lower portion of the figure.



Word Recognition Ability. A compact disc (CD) player (JVC, Model XL-V282) was used to deliver pre-recorded speech materials (VAMC, Audiology Section, Disc 1.0) for the speech reception threshold (SRT) (CID W-1), and word recognition tests (CID W-22) in quiet at the subjects' most comfortable listening level with a minimum score of 50% for hearing-impaired subjects. Stimulus words were presented using a carrier phrase. Subjects were encouraged to guess, if the stimulus word was not understood. The word recognition task was scored as whole word (correct / incorrect) as done in standard audiometric testing. Experimental word recognition described later in this chapter was scored as individual phonemes. Under earphones, the subjects had word recognition scores ranging from 58% to 100% correct. The mean word recognition performance was 73.7% correct in the right ear (SD = 13.07%) and 78.8% correct in the left ear (SD = 14.07%). The data indicate that even at the subjects most comfortable listening levels, the subjects had some difficulty understanding speech.

Middle Ear Function. A middle ear analyzer (Grason-Stadler, Model 33) was used to perform tympanometry at the probe frequency of 226 Hz. All subjects had normal middle ear functions defined as a peaked tympanogram with static admittance within ± 100 daPa and middle ear volume between 0.3 ml and 1.5 ml.

HEARING INSTRUMENTS

Description of Hearing Instruments. The subjects were fit binaurally with Phonak PiCS Sono-Forte 332X AudioZoom hearing aids. The hearing aids were digitally

programmable and were used in an omni-directional microphone mode or with the dual microphone array for directional microphone operation. The earmold configuration (skeleton versus full-shell) varied from subject to subject depending on the degree and configuration of the loss, however, all subjects used a closed earmold with no venting.

The ALD was attached to the binaural hearing aids using audio shoes and direct audio-input (MicroLINK, ML6). In the ALD conditions, the remote microphone transmitted FM signals to the receivers which were directly connected to the hearing aids. The remote microphone transmitter was set for omni-directional or directional microphone operation.

A description of the key functional components of each of the amplification systems is shown in Table 1. In each case, the omnidirectional microphone is a single microphone and the directional microphone is an array of two microphones. It is important to note that the microphones in both hearing aids accepted inputs -- at reduced levels -- in all ALD conditions.

Hearing Instrument Verification Procedures. Electroacoustic measurements of hearing instrument function were performed using the Audioscan Model RM500 Real Ear Measurement and Hearing Aid Test System, located in a sound treated test room. Hearing instruments were electroacoustically analyzed using the Audioscan Model RM500

Table 1. Amplification devices and types of microphone technology used in this study.

Amplification Device	Designator	Hearing Aid Microphone	Remote Microphone
<u>Hearing Aids</u>	HA-O	Omni-directional	None
	HA-D	Directional plus noise suppression circuitry	None
<u>Assistive Listening Devices</u>	ALDI-O and ALDC-O	Omni-directional with a 20-dB reduction in sensitivity	Omni-directional with noise suppression circuitry
	ALDI-D and ALDC-D	Directional with a 20-dB reduction in sensitivity	Directional microphone with noise suppression circuitry

prior to fitting, to verify that the hearing instruments were performed in accordance with manufacturer specifications and ANSI S3.22 (1996).

Based on the pure-tone thresholds of audibility, the desired gain of the hearing instrument was determined using the National Acoustics Laboratories' (NAL-R) procedure (Byrne & Dillon, 1986). Real-ear insertion gain measurements were obtained unaided and with each hearing aid coupled to the probe-microphone system. The hearing aid was adjusted until the frequency response (REIR) matched within 5-dB for all frequencies 200 Hz to 3000 Hz and within 10 dB 3000 Hz to 6000 Hz to the NAL-R target response (Mueller, Hawkins, & Northern, 1992). Changes to REIR were made

based on subjective evaluation of sound quality. Verification of the hearing instrument function was determined at two presentation levels, the first was below the compression kneepoint (approximately 40 dB SPL) and the second was above the compression kneepoint (approximately 70 dB SPL). The frequency response and acoustic gain measures were digitally programmed into the hearing aid. No modifications to the gain of the hearing aid were made when the hearing aid was switched to the directional microphone or when the ALD was coupled to the hearing aid. Although, this did result in changes in the output signal levels – reduction ranged from 8.0 dB to 1.7 dB, it was the expected format that a hearing aid user would utilize. That is, given the programming constraints of this amplification device, the hearing aid user would not change the programming of the instrument when the ALD was coupled to the hearing aid.

EXPERIMENTAL MEASURES

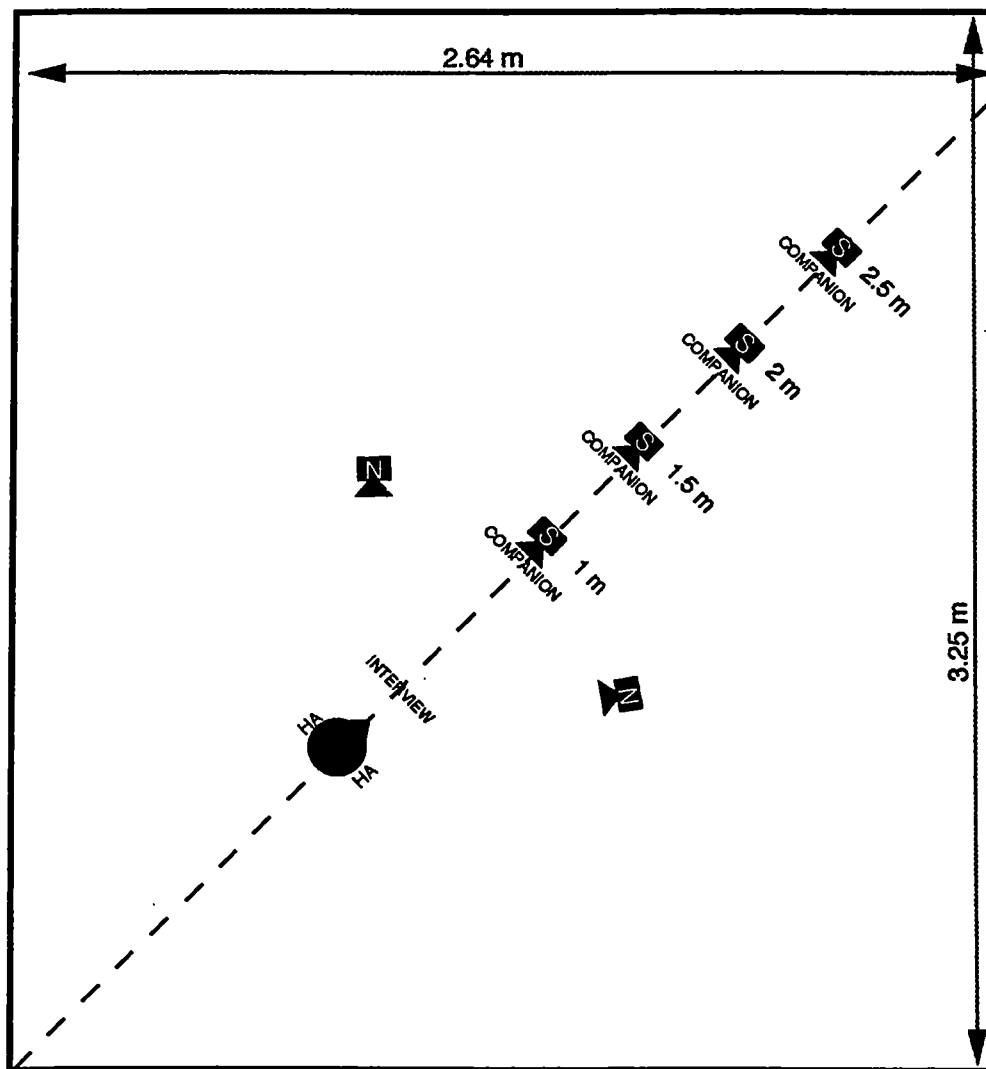
Hearing instrument performance was evaluated using measurements of signal and noise levels at the tympanic membrane at four talker-listener separation distances and by performance on word recognition in multi-talker babble tasks at two distances for two amplification systems. (1) Amplification with binaural digitally programmable behind-the-ear hearing aids (HA). (2) Amplification with the same binaural HAs coupled with an ALD using a remote microphone detecting acoustic signals and routing the same signals to each of the BTE hearing aids using a frequency-modulated radio transmission system. The ALD was subdivided into two modes of operation. (2a) The remote microphone was placed as though it was hand-held by the listener at 0.41 m (16 inches) from the center of

the listener's head (interview mode). (2b) The remote microphone was mounted on a stand as though it was hand-held by the talker, positioned at a right angle to the loudspeaker at 0.08 m (3 inches) directly out from the loudspeaker and 0.15 m (6 inches) directly below the loudspeaker sound source (companion mode). This paradigm was used to simulate the position used by a talker if handed the remote microphone without any additional instructions as to placement.

Following the initial programming of the hearing aids, subjects were asked to adjust the gain control of the hearing instrument to a level that was considered to be comfortable to them while listening to continuous discourse presented at 55 dB HL at the 1.5 meter stimulus speaker location. This level was programmed into the hearing aid and no further adjustments to the gain of the hearing aid or ALD were made during the experiment. For listening in background noise, the 'party' algorithm was used to set the gain and frequency response of the devices.

Test Conditions. A schematic drawing of the test conditions is shown in Figure 2. The listener was seated in a fixed position and the two loudspeakers used to present multi-talker babble were fixed in position at 1 m from the listener at 45 degree azimuth, one to the right and one to the left of the listener. The third loudspeaker used to present the stimulus speech was set at 1, 1.5, 2 and 2.5 m from the listener.

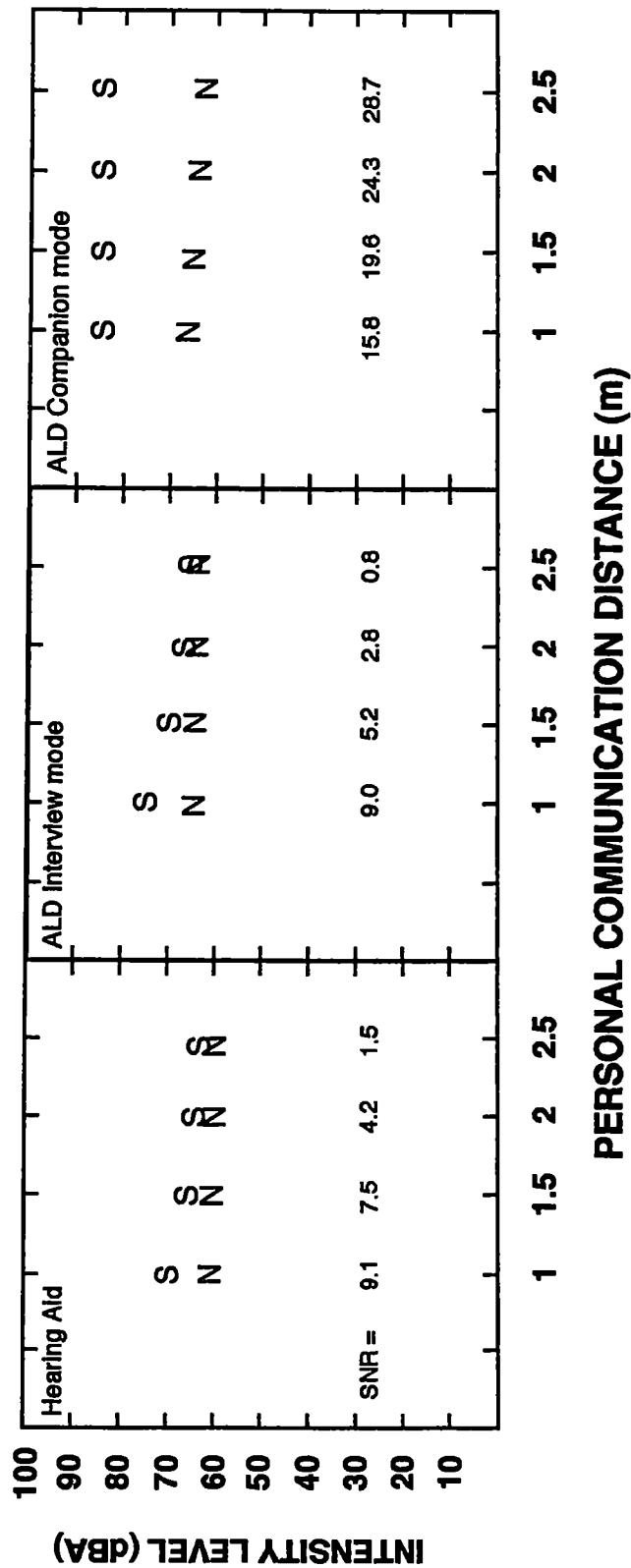
Figure 2. Schematic drawing of audiometric test suite used for experimental listening conditions including placement of subject in reference to stimulus loudspeaker and noise speakers. Amplification microphone is labeled according to type of device (HA represents hearing aids, INTERVIEW indicates listener-held remote microphone for the ALD, and COMPANION is used to identify speaker-held remote microphone for the ALD).



