THE VALIDATION OF A SOUND FIELD SIMULATOR FOR MEASUREMENT OF THE REAL WORLD PERFORMANCE OF DIRECTIONAL MICROPHONES FOR HEARING AIDS

By

CYNTHIA LYNN COMPTON

A dissertation submitted to the Graduate Faculty in Speech and Hearing Sciences in partial fulfillment of the requirements for the degree of Doctor of Philosophy, The City University of New York

2002

UMI Number: 3047206

ProQuest Information and Learning Company
300 North Zeeb Road
P.O. Box 1346
Ann Arbor, MI 48106-1346

This manuscript has been read and accepted for the Graduate Faculty in Speech and Hearing Sciences in satisfaction of the dissertation requirement for the degree of Doctor of Philosophy.

April 25, 2002
Date

Chair of Examining Committee

April 25, 2002
Date

Executive Officer

Harry Levitt

Mead C. Killion

David A. Preves

THE CITY UNIVERSITY OF NEW YORK

Abstract

THE VALIDATION OF A SOUND FIELD SIMULATOR FOR MEASUREMENT OF THE REAL WORLD PERFORMANCE OF DIRECTIONAL MICROPHONES FOR HEARING AIDS

By

CYNTHIA LYNN COMPTON

Advisor: Professor Arlene Neuman

This investigation compared the effectiveness of four test environments in assessing real-world directional microphone performance: Two traditional sound booth environments, a real restaurant, and a newly developed sound-field simulation system consisting of an octagonal array of eight loudspeakers.

Tested with KEMAR recordings of three hearing aid microphones having radically different directional characteristics, normal hearing subjects required nearly identical signal-to-noise ratios (SNRs) for 50% correct performance in either the real or the simulated environment and scores improved as directivity increased. This did not occur in the traditional settings. The rank ordering of hearing aid microphone performance in the test booth with a single noise source overhead (90^0) was similar to the diffuse environments, with the exception of significantly better performance for the array microphones (which had pickup nulls overhead). Compared to the other three conditions, performance in the test booth with a single source behind (180^0) was slightly worse for the omnidirectional microphones, similar for the super-cardioid

microphones (D-MIC), and significantly worse for the array microphones. Performance was attributed to the fact that the omnidirectional microphones had slightly better pickup from behind than from overhead and the array microphones had a large pickup lobe located directly behind.

It was concluded that, if one's goal is to assess hearing aid microphone performance under realistic conditions where speech is directly in front (0^0) of the listener and ambient noise is more or less diffuse and surrounds the listener (e.g. in a restaurant or other social situation), then the simulator technique comes closer to imitating such a situation than do the other conditions - and does so using real, unaltered restaurant noise. Further, the efficacy of assessing hearing aid microphone performance by placing the target speech signal directly in front of the listener and using a single competing noise source either behind or directly overhead is unrealistic and can provide a contrived performance advantage or deficit, depending on the polar pattern of the microphone. The major strength of the simulator system is that a realistic estimate of the in-noise performance of any hearing aid microphone can be assessed without having detailed information about its polar pattern.

Dedication

To the memory of my father, William Russo, who last loving words advised me to "hangeth in there" . . . and to Arlene and Mead, who helped me keep my grip.

Acknowledgements

I've visualized writing this section numerous times during the process, especially during those times when I wondered if I would ever finish in this lifetime! But I did finish and now I have the great pleasure of acknowledging those who helped make this dream come true.

This dissertation was a large undertaking and could not have been carried out without a team effort. First and foremost, I'd like to thank Etymotic Research for providing me with the equipment, facilities, and personnel needed for the majority of the work related to this project. With the exception of data collection for the main investigation, most of the pilot work and acoustic analysis was carried out on the premises of Etymotic Research in Elk Grove, Illinois. Thanks goes to Bob Schulein and Diane Green for their help with the microphone measurements and to Gail Gudmundson for assistance in securing participants for the pilot project. A debt of gratitude goes to Larry Revit, sound engineer par excellence. Larry's expertise facilitated the many tracks of recordings and mixings necessary for token preparation. If it were not for Larry, I would still be trying to lay cable. I'd also like to thank the employees of Etymotic Research for serving as subjects, as restaurant party noise, and as my extended family during my many stays in Elk Grove during the last three years.