Signal and Noise Levels in the Environment. In the determination of signal and noise levels, speech spectrum noise was used to represent the level of the speech and background noise. The level of the speech spectrum noise was calibrated using ANSI S3.6-A.6 guideline for sound field based on findings from Stream & Dirks (1974). Signal and noise levels were specified by measuring speech noise for both the signal and the noise under the three microphone locations: (1) at the auricle for one ear, (2) at subjects' arms length (approximately 0.41 m from center of head), and (3) at 0.8 m in front of and 0.15 m below the stimulus speaker using a Larson-Davis Sound Level Meter with a 1 inch omnidirectional microphone. The measurements that are shown in Figure 3 were made on one individual with the assumption that sound levels would be similar for all subjects seated in the same position.

Signal and Noise Levels near the Tympanic Membrane. Speech and background noise were also measured near the tympanic membrane using the probe-microphone measurement system (Audioscan, Model RM500) that was used for verifying the hearing aid fittings and electroacoustical analysis of the hearing aids. Sound levels were measured at 65 frequencies, measured in 1/12th octave steps, over the frequency range of 200 to 8000 Hz using the probe-microphone system. All performance data was initially stored in the Audioscan RM500 and then downloaded to a personal computer using the Audioscan RM500 XDATA32 data extraction program. This program converted all probe-microphone data stored on the Audioscan into ASCII text files. The ASCII text

Figure 3. Signal and noise level measurements made in the environment at each of the four signal loudspeaker distances using an omni-directional sound level meter.



files were then saved and stored as individual subject files in Microsoft Excel. Subsequent data analysis was completed using the Microsoft Excel (Version 5.0) program and Statistica software.

Probe-microphone measurements of speech and background noise levels were obtained at the tympanic membrane for one ear of each subject for each stimulus speaker location (1, 1.5, 2, and 2.5 m) and for the each of the following experimental conditions: (1) hearing aid using an omni-directional microphone (HAO); (2) hearing aid using directional microphone (HAD); (3) assistive listening device with the remote microphone 0.41 m from the listener (Interview mode) using omni-directional microphone (ALDI-O); (4) ALD - Interview mode using directional microphone (ALDI-D); (5) assistive listening device with the remote microphone 0.15 m below the talker (Companion mode) using an omni-directional microphone (ALDC-O); and (6) ALD - Companion mode using a directional microphone (ALDC-D). The average sound level measurements at each of the loudspeaker distances and microphone locations are shown in Appendix C.

The experimental listening conditions represented in Figure 1 illustrate the talker, listener, and noise source locations, however, the signal and noise levels reaching the listener varied significantly with each of the three amplification systems. In the HA condition, the microphone of the amplification device was located at the listener's ear. The signal picked up by the HA microphone was dependent on the location of the source. As the talker loudspeaker was moved from 1m to 2.5 m, the distance between the hearing

aid microphone and the signal loudspeaker increased. This increase in the distance resulted in a decrease in the intensity of the signal reaching the listener's ear. While the talker loudspeaker changes position, the noise loudspeaker remained in the same position, thereby keeping the intensity of the noise constant. Changes in signal-to-noise ratio were a result of changes in the signal level. This is also true for HAD. When using the directional microphone, noise from behind the listener was suppressed and the signal coming from in front of the listener was amplified. The noise suppression characteristics illustrated by polar sensitivity patterns for omni-directional and directional microphones are shown in Appendix D.

In the ALDI condition, the microphone was hand-held by the listener. For this study, the listener was asked to hold the microphone comfortably in front of them and pointed in the direction of the talker loudspeaker, thus the microphone of the ALDI was approximately 0.41 m closer to the talker loudspeaker than was the HA microphone. As in the HA condition, as the talker loudspeaker was moved from 1 m to 2.5 m, the sound level arriving at the ear decreased. The noise loudspeakers remained in the same position, so that the level of the noise remained constant. SNR decreased with an increase in the talker to listener distance. The hand-held remote microphone, when switched to directional mode, accepts more sound from the central focal region and suppressed sound from the side and back. Thus, the focus was created in front of the microphone. Unlike HA-D, which suppressed sound from behind the listener only, the ALDI-D suppressed

sound from the side as well. The polar sensitivity pattern for the remote microphone is shown in Appendix D.

In the ALDC condition, the remote microphone was placed 0.8 m in front of and 0.15 m below the talker loudspeaker. As the talker loudspeaker was moved from 1 m to 2.5 m, the signal level remained relatively constant, because the remote microphone moved with the talker loudspeaker as the distance was increased. Additionally, in the ALDC, as the talker loudspeaker was being moved away from the listener, the remote microphone was also being moved away from the noise, therefore changes in SNR were a result of changes in noise level rather than signal level as the distance was increased.

Word Recognition in Multi-talker Babble. Word recognition performance was obtained using the Northwestern University Auditory Test No. 6 (List 1 - 4, Randomizations A - D) presented in a six-person multi-talker babble noise. A 1000 Hz calibration tone was used to determine signal level at the ear prior to initiating test procedures. As shown in Figure 2, the speech stimulus was presented from a sound field speaker located at 0° azimuth whereas the multi-talker babble noise was presented from two sound field speakers 1 meter from the listener at 45° azimuth, one to the right and one to the left of the listener. The speech stimulus speaker was placed at 1 and 2 meters for experimental conditions. Word recognition was measured at two representative talker-listener distances: 1 and 2 m. The two stimulus speaker locations were chosen to best represent normal conversational distances. Word recognition was scored using the

individual phonemes of each CVC word -- allowing for 150-item analysis of each 50-word list to increase the stability of the measurement (Gelfand, 1998). Word recognition performance was evaluated by three examiners. Two of the examiners were graduate students in the Master's program in Audiology at Lamar University. The third examiner was the principal investigator of this study. Each word list was independently scored by the examiners and averaged.

EXPERIMENTAL PROTOCOL

Subjects participated in two 90-minute test sessions. In test session one, the qualification procedures were conducted and the informed consent was obtained. Unaided experimental measures were obtained for the word recognition in multi-talker babble task at talker-listener distances of 1 and 2 m. Subjects were then fit with binaural BTE hearing aids. At this time, probe-microphone measurements were obtained to determine proper fit of the amplification system. Measurements were made for the signal and noise levels at the tympanic membrane for each of the four stimulus loudspeaker locations and each microphone condition.

In test session two, the hearing aids were re-programmed to the listeners' setting initially acquired in test session one. Aided word recognition in multi-talker babble was obtained for each of the three amplification devices and the two microphone conditions at talker-listener distances of 1 m and 2 m in a randomized presentation order. Thus, results were obtained for twelve 50-word lists for each subject in session two.

CHAPTER FOUR

RESULTS

BEHAVIORAL MEASUREMENT OF HEARING INSTRUMENT FUNCTION

Performance for each device was determined using word recognition scores at two distances. The mean performance scores are listed in Table 2 and shown in Figure 4. At both distances evaluated, word recognition performance was the poorest with the HA, improving with the ALDI and best when using the ALDC device. The use of directional microphones improved word recognition for each device but the improvement was not sufficient to change the rank ordering of devices. Specifically, performance with HA-D was still poorer than with ALDI-O and performance with ALDI-D was still poorer than ALDC-O.

A three factor MANOVA was conducted on the performance data. The factors were amplification device (3 levels), microphone directionality (2 levels) and communication distance (2 levels). A significant main effect was found for amplification device with the HA having the poorest mean performance of 65.3%, the ALDI improving performance to 77.4%, and the ALDC having the best word recognition performance with 82.8% for scores averaged across both distances and microphone directionality.

For each device, performance improved significantly with the use of directional over performance with omni-directional microphone. Performance improved by 9.4% when using HA-D over performance with HA-O (60.6% to 70.0%); improved by 4.6% for ALDI-D over performance with ALDI-O (75.1% to 79.7%), and remained essentially

Table 2. Multivariate Analysis of Variance (MANOVA) for Word Recognition Performance.

Source	df	F	p
Communication Distance (CD)	1, 19	28.23	.000 *
Device (D)	2, 38	65.63	.000 *
Microphone (M)	1, 19	135.31	.000 *
CD x D	2, 38	13.90	.000 *
CD x M	1, 19	0.55	.464
D x M	2, 38	47.73	.000 *
CD x D x M	2, 38	0.24	.787

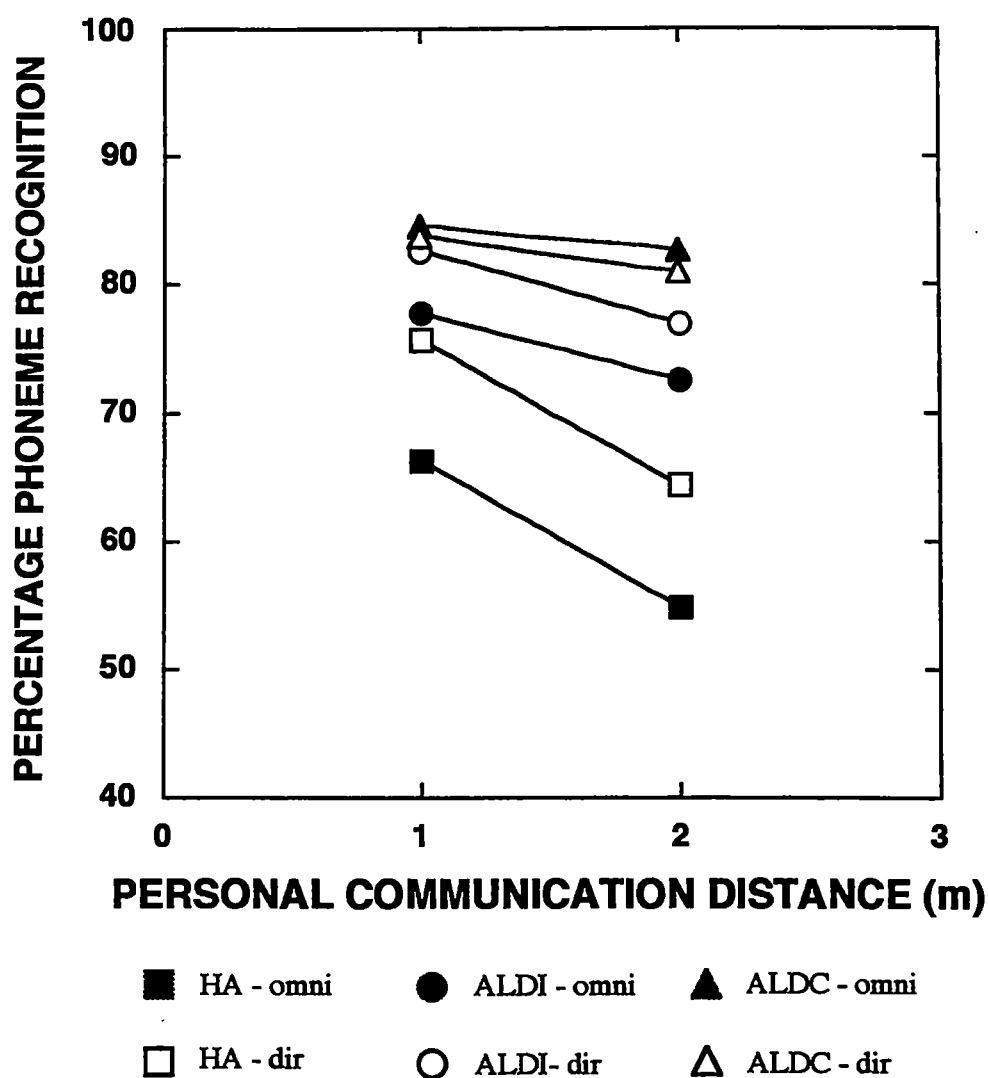
* significant

constant for ALDC with scores of 83.5% for ALDC-O and 82.2% for ALDC-D.

With each device, performance decreased significantly as communication distance was increased from 1 to 2 m. Performance decreases were greatest for HA and ALDI because talker-listener distance increased with communication distance. Performance decreases were greatest with HA --11.4% (71.0% to 59.6%). With ALDI, performance was reduced by 5.5% (80.1% to 74.6%). With ALDC, talker-listener distance remained constant with increased communication distance, and the performance decreased by a very small amount -- 2.3% (84.0% to 81.7%)

Figure 4 illustrates the ordering of performance by communication distance. At 1 m, the poorest performance was obtained when using HA-O. Performance improved with the use of HA-D and with ALDI-O. The best performance was obtained using ALDI-D,

Figure 4. Mean word recognition performance for each amplification device (Hearing aid - $\square$); (ALD - Interview - $\circ$); (ALD - Companion - $\triangle$) by communication distance. Filled symbols represent each device in omnidirectional mode and the unfilled symbols represent each amplification system in the directional mode.



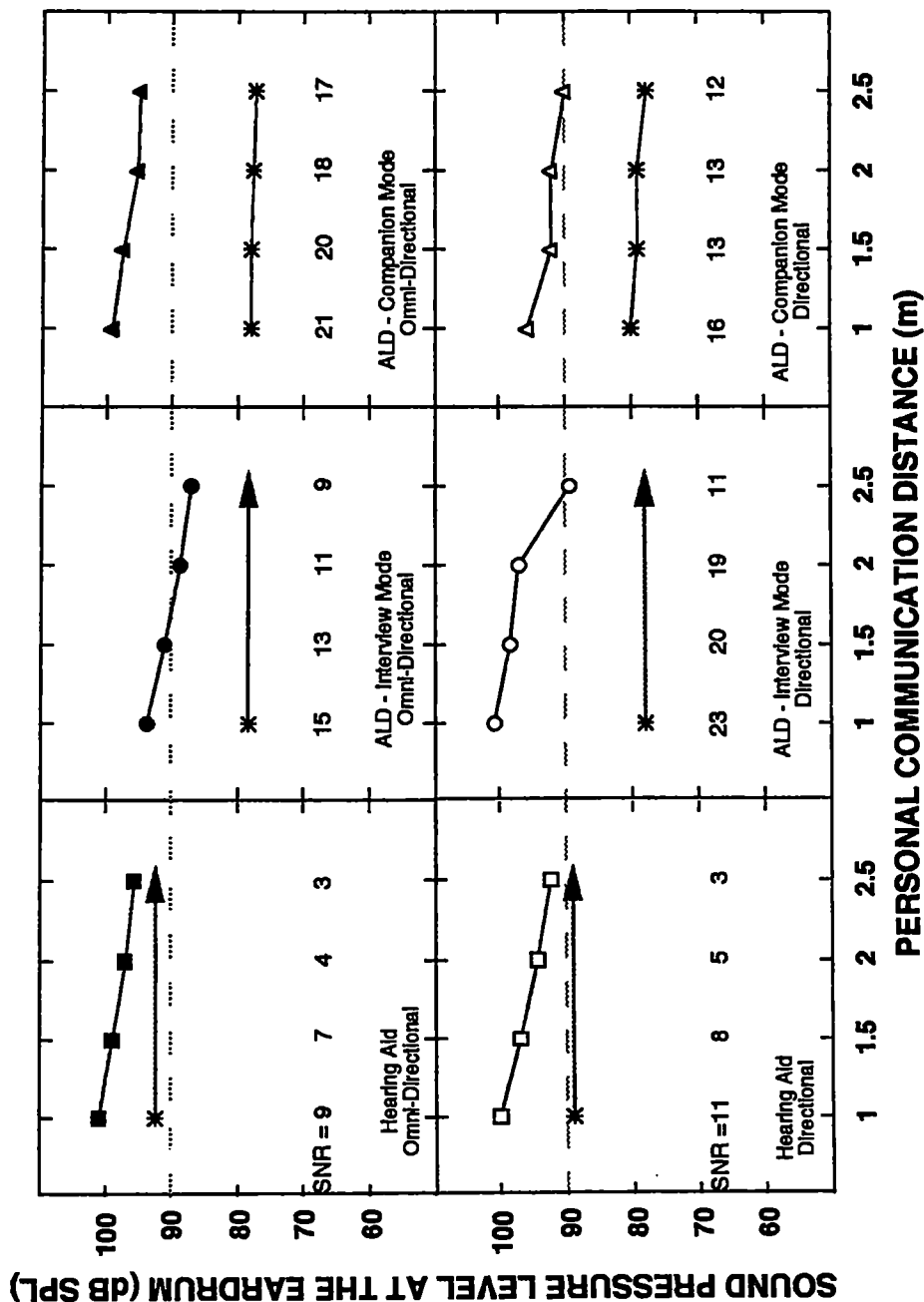
ALDC-O or ALDC-D. Although performance improved for each amplification device with the use of directional microphones, performance did not improve to the level of the next amplification device. At 2 m, word recognition performance decreases for all three amplification devices. The degree of separation is greater between the types of devices, however, the ordering of the devices remains constant. The significant interaction between amplification device and communication distance is clearly illustrated in Figure 4. At the extremes, HA performance decreased substantially with increased distance while ALDC performance decreased only minimally. The interaction between amplification device and microphone directionality was significant because HA and ALDI performance improved substantially with directional microphones while ALDC performance did not.

ACOUSTICAL MEASUREMENTS OF HEARING INSTRUMENT FUNCTION

Acoustical measures were made at four distances (1, 1.5, 2, 2.5 m) to aid in describing the behavioral measurements. Statistical comparisons were made among measurements of signal and noise levels at 1 and 2 m, to determine the relations to word recognition performance measures.

Signal-to-Noise Ratio. Overall SPL values for signal and noise levels were computed based on the sixty-five frequency sampling (200-8000 Hz) data. The mean signal levels, noise levels, and resulting signal-to-noise ratio (SNR) for each device are shown in Figure 5. The upper figure illustrates the signal and noise levels for the omnidirectional mode and the lower figure represents the directional mode. The HA and ALDI noise levels remain constant as talker to listener distance increases, therefore changes in SNR are a result of changes in signal level rather than noise level. The noise levels of the

Figure 5. Sound pressure level at tympanic membrane for each amplification device in omni-directional and directional microphone placements. The star symbols represent the multi-talker babble intensity levels. The dashed line is an arbitrary reference line to aid in viewing differences in signal and noise levels. Resulting SNR values (rounded to the nearest whole number) are given for each device and distance.



ALDC were expected to decrease as the microphone was moved away from the noise, whereas the signal level was expected to remain constant.

As communication distanced increased, the signal level decreased for both HA-O (101.25 to 95.86 dB SPL) and HA-D (100.12 to 93.12 dB SPL). The noise level remained constant at all communication distances, however, the noise level of HA-D (88.99 dB SPL) was less than that of HA-O (92.5 dB SPL), thereby resulting in a better SNR at all distances for HA-D than for HA-O.

When the ALD was coupled to the HA, the input was decreased by 20-dB. When the ALD remote microphone was hand-held by the listener, the distance between the microphone and the stimulus loudspeaker was reduced from that of the HA. The resulting effect was a decrease in the overall signal level from that of the HA, however, the noise level was also reduced by 20-dB. For the ALDI-O, the reduction in signal level coupled with the greater reduction in noise resulted in an improved SNR over that of the HA. When the ALDI was switched to directional mode, the signal level was increased, however, the noise level remained similar to that of the ALDI-O, thereby, resulting in a greater SNR with ALDI-D than with ALDI-O.

The placement of the remote microphone next to the speaker in the ALDC condition nullified the 20-dB loss caused by the HA-ALD coupling. Therefore the signal level in the ALDC-O condition was louder than in the HA and the ALDI conditions. When the ALDC was switched to the directional mode (ALDC-D), the signal level decreased and the noise level increased from that in the ALDC-O. This may have been

due to the angle of the microphone, located 6 inches directly underneath the speaker, instead of being angled towards the speaker diaphragm, and by picking up more noise from the environment at the close distances and less as the distance increases. The SNR was less favorable for the ALDC-D than for the ALDC-O or the ALDI-D and similar to that of the ALDI-O.

One three-factor MANOVA was conducted on the SNR data and is shown in Table 3. The factors were amplification device (3 levels), microphone directionality (2 levels) and communication distance (2 levels). A significant main effect was found for amplification device with SNR being the least favorable for HA, SNR improved with ALDI and was the most favorable for ALDC.

A significant main effect was found for microphone directionality, indicating that

Table 3. Multivariate Analysis of Variance for Signal-to-Noise Ratio.

Source	df	F	p
Communication Distance (CD)	1, 19	268.75	.000 *
Device (D)	2, 38	35.05	.000 *
Microphone (M)	1, 19	12.41	.002 *
CD x D	2, 38	6.18	.005 *
CD x M	1, 19	1.25	.277
D x M	2, 38	25.00	.000 *
CD x D x M	2, 38	2.13	.133

*significant

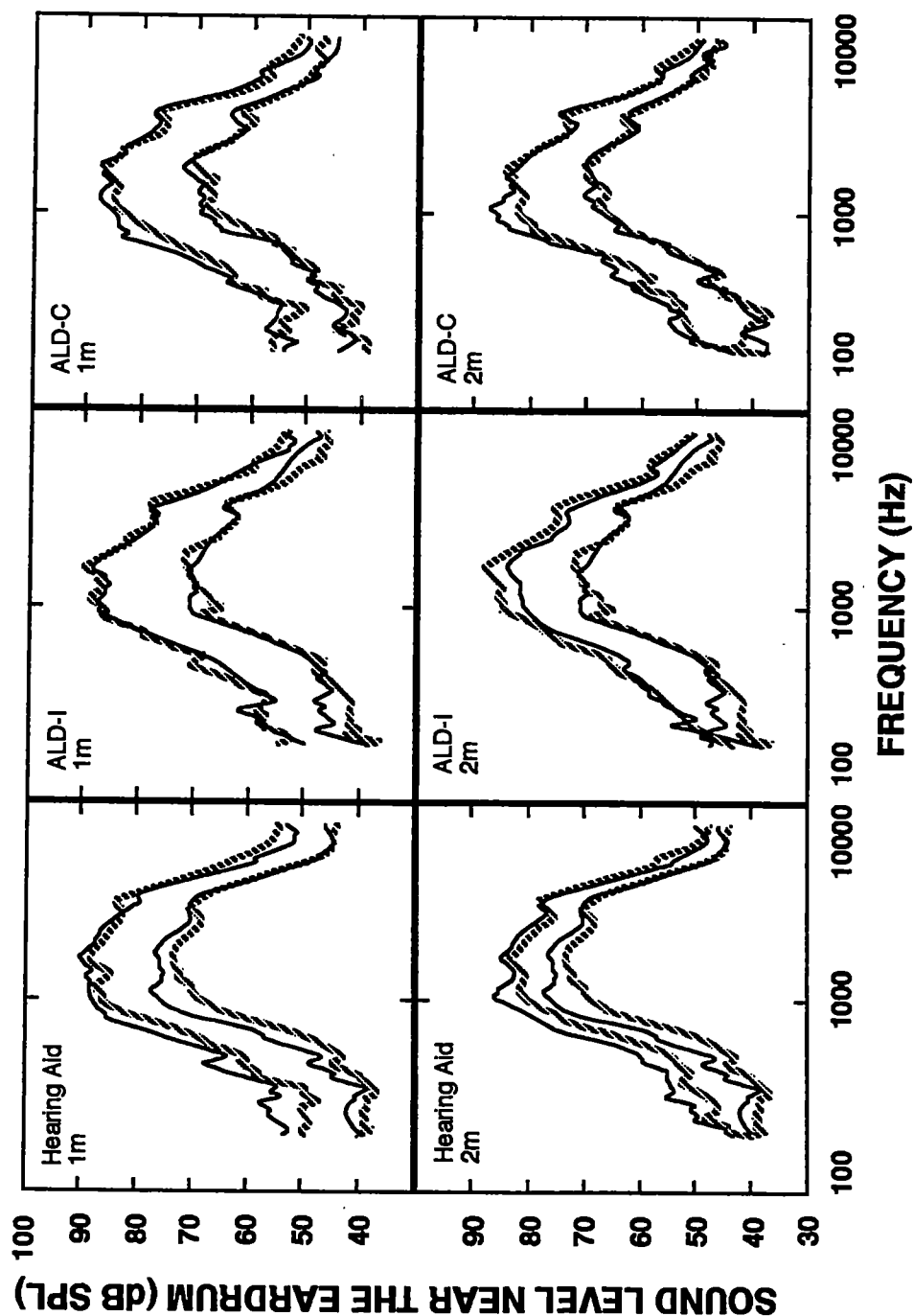
the use of directional microphones increased the SNR for the amplification devices over use of omni-directional microphones. Figure 6 illustrates the frequency response of the signal and noise levels for each amplification device in omni-directional and directional modes at 1 and 2 m. A reduction in both signal and noise levels were seen for HA-D and were primarily low frequency, from 200 – 1500 Hz, however, the improvement in SNR that occurred in the frequency range from 200 to 1500 Hz had only a slight effect on the overall SNR computed from 200-8000 Hz. Thus, the effect of reducing the level of the low frequency region, while may be minimal in SNR, may have a correlative effect to significantly improve word recognition performance (Valente, 1998).

The frequency response for the ALDI at 1 m was similar for both ALDI-O and ALDI-D, with the ALDI-D having a greater signal level in the mid-frequencies and a reduced noise level in the mid-frequencies. At 2 m, the ALDI-D was a signal enhancer for low, mid and high frequencies over ALDI-O, which resulted in a better SNR.

The ALDC frequency response was broader for ALDC-O than for ALDC-D and ALDC-O was greater across the all frequency regions than ALDC-D. This resulted in a greater SNR for ALDC-O than for ALDC-D. The difference in frequency response between the ALDC-O and ALDC-D may have been a result of poor microphone positioning for the ALDC-D due to the angle of the microphone, located 6 inches directly underneath the speaker, instead of being angled towards the speaker diaphragm.