Maintaining a new faculty position and "dissertating" requires a strong support system and I'd be remiss if I did not acknowledge those individuals who have cheered me on and helped ease the workload by providing materials to assist in the development of a seemingly endless number of new course preparations. Thanks to Ruth Bentler, Mike Valente, Dave Chandler, Todd Ricketts, Dawna Lewis, Laurel Christiansen, David Hawkins, Dave Fabry, and Dave Preves I was able to make headway on the proposal and pilot study.

I also would like to give special thanks to Fred Brandt, audio engineer at Gallaudet University. A good friend, Fred has been a source of consistent encouragement through the years. Whether it was assisting in the set-up of the data collection lab or figuring out how to de-bug email, Fred was there. I'd also like to thank my colleague and good friend, Lisa Devlin, for teaching one of my summer classes so I could take some time off to work on the pilot study. Finally, I'd like to acknowledge Lee VanMiddlesworth for his encouragement, friendship and computer support.

A big thank you goes to a couple of unofficial committee members. First I'd like to thank Arthur Boothroyd, who has always been just an email or phone call away to discuss the nuances of research design. I don't know if Arthur realizes how much his help and support has meant to me. He's gotten me out of many statistical quandarys and has taught me a great deal in the process. I'd also like to thank Dave Preves, my outside reader. No matter what I needed or how busy he was, Dave was always there to help out. He even read my dissertation while he was in the islands on vacation. (I have a photo to prove it.) I'd also like to thank Wim Soede who spent most of a Sunday helping me make sense of some interesting test results and supplied me with an abundance of helpful information on various methods used to calculate AI-weighted frequency responses.

It's not often that people have this happy circumstance, but I am pleased to brag that I had the best Dissertation Committee possible. Even the defense was fun. A big thank you goes to committee member Harry Levitt who was instrumental in defining the research design. Harry is one of the most remarkable people I know. A brilliant mathematician and engineer, Harry has that ability to show unconditional positive regard for people. No matter how significant or insignificant my question was, he was always there to answer it in a patient and enlightening fashion. Mead Killion, President of Etymotic Research, also served as a member of my committee and is the sole reason why this project even got off the ground. A true scientist and humanitarian, Dr. Killion has supported and continues to support independent graduate research. He is that rare individual who can run a company, yet adhere to the important scientific principle of impartiality. In fact, he is the only person I have ever met who has actually attempted to pay back an NIH grant. Mead has been a true mentor to me in so many ways. I will never be able to repay the debt I owe him except perhaps to return the favor to my own students someday. Finally, I am forever indebted to Arlene Neuman, my advisor. Arlene's interest in seeing me complete this project never waned through this very long haul. No matter how busy she was, Arlene has always given freely of her time and expertise. A top-notch researcher, professor, and writer, I have been able to benefit from all of her gifts.

It's been blessing to have been afforded so much of my Committee's time. Not only have I ended up with a nice dissertation, but I have gained some dear, life-long friends in the process.

Finally, I would like to thank my fiancé, Jeffrey Conley, for his patience, encouragement, love and support. Thankfully, as U.S. Naval Academy graduate and retired Naval Commander, he understands and supports the concepts of self-sacrifice and perseverance. Now we can spend more time sailing.

TABLE OF CONTENTS

LIST OF TABLES

LIST OF FIGURES

INTRODUCTION AND PURPOSE

Background

Directional microphone hearing aids (DMHAs) have been shown to improve speech perception in acoustic environments containing noise and reverberation under laboratory test conditions. However, it has been demonstrated that directivity decreases in real world environments (Hawkins & Yacullo, 1984; Nielsen & Ludvigsen, 1978; Studebaker, Cox, & Formby, 1980). Since DMHAs are becoming more popular in the marketplace, research and clinical facilities require an efficient and practical method to predict how well DMHAs will perform in real acoustic environments (such as restaurants, living rooms, classrooms, churches, and in automobiles) characterized by the presence of both noise and reverberation.