A significant main effect was also obtained for communication distance indicating SNR decreased as the distance was increased for all three devices. SNR as a function of

Figure 6. Mean frequency response for signal and noise measurements made near the tympanic membrane for each amplification device at 1 m and 2 m. The solid black line represents measurements made in the omnidirectional conditions; the dashed black line represents measurements made in the directional conditions. The signal measurements are the upper two curves of each graph and the noise measurements are the bottom two curves in each graph.



communication distance is shown in Figure 7. The rank ordering of each amplification device by SNR was similar, but not identical to the order established in the word recognition performance measures. Based on SNR measures, the HA-O had the lowest SNR at each distance. There was an increase of 2.4 dB at 1 m in SNR with the use of HA-D over HA-O. SNR improved by 6.6 dB at 1 m when using the ALDI-O over the HA-O and by 6.9 dB at 1 m when using the ALDC-D over HA-O. A large improvement in SNR was seen when using ALDC-O and ALDI-D. The SNR improved by 12.5 dB when using the ALDC-O and by 13.9 dB with ALDI-D over that obtained with HA-O at 1 m. The SNR values decrease as distance increases. The ordering effect of each amplification device remains constant until after 2 m. At 2.5 m, the SNR of the ALDI-D decreases to the level of the ALDC-D, but remains slightly better than the ALDI-O.

WORD RECOGNITION AS A FUNCTION OF SIGNAL-TO-NOISE RATIO.

Since the rank ordering of SNR and word recognition were not identical, a graphical comparison between the two measures was made. A plot of word recognition performance by SNR is shown in Figure 8. All three amplification devices were plotted using both omni-directional and directional word recognition scores at 1 and 2 m. As the SNR became more favorable, large improvements in word recognition performance occurred. Above 15 dB SNR, the word recognition scores plateaued between 78% and 84%. Regardless of the amplification system used, the listeners in this study did not obtain word recognition scores greater than 85% even with a 23 dB SNR. This is most likely due to the degree of the listeners' hearing loss. For this experimental group, the 15-dB SNR may be described as a critical SNR (CSNR).

Figure 7. Represents signal-to-noise ratio for each amplification device (Hearing aid - $\square$); (ALD - Interview - $\circ$); (ALD - Companion - $\triangle$) by communication distance. Filled symbols represent each device in omni-directional mode and the unfilled symbols represent each amplification system in the directional mode.

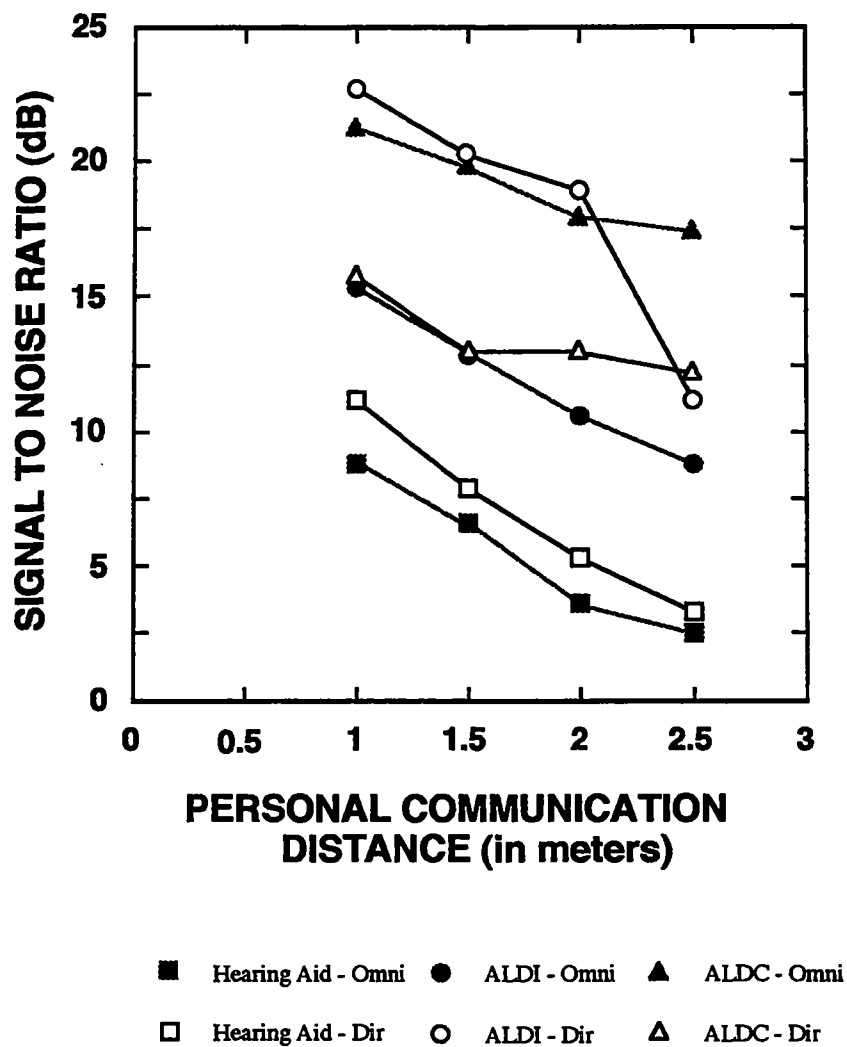
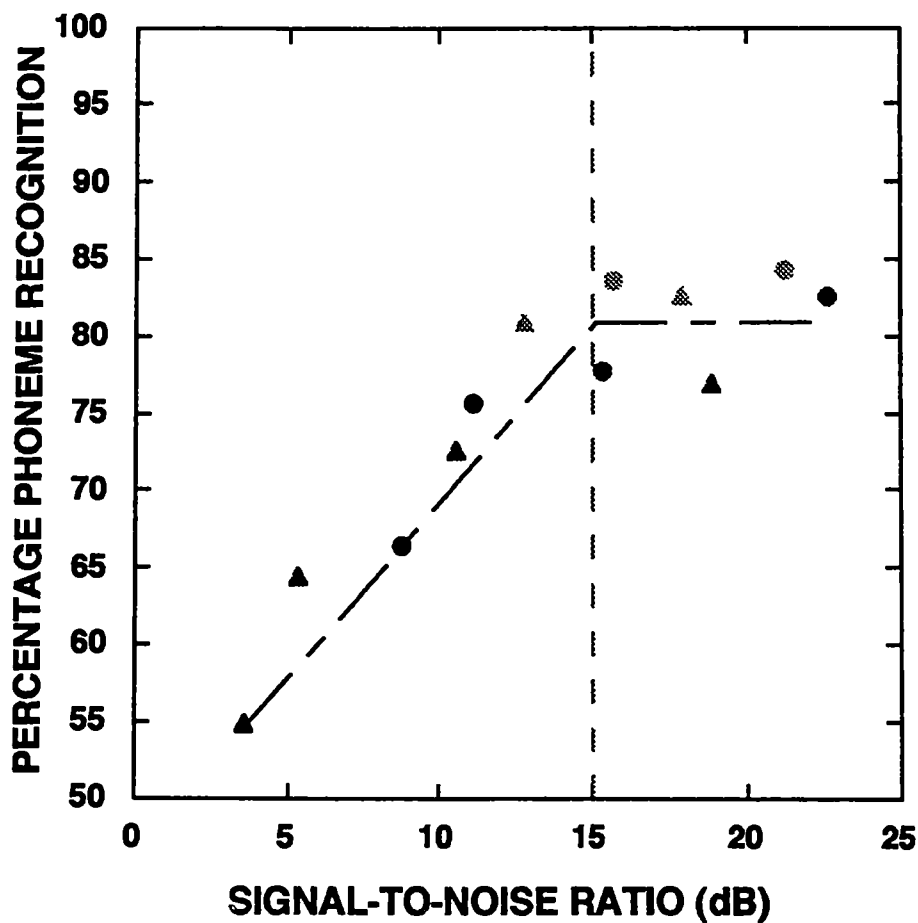


Figure 8. Mean word recognition performance as a function of signal to noise ratio. Each amplification device is plotted based on distance with the same symbol representing both omni-directional and directional listening modalities.



- | | | | |
|---|---------------------|---|----------------------|
| ● | Hearing Aid 1 meter | ▲ | Hearing Aid 2 meters |
| ● | ALD-I 1 meter | ▲ | ALD-I 2 meters |
| ◆ | ALD-C 1 meter | ▲ | ALD-C 2 meters |

CHAPTER FIVE

DISCUSSION

CRITICAL SNR

For the listeners in this study, the maximum mean word understanding ability was 84% in the presence of background noise. As shown in Figure 8 of the previous chapter, word recognition changes as a function of SNR. As the SNR became more favorable, word recognition improved systematically, however, increasing the SNR above 15-dB resulted in minimal improvements in word recognition. These findings are in agreement with previous research comparing hearing-impaired listeners and normal hearing listeners on measures of speech discrimination. Norwood-Chapman et al (1997) found, for listeners with a mild to moderate high-frequency sensorineural hearing loss, that the most dramatic improvement in word understanding ability occurred between 5-dB and 15-dB SNR. At low SNRs (0 - 5 dB), word recognition performance was minimal, but as the SNR was incrementally made more positive, the word recognition improved greatly, yet the improvements did not continue after 15 dB SNR. Above 15 dB SNR a plateau effect occurred, indicating a stabilization in performance ability. Although the listeners in the current study had more hearing loss than the listeners in the Norwood-Chapman study, the effect of SNR on word recognition remained the same. Improving the SNR resulted in improvement of word recognition ability, however, performance did not continue to improve upon reaching a critical SNR value for the listener. Killion and Niquette (2000), further emphasized the differences in SNR between hearing-impaired and normal hearing

listeners. These researchers reported that an additional 10-dB SNR was required for hearing-impaired listeners to perform at an equal level as the normal hearing listeners. At poor SNRs, word recognition performance can improve substantially with small improvements in SNR. Based on the current study, two acoustic changes, difference in overall SNR and low frequency signal and noise suppression, may be responsible for the dramatic improvements in word recognition performance.

In the current study, as the SNR was increased from 5 dB to 15 dB, the mean word understanding ability increased by 25%. While the mean overall improvement was 25%, individual differences occurred. One listener had 33% word recognition at 2 meters using HA-O, but switching to HA-D improved the word recognition to 52%. Switching to ALDI-D improved this listener's word recognition to 85% and when the ALDC-O was used, the word recognition improved to 92%. This allowed the listener to improve from 33% understanding to 92% understanding at a normal conversational distance. A 59% improvement in understanding ability can dramatically improved that listeners' auditory perception. Secondly, another listener improved from 18% with HA-O to 53% using ALDI-D and finally to 65% with ALDC-O. Although the final understanding ability could be considered minimal, a 47% improvement to that listener may have been significant.

The amount of improvement in word recognition is highly dependent on the listener's ability to understand in background noise. Determining each listeners critical SNR and their maximum level of performance becomes a pivotal counseling tool.

Additional assessment of the critical SNR can assist the audiologist in making appropriate recommendations for amplification. Using the example provided above, illustrating to a listener that even with HA-D, they are likely to only get 50% of the words, but with the ALD they could improve to 85-90%, could impact on the listener's decision of which amplification system to purchase. In addition, it can be used to establish realistic expectations by the listener. If a listener knows that even with a 20 dB SNR, s/he will only understand 80% of speech, the psychosocial factors related to hearing loss may be reduced. While one listener may be satisfied with good, but not optimal understanding, another listener may not. However, knowing ahead of time that understanding ability will be compromised in the presence of background noise can lead to less product dissatisfaction from the listener.

AMPLIFICATION EFFECTS

Omni-directional hearing aids have been the standard fitting for amplification. While HA-O provides the listener with control of the device and could be considered convenient, the rank ordering established in this study based on word recognition performance and SNR indicated that the poorest performance occurred when using HA-O. Switching to HA-D improved both speech understanding and SNR. In 100% of the cases in the present study, word recognition performance was better with HA-D than in HA-O, but even when using HA-D, the critical SNR was not reached. Therefore, performance did not reach the maximum level of understanding.

Patient satisfaction research has indicated that hearing aid users prefer the directional microphone hearing aids in noisy situations and the omni-directional microphones in quiet environments (Kochkin, 1993). Although directional microphones have been available in hearing aids for over 20 years, most users have been fitted with omni-directional microphones because they are preferred in the absence of noise and there was not a switching mechanism to allow listeners to change from one mode to the other. More recently, hearing aids have become available that have a switch that allows the user to change the microphone from omni-directional to directional depending on their listening situation. The microphone switch (omni-directional to directional) provides the user greater control for hearing in a variety of listening situations.

When using ALDI-O, improvements in word recognition performance and SNR occurred. When SNR increased about the critical 12-dB, performance neared maximum potential. Switching to ALDI-D improved performance for 38/40 or 95% of the listeners. Based on the SNR findings at 2.5 m, the advantages of ALDI-D are reduced, indicating there is a practical limit to the communication distance in which this device can be used. In order to receive near maximum benefit, the listener must be willing to hand-hold the remote microphone and position it in the direction of the speaker.

The greatest improvements in word recognition and SNR were obtained when using ALDC. Both word recognition and SNR were optimal for the listeners in this study. With ALDC-O, 98% (39/40) performed better than ALDI-O and 78% (31/40) performed

better than they did when using the ALDI-D. When using ALDC, the listener must be willing to pass the remote microphone to the speaker, thereby surrendering control of their listening needs. Although the use of directional microphones enhanced word recognition performance and improved SNR for HA and ALDI, ALDC-D did not enhance SNR nor improve word recognition performance as much as ALDC-O. This may be attributed to the microphone placement. It is evident from this study that although the overall SNR was not substantially effected, the frequency response and word recognition were affected. Further investigation needs to determine if a critical position for the directional microphone exists. If the angle and position of the directional microphone are critical, a significant problem may arise for the listener due to lack of control of the remote microphone. The talker may inadvertently alter the position of the microphone with little regard for the effect on the listener.

The ALDC did not decrease the intensity level of the noise as much as expected as the distance was increased. As the loudspeaker was moved out of the noise, it was expected that the noise level would drop significantly because the microphone was also being moved away from the noise. Instead, the noise level remained essentially constant. There are two possible explanations for this occurrence. First, when the ALD is coupled to the hearing aid, it is a combination FM+HA system, however, the hearing aid microphone input was decreased by 20 dB. The hearing aid microphone was closer to the noise loudspeakers than the ALDC microphone possibly resulting in a difference greater than 20 dB. If so, the hearing aid microphone would begin to input the signal in addition

to the signal coming from the ALD microphone, thereby increasing the overall noise level and negating any enhancement from the ALDC alone. A second possible explanation for the lack of change in the noise level may be due to using a sound treated room. Refer to Figure 1 (p. 44) for room schematic. As the loudspeakers were moved away from the listener, it was in actuality being moved closer into a corner of the booth where more sound absorption may have taken place. The sound booth used in this study was quite large for most clinical settings and did offer the capability of achieving larger distances (up to 2.5 meters). The next step in determining if this was truly an effect of the hearing aid microphone or the booth would be to utilize the system in an open area. While this has been done for classrooms, auditoriums and theaters, the distances tested were always greater than 3 meters. Evaluating the ALDC at personal communication distances in an open room could provide additional insight on signal and noise levels.

Prior to this study, the ALDC had not been evaluated at personal communication distances. It was understood throughout this study that ALDC was the best tool for improving SNR and word recognition at distances greater than 3 meters. Based on the results of this study, it is now known that the ALDC works well even at closer distances. A disadvantage of using ALDC remains that the microphone must be passed to the speaker and then only one person can speak at a time. While this may be the least desirable for self-conscious hearing-impaired listeners, there are many situations in which this would still be the best option. While the ALDC may be the best option for distances greater than 3 meters, the ALDI has shown that it too can improve word recognition and

SNR significantly.

FEASIBILITY

While the ALD system is flexible, it is also costly to the patient. Asking the patient to purchase not only programable hearing aids but also an ALD may cause the listener to find a less expensive alternative. Due to financial constraints of the listener, an extended trial period with these devices may not be possible. The next alternative may be to test the system in the clinic and determine how effective the system is for each listener based on their individual listening needs. Administering a hearing handicap inventory or hearing needs assessment could be beneficial in determining which patient's to test with the system. Listeners with considerable difficulty in reverberant conditions or in the presence of background noise may benefit from using an ALD over hearing aids with directional microphones.

The ALD has many practical applications for the socially active listener. For the active person, the ALD can be used for one-on-one meetings, business lunches or dinners, or in the conference room. When in an one-on-one situation, the listener may use either his/her hearing aids depending on the amount of background noise or use the ALD in the interview mode in either the omni-directional or directional mode. When noise levels increase, such as in restaurants, the listener may chose to use the ALD-I in the directional mode. This would allow the listener the flexibility to point the remote microphone towards each speaker, or the listener may chose to use the ALDC in omni-directional

mode. For the listener that doesn't want to appear hearing-impaired, the ability to place the microphone on the table and not draw attention to himself / herself as they point it towards each person may seem less noticeable. Others may chose to sit at the head of a table and place the remote microphone next to them and point the microphone in the general direction of the others at the table. When dining, it would be difficult to hold the microphone in one hand and eat with the other. Therefore, being able to set the remote microphone either in the center of the table or beside them and use the directional mode would allow the listener the ability to perceive the conversation around them with much less strain. In a conference room, the listener may use any of the three systems, depending on their needs and the noise levels in the conference room. In a small intimate meeting with four or five colleagues, the listener could use either their hearing aids or the ALDI or ALDC. Larger meetings, with multiple talkers, may prohibit the use of ALDC, making ALDI easily accessible and practical for the listener.

CONSIDERATIONS FOR FUTURE RESEARCH

Future research needs to evaluate the ALDI at distances greater than 2 meters for word recognition performance and greater than 2.5 meters for SNR. It would also be interesting to note if differences in performance could be obtained from an acoustically treated environment versus a more natural environment exist. Additionally, the remote microphone has the capability of a more focused dual microphone system. The performance characteristics of this system, for both SNR and word recognition are unknown. Further research needs to be undertaken to determine the benefit, if any, of the

super-directional microphone. In addition, subjective performance ratings should be measured for the ALD in all modalities, (companion and interview - with all three microphone directions).

Research should also be conducted for the ALD used in the companion mode to determine the best angle of microphone placement. In this study, there were directional effects with the ALDC that could not be explained. Additional research on the angle of the directional microphones could be used to determine if even a greater benefit than the one illustrated in this study can be achieved.

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APPENDICES

APPENDIX A
THRESHOLDS OF HEARING ACUTY

Pure-Tone Air Conduction Thresholds - Right Ear (dB HL)										
S#	250 Hz	500 Hz	1000 Hz	1500 Hz	2000 Hz	3000 Hz	4000 Hz	6000 Hz	8000 Hz	
S# 1	60	60	55	55	50	60	60	55	45	
S# 2	40	40	35	50	55	55	60	70	75	
S# 3	55	60	60	60	55	60	65	80	75	
S# 4	55	55	50	60	60	60	65	60	65	
S# 5	55	50	55	50	50	55	60	65	60	
S# 6	40	40	45	45	40	45	40	60	70	
S# 7	50	50	55	50	50	70	70	80	90	
S# 8	40	40	50	55	60	70	75	85	90	
S# 9	35	35	35	35	45	60	65	45	55	
S# 10	80	85	85	80	70	80	80	90	95	
S# 11	35	40	70	65	65	65	60	75	70	
S# 12	40	45	45	45	40	40	50	45	65	
S# 13	55	55	50	60	60	60	65	60	65	
S# 14	60	60	55	55	50	60	60	55	45	
S# 15	40	50	65	70	80	80	85	90	95	
S# 16	55	60	60	60	55	60	65	75	70	
S# 17	40	45	60	60	60	60	65	85	90	
S# 18	35	30	55	65	65	70	85	85	90	
S# 19	50	50	55	60	60	70	85	75	70	
S# 20	40	40	65	65	65	65	70	75	70	
Mean	48.00	49.50	55.25	57.25	56.75	62.25	66.50	70.50	72.50	
SD	11.52	12.24	11.41	9.93	9.90	9.80	11.48	14.13	15.26	
Min	35	30	35	35	40	40	40	45	45	
Max	80	85	85	80	80	80	85	90	95	
Range	45	55	50	45	40	40	45	45	50	
N	20	20	20	20	20	20	20	20	20	

Pure-Tone Air Conduction Thresholds - Left Ear (dB HL)											
S#	250 Hz	500 Hz	1000 Hz	1500 Hz	2000 Hz	3000 Hz	4000 Hz	6000 Hz	8000 Hz		
S# 1	60	65	65	70	70	70	70	65	60		
S# 2	40	40	35	45	60	60	65	65	75		
S# 3	55	55	55	55	50	65	75	70	70		
S# 4	55	50	50	60	60	65	75	60	75		
S# 5	45	45	45	45	40	45	50	65	60		
S# 6	45	45	40	40	40	45	40	60	70		
S# 7	45	45	40	45	45	60	65	65	60		
S# 8	35	40	55	55	55	70	75	85	90		
S# 9	35	35	35	40	50	75	70	60	65		
S# 10	60	60	65	65	65	60	70	85	90		
S# 11	45	60	65	65	65	60	65	70	90		
S# 12	40	50	60	60	55	55	55	50	55		
S# 13	55	50	50	60	60	65	75	60	75		
S# 14	60	65	65	70	70	70	70	65	60		
S# 15	40	40	50	75	80	85	85	90	95		
S# 16	55	55	55	55	60	65	70	70	70		
S# 17	40	40	55	55	50	65	75	90	90		
S# 18	25	30	30	55	45	65	60	55	55		
S# 19	50	50	55	75	80	70	80	75	65		
S# 20	45	45	65	65	65	65	60	70	85		
Mean	46.50	48.25	51.75	57.75	58.25	64.00	67.50	68.75	72.75		
SD	9.61	9.63	11.15	10.82	11.73	9.12	10.58	11.22	13.13		
Min	25	30	30	40	40	45	40	50	55		
Max	60	65	65	75	80	85	85	90	95		
Range	35	35	35	35	40	40	45	40	40		
N	20	20	20	20	20	20	20	20	20		

APPENDIX B
SIGNAL AND NOISE LEVELS
MEASURED IN THE ENVIRONMENT

Average sound level measurement (dB SPL) for speech and noise by amplification device and personal communication distance measured in the environment with omni-directional microphone.

Personal Communication Distance 1 meter

	HA	ALD-Interview	ALD-Companion
Speech	76.3	79.1	89.3
Noise	67.2	70.1	73.5
SNR	9.1	9.0	15.8

Personal Communication Distance - 1.5 meters

	HA	ALD-Interview	ALD-Companion
Speech	74.7	75.3	92.0
Noise	67.2	70.1	72.4
SNR	7.5	5.2	19.6

Personal Communication Distance - 2 meters

	HA	ALD-Interview	ALD-Companion
Speech	71.4	72.9	93.4
Noise	67.2	70.1	69.1
SNR	4.2	2.8	24.3

Personal Communication Distance - 2.5 meters

	HA	ALD-Interview	ALD-Companion
Speech	68.7	70.9	94.0
Noise	67.2	70.1	65.3
SNR	1.5	0.8	28.7

APPENDIX C
SIGNAL AND NOISE MEASUREMENTS
MADE NEAR THE TYMPANIC MEMBRANE

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Mathematical Equation for Summing SPL Measurements to Obtain Overall SPL

$$\text{Overall SPL} = 10 \log_{10} \left[\frac{P_1^2}{P_0^2} + \frac{P_2^2}{P_0^2} + \frac{P_3^2}{P_0^2} + \dots + \frac{P_n^2}{P_0^2} \right]$$

In each frequency band, you measure:

$$\text{SPL}_x = 10 \log_{10} \left[\frac{P_x^2}{P_0^2} \right] \text{ where } P_0^2 \text{ is unknown.}$$

$$\text{whereas } \frac{\text{SPL}_x}{10} = \log_{10} \left[\frac{P_x^2}{P_0^2} \right]$$

$$\text{therefore } \frac{P_x^2}{P_0^2} = 10^{\frac{\text{SPL}_x}{10}}$$

$$\text{Overall SPL} = 10 \log_{10} \left[10^{\frac{\text{SPL}_1}{10}} + 10^{\frac{\text{SPL}_2}{10}} + 10^{\frac{\text{SPL}_3}{10}} + \dots + 10^{\frac{\text{SPL}_{65}}{10}} \right]$$

PROBE MICROPHONE MEASUREMENTS

Scores obtained at 1 meter													
Subjects	HA - Speech		HA - Noise		ALD-I Speech		ALD-I Noise		ALD-C Speech		ALD-C Noise		
	Omni-Dir	Dir	Omni-Dir	Dir	Omni-Dir	Dir	Omni-Dir	Dir	Omni-Dir	Dir	Omni-Dir	Dir	Dir
S# 1	102.38	100.57	96.01	95.05	90.12	102.47	74.39	74.63	102.16	97.28	75.56	77.23	
S# 2	97.58	96.97	91.59	88.69	85.40	97.91	70.02	69.94	96.44	89.80	70.07	70.09	
S# 3	99.49	97.58	93.40	89.90	86.94	98.02	72.06	70.46	98.37	92.16	72.85	72.41	
S# 4	96.37	94.01	90.59	86.02	85.38	97.94	70.68	70.37	94.29	88.48	67.99	78.67	
S# 5	104.21	102.56	98.67	94.94	92.96	104.91	73.94	75.97	103.38	97.96	77.43	78.54	
S# 6	97.97	96.44	91.89	88.34	86.23	99.06	71.06	71.32	96.12	90.47	71.84	71.95	
S# 7	103.37	101.69	98.21	92.67	92.38	104.87	72.86	77.85	101.98	95.29	75.03	76.79	
S# 8	102.38	101.02	97.03	92.39	89.91	101.63	72.36	72.63	100.89	95.96	74.95	75.94	
S# 9	96.97	94.54	90.04	87.05	84.63	97.38	75.86	69.84	95.12	89.04	69.12	78.65	
S# 10	97.58	96.31	91.79	88.38	85.93	97.92	71.98	69.62	96.23	90.12	69.94	71.91	
S# 11	101.59	99.47	96.68	91.09	89.16	102.10	70.89	74.36	100.89	95.48	74.81	75.84	
S# 12	97.91	96.69	92.36	88.23	86.04	98.38	74.12	69.96	95.87	89.05	69.18	77.43	
S# 13	101.87	97.28	86.04	80.07	104.38	104.54	95.41	93.61	102.85	105.64	93.57	98.43	
S# 14	99.67	97.04	79.22	79.64	101.32	102.14	79.95	79.59	98.46	101.24	81.14	78.07	
S# 15	105.43	111.75	96.53	95.25	93.21	93.56	79.24	79.88	92.60	91.69	79.94	79.25	
S# 16	105.58	103.52	91.34	88.38	106.88	105.42	90.26	91.21	105.90	106.42	88.95	89.67	
S# 17	100.60	96.55	89.01	81.16	105.30	103.50	86.60	84.73	102.49	104.23	89.91	85.59	
S# 18	103.13	103.13	91.81	89.00	106.08	104.93	93.78	95.39	105.26	91.92	92.03	91.93	
S# 19	105.43	111.75	96.53	95.25	93.21	93.56	79.24	79.88	92.60	91.69	79.94	79.25	
S# 20	105.58	103.52	91.34	88.38	106.88	105.42	90.26	91.21	105.90	106.42	88.95	89.67	
Mean	101.25	100.12	92.50	88.99	93.62	100.78	78.25	78.12	99.39	95.52	78.16	79.87	
SD	3.16	4.96	4.58	4.72	8.27	3.83	8.36	8.69	4.33	6.21	8.34	7.45	
Min	96.37	94.01	79.22	79.64	84.63	93.56	70.02	69.62	92.60	88.48	67.99	70.09	
Max	105.58	111.75	98.67	95.25	106.88	105.42	95.41	95.39	105.90	106.42	93.57	98.43	
Range	9.21	17.74	19.45	15.61	22.25	11.86	25.39	25.77	13.30	17.94	25.58	28.34	

PROBE MICROPHONE MEASUREMENTS													
Scores obtained at 1.5 meters													
Subjects	HA - Speech		HA - Noise		ALD-I Speech		ALD-I Noise		ALD-C Speech		ALD-C Noise		
	Omni-Dir	Dir	Omni-Dir	Dir	Omni-Dir	Dir	Omni-Dir	Dir	Omni-Dir	Dir	Omni-Dir	Dir	Dir
S# 1	98.56	98.56	96.01	95.05	88.46	100.16	74.39	74.63	100.87	96.46	75.48	76.63	76.63
S# 2	96.77	93.91	91.59	88.69	83.38	94.31	70.02	69.94	93.04	87.65	70.02	70.02	70.02
S# 3	97.56	94.57	93.40	89.90	84.63	95.67	72.06	70.46	96.35	90.49	72.91	71.94	71.94
S# 4	95.03	91.86	90.59	86.02	83.14	94.48	70.68	70.37	92.02	85.45	68.02	78.35	78.35
S# 5	102.24	100.29	98.67	94.94	89.43	101.36	73.94	75.97	101.96	96.04	76.38	78.92	78.92
S# 6	94.98	94.91	91.89	88.34	84.18	96.49	71.06	71.32	94.82	88.16	71.46	70.74	70.74
S# 7	100.69	99.48	98.21	92.67	89.89	101.32	72.86	77.85	100.10	93.93	76.75	75.26	75.26
S# 8	100.78	98.39	97.03	92.39	86.48	99.46	72.36	72.63	98.63	93.14	74.97	74.87	74.87
S# 9	94.63	91.38	90.04	87.05	82.78	94.24	75.86	69.84	94.29	87.90	69.18	78.04	78.04
S# 10	95.34	94.89	91.79	88.38	83.29	94.63	71.98	69.62	93.87	87.86	70.49	70.38	70.38
S# 11	98.36	96.18	96.68	91.09	87.46	99.82	70.89	74.36	98.73	93.29	74.26	75.16	75.16
S# 12	94.67	96.35	92.36	88.23	83.29	96.31	74.12	69.96	94.18	87.18	69.02	76.42	76.42
S# 13	99.33	95.66	86.04	80.07	103.22	105.25	95.41	93.61	102.13	91.46	91.46	91.63	91.63
S# 14	94.13	94.13	79.22	79.64	99.85	98.13	79.95	79.59	99.79	78.68	78.68	77.41	77.41
S# 19	104.91	107.08	96.53	95.25	90.13	92.41	79.24	79.88	90.53	91.72	80.40	79.25	79.25
S# 16	103.23	99.71	91.34	88.38	102.20	103.36	90.26	91.21	103.52	104.39	90.55	91.19	91.19
S# 17	97.46	83.58	89.01	81.16	102.58	101.68	86.60	84.73	101.61	100.67	86.53	87.11	87.11
S# 18	105.80	100.30	91.81	89.00	104.40	104.20	93.78	95.39	102.75	90.20	90.26	89.39	89.39
S# 19	104.91	107.08	96.53	95.25	90.13	92.41	79.24	79.88	90.53	91.72	80.40	79.25	79.25
S# 20	103.23	99.71	91.34	88.38	102.20	103.36	90.26	91.21	103.52	104.39	90.55	91.19	91.19
Mean	99.13	96.90	92.50	88.99	91.06	98.45	78.25	78.12	97.66	92.04	77.89	79.16	79.16
SD	3.87	5.25	4.58	4.72	8.05	4.07	8.36	8.69	4.44	6.25	7.99	7.13	7.13
Min	94.13	83.58	79.22	79.64	82.78	92.41	70.02	69.62	90.53	78.68	68.02	70.02	70.02
Max	105.80	107.08	98.67	95.25	104.40	105.25	95.41	95.39	103.52	104.39	91.46	91.63	91.63
Range	11.67	23.50	19.45	15.61	21.62	12.84	25.39	25.77	12.99	25.71	23.44	21.61	21.61

PROBE MICROPHONE MEASUREMENTS

Scores obtained at 2 meters

Subjects	HA - Speech		HA - Noise		ALD-I Speech		ALD-I Noise		ALD-C Speech		ALD-C Noise	
	Omni-Dir	Dir	Omni-Dir	Dir	Omni-Dir	Dir	Omni-Dir	Dir	Omni-Dir	Dir	Omni-Dir	Dir
S# 1	97.28	97.50	96.01	95.05	85.12	98.39	74.39	74.63	97.87	94.13	74.36	74.48
S# 2	93.92	91.50	91.59	88.69	80.65	93.37	70.02	69.94	89.59	85.49	69.90	69.79
S# 3	96.15	92.93	93.40	89.90	81.29	94.04	72.06	70.46	92.89	87.96	71.87	71.69
S# 4	93.44	88.49	90.59	86.02	80.68	92.96	70.68	70.37	89.74	84.19	69.10	77.67
S# 5	100.69	98.69	98.67	94.94	87.10	100.03	73.94	75.97	99.23	94.28	75.87	75.69
S# 6	93.12	91.60	91.89	88.34	81.49	95.39	71.06	71.32	91.29	86.51	71.34	69.84
S# 7	98.37	97.80	98.21	92.67	87.28	99.87	72.86	77.85	97.53	91.30	76.30	75.08
S# 8	98.63	95.68	97.03	92.39	84.39	97.39	72.36	72.63	95.07	90.98	74.28	74.30
S# 9	92.04	90.02	90.04	87.05	79.46	93.29	75.86	69.84	91.40	85.36	68.87	76.43
S# 10	93.62	91.67	91.79	88.38	80.81	92.87	71.98	69.62	90.62	85.02	69.43	69.05
S# 11	96.18	94.83	96.68	91.09	85.28	97.93	70.89	74.36	94.76	92.42	73.96	73.20
S# 12	91.89	94.07	92.36	88.23	80.14	94.63	74.12	69.96	92.05	83.76	68.85	75.83
S# 13	96.42	91.76	86.04	80.07	102.98	103.52	95.41	93.61	101.82	105.39	88.20	96.07
S# 14	90.74	92.03	79.22	79.64	97.02	97.48	79.95	79.59	97.95	100.44	79.35	77.10
S# 19	104.48	96.53	105.34	95.25	90.30	91.72	79.24	79.88	89.17	88.94	79.44	78.45
S# 16	100.71	98.37	91.34	88.38	100.67	101.82	90.26	91.21	103.09	100.33	90.79	91.56
S# 17	94.55	90.22	89.01	81.16	99.28	99.70	86.60	84.73	100.16	100.60	87.75	93.44
S# 18	103.33	98.01	91.81	89.00	100.42	103.05	93.78	95.39	102.62	90.33	90.31	89.75
S# 19	104.48	96.53	105.34	95.25	90.30	91.72	79.24	79.88	89.17	88.94	79.44	78.45
S# 20	100.71	98.37	91.34	88.38	100.67	101.82	90.26	91.21	103.09	100.33	90.79	91.56
Mean	97.04	94.33	93.39	88.99	88.77	97.05	78.25	78.12	95.46	91.84	77.51	78.97
SD	4.25	3.29	5.99	4.72	8.33	3.88	8.36	8.69	5.04	6.51	7.94	8.51
Min	90.74	88.49	79.22	79.64	79.46	91.72	70.02	69.62	89.17	83.76	68.85	69.05
Max	104.48	98.69	105.34	95.25	102.98	103.52	95.41	95.39	103.09	105.39	90.79	96.07
Range	13.74	10.20	26.12	15.61	23.52	11.80	25.39	25.77	13.92	21.63	21.94	27.02

PROBE MICROPHONE MEASUREMENTS													
Scores obtained at 2.5 meters													
Subjects	HA - Speech		HA - Noise		ALD-I Speech		ALD-I Noise		ALD-C Speech		ALD-C Noise		
	Omni-Dir	Dir	Omni-Dir	Dir	Omni-Dir	Dir	Omni-Dir	Dir	Omni-Dir	Dir	Omni-Dir	Dir	
S# 1	95.17	94.84	96.01	95.05	83.67	84.38	74.39	74.63	96.91	91.72	74.45	73.60	
S# 2	94.92	89.46	91.59	88.69	78.65	86.36	70.02	69.94	89.86	82.56	69.92	69.98	
S# 3	94.48	89.97	93.40	89.90	79.94	81.42	72.06	70.46	93.90	85.04	71.70	70.96	
S# 4	92.86	86.39	90.59	86.02	78.37	80.63	70.68	70.37	89.26	81.30	68.39	76.35	
S# 5	99.18	96.93	98.67	94.94	84.94	86.89	73.94	75.97	99.10	92.05	75.47	75.04	
S# 6	92.06	88.12	91.89	88.34	79.14	82.39	71.06	71.32	90.87	83.48	71.02	68.92	
S# 7	96.48	95.40	98.21	92.67	84.39	86.31	72.86	77.85	97.32	89.21	75.98	74.94	
S# 8	95.12	92.80	97.03	92.39	81.68	83.17	72.36	72.63	94.89	87.63	73.87	73.89	
S# 9	93.40	88.50	90.04	87.05	77.18	80.72	75.86	69.84	90.98	81.59	68.64	75.18	
S# 10	91.46	89.07	91.79	88.38	78.94	79.89	71.98	69.62	90.32	82.64	69.18	68.29	
S# 11	93.87	92.96	96.68	91.09	83.19	86.13	70.89	74.36	94.45	90.31	73.25	72.74	
S# 12	90.10	91.18	92.36	88.23	77.98	82.03	74.12	69.96	91.76	80.48	68.03	74.29	
S# 13	94.99	91.75	86.04	80.07	102.91	103.70	95.41	93.61	101.79	105.96	87.56	89.09	
S# 14	89.85	89.93	79.22	79.64	93.99	96.75	79.95	79.59	98.13	98.11	79.42	77.36	
S# 19	103.84	96.53	104.97	95.25	88.60	92.09	79.24	79.88	88.76	88.79	80.41	79.08	
S# 16	99.70	96.05	91.34	88.38	99.18	100.00	90.26	91.21	98.95	98.61	90.81	91.51	
S# 17	93.84	87.30	89.01	81.16	99.21	98.41	86.60	84.73	99.78	99.99	87.87	83.44	
S# 18	102.39	95.93	91.81	89.00	101.25	102.44	93.78	95.39	102.05	90.16	90.38	90.09	
S# 19	103.84	96.53	104.97	95.25	88.60	92.09	79.24	79.88	88.76	88.79	80.41	79.08	
S# 20	99.70	96.05	91.34	88.38	99.18	100.00	90.26	91.21	98.95	98.61	90.81	91.51	
Mean	95.86	92.28	93.35	88.99	87.05	89.29	78.25	78.12	94.84	89.85	77.38	77.77	
SD	4.24	3.54	5.91	4.72	8.94	8.14	8.36	8.69	4.53	7.25	8.11	7.48	
Min	89.85	86.39	79.22	79.64	77.18	79.89	70.02	69.62	88.76	80.48	68.03	68.29	
Max	103.84	96.93	104.97	95.25	102.91	103.70	95.41	95.39	102.05	105.96	90.81	91.51	
Range	13.99	10.54	25.75	15.61	25.73	23.81	25.39	25.77	13.29	25.48	22.78	23.22	

SIGNAL TO NOISE RATIO						
Scores obtained at 1 meter						
	HA		ALD-I		ALD-C	
Subjects	Omni-Dir	Dir	Omni-Dir	Dir	Omni-Dir	Dir
S# 1	6.37	5.52	15.73	27.84	26.60	20.05
S# 2	5.99	8.28	15.38	27.97	26.37	19.71
S# 3	6.09	7.68	14.88	27.56	25.52	19.75
S# 4	5.78	7.99	14.70	27.57	26.30	9.81
S# 5	5.54	7.62	19.02	28.94	25.95	19.42
S# 6	6.08	8.10	15.17	27.74	24.28	18.52
S# 7	5.16	9.02	19.52	27.02	26.95	18.50
S# 8	5.35	8.63	17.55	29.00	25.94	20.02
S# 9	6.93	7.49	8.77	27.54	26.00	10.39
S# 10	5.79	7.93	13.95	28.30	26.29	18.21
S# 11	4.91	8.38	18.27	27.74	26.08	19.64
S# 12	5.55	8.46	11.92	28.42	26.69	11.62
S# 13	15.83	17.21	8.97	10.93	9.28	7.21
S# 14	20.45	17.40	21.37	22.55	17.32	23.17
S# 15	8.90	16.50	13.97	13.68	12.66	12.44
S# 16	14.24	15.14	16.62	14.21	16.95	16.75
S# 17	11.59	15.39	18.70	18.77	12.58	18.64
S# 18	11.32	14.13	12.30	9.54	13.23	-0.01
S# 19	8.90	16.50	13.97	13.68	12.66	12.44
S# 20	14.24	15.14	16.62	14.21	16.95	16.75
Mean	8.75	11.13	15.37	22.66	21.23	15.65
SD	4.41	4.14	3.30	7.15	6.36	5.66
Min	4.91	5.52	8.77	9.54	9.28	-0.01
Max	20.45	17.40	21.37	29.00	26.95	23.17
Range	15.54	11.88	12.60	19.46	17.67	23.18

SIGNAL TO NOISE RATIO						
Scores obtained at 1.5 meters						
	HA		ALD-I		ALD-C	
Subjects	Omni-Dir	Dir	Omni-Dir	Dir	Omni-Dir	Dir
S# 1	2.55	3.51	14.07	25.53	25.39	19.83
S# 2	5.18	5.22	13.36	24.37	23.02	17.63
S# 3	4.16	4.67	12.57	25.21	23.44	18.55
S# 4	4.44	5.84	12.46	24.11	24.00	7.10
S# 5	3.57	5.35	15.49	25.39	25.58	17.12
S# 6	3.09	6.57	13.12	25.17	23.36	17.42
S# 7	2.48	6.81	17.03	23.47	23.35	18.67
S# 8	3.75	6.00	14.12	26.83	23.66	18.27
S# 9	4.59	4.33	6.92	24.40	25.11	9.86
S# 10	3.55	6.51	11.31	25.01	23.38	17.48
S# 11	1.68	5.09	16.57	25.46	24.47	18.13
S# 12	2.31	8.12	9.17	26.35	25.16	10.76
S# 13	13.29	15.59	7.81	11.64	10.67	-0.17
S# 14	14.91	14.49	19.90	18.54	21.11	1.27
S# 15	8.38	11.83	10.89	12.53	10.13	12.47
S# 16	11.89	11.33	11.94	12.15	12.97	13.20
S# 17	8.45	2.42	15.98	16.95	15.08	13.56
S# 18	13.99	11.30	10.62	8.81	12.49	0.81
S# 19	8.38	11.83	10.89	12.53	10.13	12.47
S# 20	11.89	11.33	11.94	12.15	12.97	13.20
Mean	6.63	7.91	12.81	20.33	19.77	12.88
SD	4.39	3.81	3.18	6.33	5.98	6.29
Min	1.68	2.42	6.92	8.81	10.13	-0.17
Max	14.91	15.59	19.90	26.83	25.58	19.83
Range	13.23	13.17	12.98	18.02	15.45	20.00

SIGNAL TO NOISE RATIO						
Scores obtained at 2 meters						
	HA		ALD-I		ALD-C	
Subjects	Omni-Dir	Dir	Omni-Dir	Dir	Omni-Dir	Dir
S# 1	1.27	2.45	10.73	23.76	23.51	19.65
S# 2	2.33	2.81	10.63	23.43	19.69	15.70
S# 3	2.75	3.03	9.23	23.58	21.02	16.27
S# 4	2.85	2.47	10.00	22.59	20.64	6.52
S# 5	2.02	3.75	13.16	24.06	23.36	18.59
S# 6	1.23	3.26	10.43	24.07	19.95	16.67
S# 7	0.16	5.13	14.42	22.02	21.23	16.22
S# 8	1.60	3.29	12.03	24.76	20.79	16.68
S# 9	2.00	2.97	3.60	23.45	22.53	8.93
S# 10	1.83	3.29	8.83	23.25	21.19	15.97
S# 11	-0.50	3.74	14.39	23.57	20.80	19.22
S# 12	-0.47	5.84	6.02	24.67	23.20	7.93
S# 13	10.38	11.69	7.57	9.91	13.62	9.32
S# 14	11.52	12.39	17.07	17.89	18.60	23.34
S# 15	7.95	10.12	11.06	11.84	9.73	10.49
S# 16	9.37	9.99	10.41	10.61	12.30	8.77
S# 17	5.54	9.06	12.68	14.97	12.41	7.16
S# 18	11.52	9.01	6.64	7.66	12.31	0.58
S# 19	7.95	10.09	11.06	11.84	9.73	10.49
S# 20	9.37	9.99	10.41	10.61	12.30	8.77
Mean	4.53	6.22	10.52	18.93	17.95	12.86
SD	4.19	3.57	3.11	6.21	4.87	5.71
Min	-0.50	2.45	3.60	7.66	9.73	0.58
Max	11.52	12.39	17.07	24.76	23.51	23.34
Range	12.02	9.94	13.47	17.10	13.78	22.76

SIGNAL TO NOISE RATIO						
Scores obtained at 2.5 meters						
	HA		ALD-I		ALD-C	
Subjects	Omni-Dir	Dir	Omni-Dir	Dir	Omni-Dir	Dir
S# 1	-0.84	-0.21	9.28	9.75	22.46	18.12
S# 2	3.33	0.77	8.63	16.42	19.94	12.58
S# 3	1.08	0.07	7.88	10.96	22.20	14.08
S# 4	2.27	0.37	7.69	10.26	20.87	4.95
S# 5	0.51	1.99	11.00	10.92	23.63	17.01
S# 6	0.17	-0.22	8.08	11.07	19.85	14.56
S# 7	-1.73	2.73	11.53	8.46	21.34	14.27
S# 8	-1.91	0.41	9.32	10.54	21.02	13.74
S# 9	3.36	1.45	1.32	10.88	22.34	6.41
S# 10	-0.33	0.69	6.96	10.27	21.14	14.35
S# 11	-2.81	1.87	12.30	11.77	21.20	17.57
S# 12	-2.26	2.95	3.86	12.07	23.73	6.19
S# 13	8.95	11.68	7.50	10.09	14.23	16.87
S# 14	10.63	10.29	14.04	17.16	18.71	20.75
S# 15	7.31	9.54	9.36	12.21	8.35	9.71
S# 16	8.36	7.67	8.92	8.79	8.14	7.10
S# 17	4.83	6.14	12.61	13.68	11.91	16.55
S# 18	10.58	6.93	7.47	7.05	11.67	0.07
S# 19	7.31	9.72	9.36	12.21	8.35	9.71
S# 20	8.36	7.67	8.92	8.79	8.14	7.10
Mean	3.36	4.13	8.80	11.17	17.46	12.08
SD	4.58	4.09	2.88	2.45	5.82	5.40
Min	-2.81	-0.22	1.32	7.05	8.14	0.07
Max	10.63	11.68	14.04	17.16	23.73	20.75
Range	13.44	11.90	12.72	10.11	15.59	20.68

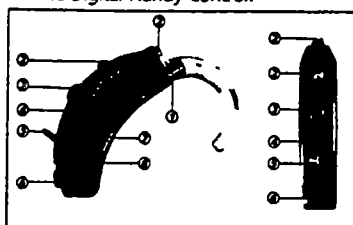
APPENDIX D
MANUFACTURER SPECIFICATIONS


PHONAK

Sono-Forte 332X AZ

Technical Data

Digitally programmable power BTE with exclusive Audio Zoom Multi-Microphone-Technology, three hearing programs and Multi-Mode-Limiting. Remote controllable via the Digital Handy Control.



- ① Special broad band receiver
- ② Multiple microphone
- ③ Volume control
- ④ Socket for programming cord
- ⑤ O-T-Off switch
- ⑥ Battery compartment
- ⑦ Serial number
- ⑧ Audio-input contacts



Technical Data

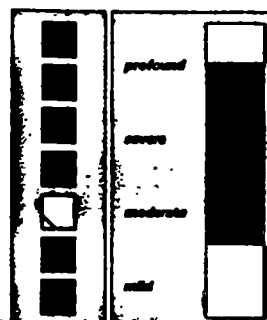
- SSPL 90 (peak): **133 dB SPL**
- HF Average SSPL 90: **128 dB SPL**
- Full-on Gain (peak): **70 dB**
- HF Average Full-on Gain Input 50 dB SPL: **66 dB**
- Reference Test Gain: **51 dB**
- Frequency Response Range: **270-6200 Hz**
- THD at 800 Hz: **2%**
- THD at 1250 Hz: **2%**
- THD at 1600 Hz: **1%**
- Equivalent Input Noise Level: **21 dB SPL**
- Battery Current Drain: **1.50 mA**
- Induction Coil Sensitivity: **117 dB SPL**
- Compression Attack: **8 ms**
- Compression Recovery without a R.T.: **80 ms**
- Compression Recovery with a R.T.: **up to 1 s**
- Audio-Input Sensitivity equal 50 dB SPL at 1000 Hz: **90 μ V**
- Input-impedance at 1000 Hz: **20 kOhm**

Special features

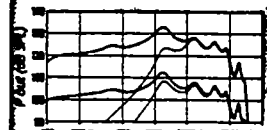
- Phonak exclusive Micro-Chip set
- Designed for use with the Digital Handy Control (DHC)
- Audio Zoom on/off function via DHC
- One basic audiological and 2 comfort programs stored in the hearing instrument
- Multi-Mode-Limiting
- 3-way audio filter
- Professional software fitting support - the Phonak Fitting Guideline
- Programming hardware options: PMC or PC (IBM compatible with Windows)
- Audio-input

User Controls

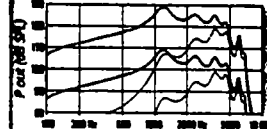
- Remote control functions with the Digital Handy Control (DHC)
- M-T-O switch (Phonak Patent) for manual control
- Manual volume control wheel



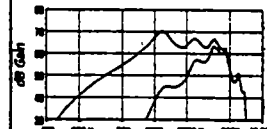
SSPL 90 Input 50 dB SPL LOW + MID max. MIN. (50 max./min. LOW + MID min.)



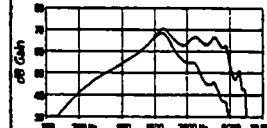
SSPL 90 Input 50 dB SPL LOW + MID max. MIN. (50 max./min. LOW + MID min.)



Max. Gain Input 50 dB SPL LOW + MID max./min.



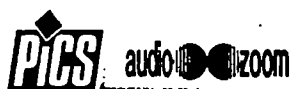
Max. Gain Input 50 dB SPL HIGH max./min.



Sono-Forte 332X AZ

Unless otherwise specified all data are obtained with the ear hook type HE 680 PC output limiting, maximum output level and no filter activation.

Measured with Phonak HE 680 Earhook according to ANSI S3.22-1987



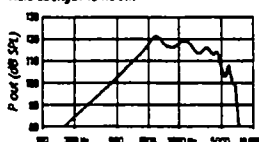
PICNAK

Sono-Forte 332X AZ

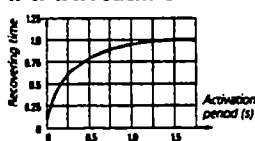
Sono-Forte 332X AZ

Technical Specifications ANSI S3.22-1987

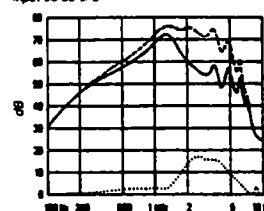
Induction Coil Sensitivity
Field Strength 10 mA/m



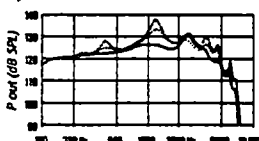
A.R.T. (adaptive Recovery Time)
depending on AEC activation period
input step
55-80-55 dB SPL at 2000 Hz



Max. Insertion Gain - KEMAR
Input 50 dB SPL

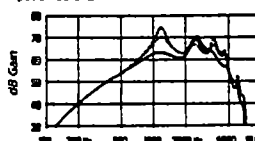


SSPL-90 with different earhooks
Input 90 dB SPL



--- HE-Hook
--- HE-680 Hook
--- PD-1000 Hook

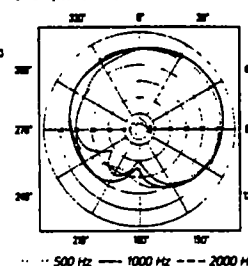
Max. Gain with different earhooks
Input 50 dB SPL



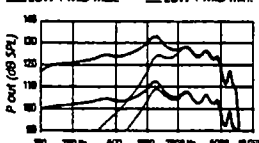
--- HE-Hook
--- HE-680 Hook
--- PD-1000 Hook

--- Max. Gain In-Situ
--- Open Ear Canal Resonance
--- Insertion Gain
--- Max. Effective Gain (difference
between curves --- and)

Polarogram
Reference test gain
DNC-1 position directional

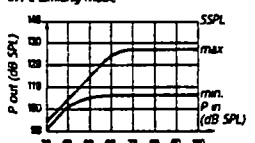


SSPL-90
Input 90 dB SPL
--- LOW + MID max.
--- LOW + MID min.

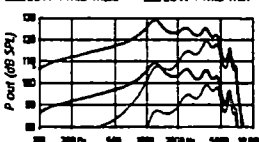


MWL (PC) max./min
--- LOW + MID max.
--- LOW + MID min.

Input/Output Characteristics
at 2000 Hz
in PC Limiting Mode

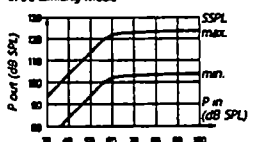


SSPL-90
Input 90 dB SPL
--- LOW + MID max.
--- LOW + MID min.



MWL (SC) max./min
--- LOW + MID max.
--- LOW + MID min.

Input/Output Characteristics
at 2000 Hz
in SC Limiting Mode



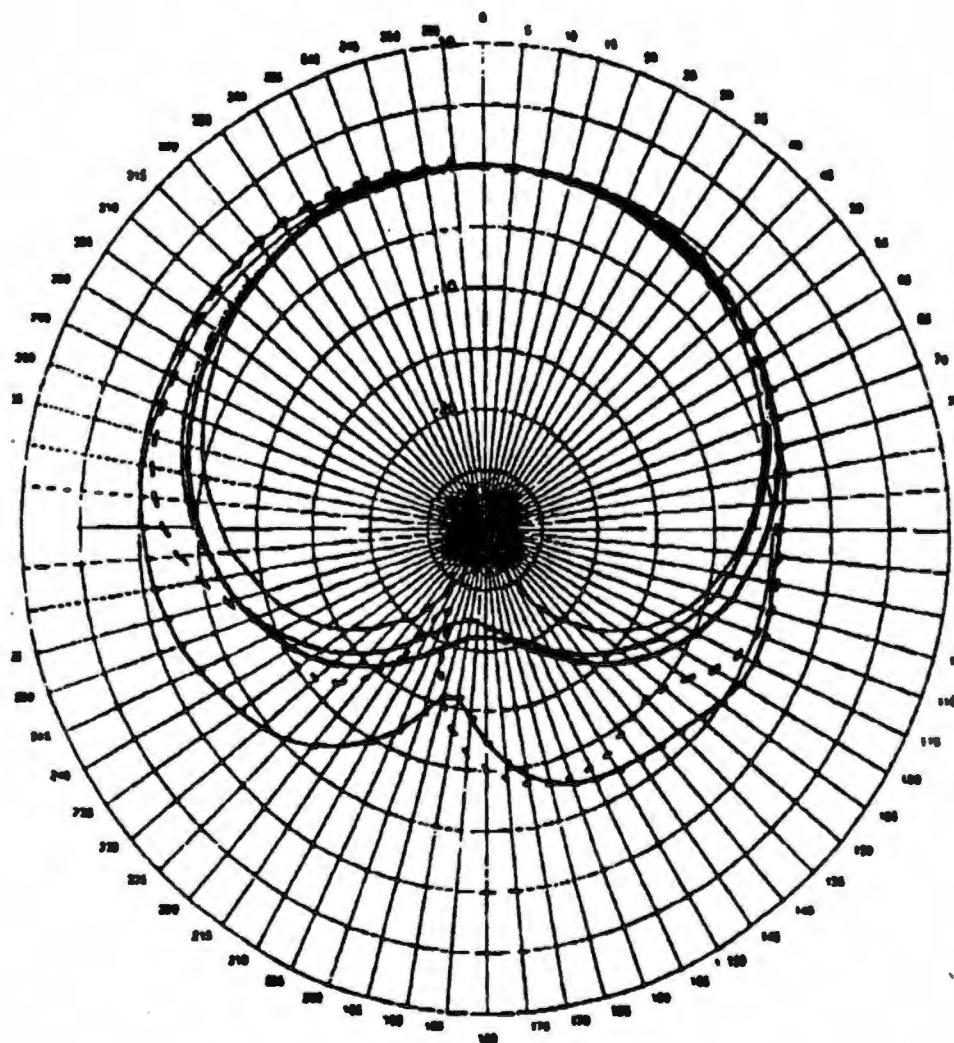
Warning to dispensers:
This hearing instrument has an output sound
pressure level that can exceed 132 dB SPL.
Special care should be used when fitting
this instrument as there may be a risk of
impairing the residual hearing of the user.

Data subject to change
without notice

US 0994

2000-01-01 PICS audio zoom

Polardigramm TX-3 Zoom



— dB 500 Hz — dB 1000 Hz — dB 2000 Hz
• • • dB 4000 Hz — dB 5000 Hz

APPENDIX E
WORD RECOGNITION SCORES
OBTAINED IN MULTI-TALKER BABBLE

WORD RECOGNITION SCORES							
Scores obtained at 1 meter in Multi-talker Babble							
	Unaided	Hearing Aid		ALD-I		ALD-C	
Subjects	---	Omni-Dir	Dir	Omni-Dir	Dir	Omni-Dir	Dir
S# 1	66.7	79.3	82.7	76.7	82.0	84.7	81.3
S# 2	42.0	71.3	83.3	77.3	84.0	85.3	82.7
S# 3	58.7	77.3	82.7	79.3	85.3	89.3	90.7
S# 4	44.7	63.3	71.3	72.7	77.3	79.3	78.7
S# 5	39.3	71.3	80.0	79.3	80.7	83.3	82.0
S# 6	48.0	74.7	81.3	82.7	90.7	91.3	88.7
S# 7	41.3	78.7	89.3	90.0	91.3	94.0	92.7
S# 8	72.0	78.7	86.0	87.3	91.3	93.3	94.0
S# 9	31.3	72.7	79.3	81.3	87.3	88.0	86.7
S# 10	63.3	80.0	86.7	88.7	92.0	95.3	94.7
S# 11	41.3	62.7	74.7	80.7	85.3	89.3	88.0
S# 12	39.3	62.0	78.7	82.7	86.0	84.7	80.7
S# 13	8.7	53.3	58.7	59.3	66.7	69.3	71.3
S# 14	3.3	52.7	60.0	61.3	64.0	74.7	76.0
S# 15	15.3	60.7	75.3	76.7	82.7	83.3	84.0
S# 16	28.7	44.0	66.7	62.7	65.3	64.0	65.3
S# 17	9.3	44.0	47.3	50.0	61.3	57.3	58.0
S# 18	31.3	76.0	82.0	89.3	96.0	94.7	92.0
S# 19	0.0	60.7	72.7	94.0	94.7	90.0	90.0
S# 20	26.7	63.3	74.7	81.3	87.3	95.3	94.7
Mean	35.56	66.34	75.67	77.67	82.56	84.32	83.61
SD	20.86	11.47	10.60	11.43	10.48	10.65	9.99
Min	0.0	44.0	47.3	50.0	61.3	57.3	58.0
Max	72.0	80.0	89.3	94.0	96.0	95.3	94.7
Range	72.0	36.0	42.0	44.0	34.7	38.0	36.7

WORD RECOGNITION SCORES							
Scores obtained at 2 meters in Multi-talker Babble							
	Unaided	Hearing Aid		ALD-I		ALD-C	
Subjects	---	Omni-Dir	Dir	Omni-Dir	Dir	Omni-Dir	Dir
S# 1	46.0	71.3	74.7	73.3	80.0	82.7	77.3
S# 2	25.3	66.0	70.0	70.7	82.7	81.3	76.7
S# 3	49.3	71.3	78.0	76.0	80.7	88.7	85.3
S# 4	39.3	56.0	64.7	68.7	74.7	77.3	74.0
S# 5	22.0	66.7	72.7	74.7	78.0	81.3	78.7
S# 6	41.3	69.3	75.3	78.0	87.3	90.7	84.7
S# 7	34.0	60.7	76.7	83.3	87.3	90.7	87.3
S# 8	54.7	69.3	82.7	78.7	86.7	88.7	91.3
S# 9	25.3	66.0	78.0	72.7	82.7	92.0	83.3
S# 10	55.3	74.7	81.3	84.0	87.3	86.7	91.3
S# 11	34.7	59.3	65.3	79.3	82.7	83.3	86.7
S# 12	28.7	57.3	66.7	77.3	81.3	79.3	76.7
S# 13	2.7	34.0	35.3	38.0	40.0	64.7	54.7
S# 14	1.3	18.0	20.7	51.3	52.7	65.3	63.3
S# 15	7.3	41.3	63.3	73.3	80.7	83.3	82.7
S# 16	12.7	34.7	36.7	60.0	65.3	76.0	79.3
S# 17	0.0	34.0	41.3	61.3	66.7	71.3	74.7
S# 18	19.3	55.3	82.0	80.7	75.3	83.3	87.3
S# 19	0.0	32.7	52.0	90.0	85.3	92.0	88.7
S# 20	16.7	57.3	69.3	77.3	78.7	93.3	90.7
Mean	25.80	54.76	64.34	72.43	76.81	82.60	80.74
SD	18.30	16.41	17.80	12.01	12.24	8.42	9.37
Min	0.0	18.0	20.7	38.0	40.0	64.7	54.7
Max	55.3	74.7	82.7	90.0	87.3	93.3	91.3
Range	55.3	56.7	62.0	52.0	47.3	28.6	36.6

APPENDIX F
INFORMED CONSENT FORM

**INFORMED CONSENT FOR A RESEARCH PROJECT ENTITLED
“HEARING AIDS AND ASSISTIVE LISTENING DEVICES WITH
DIRECTIONAL MICROPHONES USED AT PERSONAL COMMUNICATION
DISTANCES IN LISTENERS WITH MODERATE TO SEVERE HEARING
IMPAIRMENT”**

PROJECT OBJECTIVE:

This study is being conducted by the principal investigator as part of doctoral dissertation. The objective of this project is to compare the performance of two types of amplification systems for listeners with moderate to severe hearing-impairment and the listeners ability to understand speech in quiet and in the presence of background noise. The first system is two behind-the-ear (BTE) hearing aids, and the second system is an assistive listening device coupled to the BTE hearing aids. This study will require two visits to the Audiology Clinic. This project has been approved by the University of Tennessee Committee on Research Participation and Lamar University Institutional Review Board.

QUALIFICATION AND HEARING AID FITTING:

SESSION ONE: You will be seated in a comfortable chair inside a sound treated room and given a hearing test to reassess your hearing for the purposes of this project. If your hearing qualifies you to participate, you will be fitted with one of the two types of amplification systems in accordance to standard clinical procedures. Speech and noise level measurements will be made near your tympanic membrane to assess the intensity level following amplification. You will be seated in a sound treated room and listen to four 50-word recorded word lists. You will hear the words in quiet or in the presence of a multi-talker babbling noise through a sound field loudspeaker. The speech and noise will vary to simulate normal conversational levels. You will be asked to repeat words for about five minutes after which you will be given a short break for several minutes.

SESSION TWO: Procedures for session two are identical to those in session one, utilizing the second and third amplification systems.

RISKS AND BENEFITS:

There are no significant risks associated with this project. The benefit of this project is that you will be contributing to the knowledge base on alternative methods of amplification.

PRIVACY OF RECORDS:

Your research data record will be identified by an assigned research number. Your name and research number will be kept confidential in a locked file. The research data record will include:

- 1) Your age and gender
- 2) Results of the qualifying hearing tests, and

3) Results of speech understanding tests.

The results of the research will be presented at scientific meetings and published in scientific journals. Your research data will be identified by number only.

YOUR PARTICIPATION IS COMPLETELY VOLUNTARY.

YOU MAY WITHDRAW FROM PARTICIPATION AT ANY TIME WITHOUT PENALTY AND IT WILL NOT AFFECT ANY FUTURE SERVICES RENDERED BY THE LAMAR UNIVERSITY AUDIOLOGY CLINIC.

WHO TO CONTACT IF YOU HAVE CONCERNS:

If you have any concerns, please contact the principal investigator Lynn Norwood, M.A. at (409) 880-8170 or the faculty advisor James W. Thelin, Ph.D. at (423) 974-1796.

PERMISSION TO PARTICIPATE:

I have read and understand the above information. I have received a copy of this form. I agree to participate in this study.

Signature of Participant

Date

Signature of Principal Investigator

Date

VITA

Melissa Lynn Norwood was born in Eden, North Carolina on July 8, 1970. She attended schools in the public school system of Eden City Schools, where she graduated from Morehead High School in June, 1988. She entered Appalachian State University in Boone, North Carolina in August, 1988 and received her Bachelor of Science degree in Communication Disorders in May, 1992. She then entered the Master's program in Audiology at The University of Tennessee, Knoxville in August, 1992. After receiving her Master's in December of 1993, she pursued her career as an Audiologist working in an otolaryngology practice in Greensboro, North Carolina. Lynn returned to The University of Tennessee, Knoxville, in August of 1995 to pursue the Doctorate of Philosophy in Speech and Hearing Science. While in the doctoral program, she participated in the Department of Veterans Affairs pre-doctoral fellowship at the VAMC Mountain Home, Tennessee. In January, 1998, Lynn joined the faculty at Lamar University, Beaumont, Texas.