Objective

The purpose of this investigation was to compare several methods used to predict real-world performance of hearing aids with varying amounts of directivity. The performance of several binaurally fitted hearing aid microphones (having various levels of directivity) were assessed in several test environments. The first environment was that of a real restaurant. The second was the newly developed multi-microphone recording and multiple loudspeaker playback system designed to capture and closely reproduce as well as simulate the sound field present in selected real world environments. The third environment was one often used to measure the performance of a DMHA. In this environment, speech is presented in a sound treated room from a loudspeaker placed at 0^0 azimuth to the listener. The noise is presented from a single loudspeaker placed directly behind the listener at an azimuth of 180^0. The fourth environment was a sound treated room in which speech was presented from the front and noise was presented from a single loudspeaker placed directly above the listener. This is the method advocated by Mueller and Sweetow (1978).

The study was carried out to test the hypothesis that the speech reception threshold in noise with a particular hearing aid microphone measured in the simulator/reproducer, would be similar to the speech reception threshold in noise measured in the real restaurant. It also was hypothesized that the simulator/reproducer technique would provide more accurate information about performance in the real world than would the more traditional (competing) techniques of evaluation.

DEFINITION OF TERMS

Articulation Index (AI) –It is generally assumed that speech recognition will be optimized when audibility is maximized. Articulation theory (Fletcher, 1953) is a model that quantifies the relationship between audibility and speech recognition performance. In this model the Articulation Index (AI) is used as an intervening measure that relates speech recognition to the audibility of the speech spectrum. Speech spectrum audibility is calculated from the hearing thresholds of the listener, and the long-term average spectra of the speech and the noise. The AI is designed to predict the intelligibility of speech under noisy and/or frequency limiting conditions and is equal to the proportion of the speech spectrum above the threshold of hearing or above background noise, whichever is higher. Since some frequency regions have a higher density of speech cues than others, the AI uses different weights at difference frequencies. The Articulation Index is based on several basic assumptions: (a) the speech spectrum can be divided into a number of contiguous frequency regions that each make an independent contribution to intelligibility, (b) contributions of the frequency regions are additive and, (c) percent intelligibility is a monotonic function of the AI. The first major description of the AI was offered by Fletcher (1953) and subsequently expanded by French and Steinberg (1947). The computational procedures and parameters were standardized according to ANSI S3.5-1969 (1969). The current version of the *Standard, American National Standard Methods for Calculation of the Speech Intelligibility Index* (ANSI, 1997), reflects new data accumulated since 1969 regarding spread of masking, standard speech spectrum level, and relative importance of various frequencies to speech intelligibility. The Speech Intelligibility Index defines a method for computing a physical index that is highly correlated with the intelligibility of speech under a variety of adverse listening conditions such as noise, filtering and reverberation.

Articulation-Index-weighted Directivity Index (AI-DI) - This is a modification of the Directivity Index (see below) to allow prediction of speech recogntion performance. The directivity of a microphone at each frequency is weighted by the AI band importance functions (Killion et al., 1998). Several band importance weighting systems may be found in the ANSI S3.5-1997 (see Articulation Index, below). An additional method for calculating the AI-DI has been suggested by Mueller and Killion (1990) whereby the importance functions are taken from the Mueller-Killion audibility index (500Hz = 20%; 1000Hz = 23%; 2000Hz = 33%: and 4000Hz = 24%). In general, the AI-DI uses band importance weightings to assign more weight to the directional advantages for the frequencies that contribute the most to intelligibility. Instead of using a separate number at each frequency, the AI-DI provides a single average number that is predictive of the improvement in speech intelligibility resulting from the directional characteristics of the microphone. For example, with the Mueller-Killion method, if a hearing aid has DI of 3 dB for all four of the key frequencies, then the AI-DI is 3.0 (3 X 0.20 (500Hz) + 3 X 0.23 (1000Hz) + 3 X 0.33 (2000Hz) + 3 X 0.24 (4000Hz). The AI-DI is equal to the equivalent improvement in SNR (in dB) that would result if the noise were attenuated by that amount without changing its spectrum.

Ambient Noise – The all-encompassing noise associated with a given environment, being usually a composite of sounds from many sources, near and far (ANSI, 1973).

Background Noise – The total of all undesired signals in a system used for the production, detection, measurement, or recording of a signal, independent of the presence of the signal (ANSI, 1973).

Critical Distance – The distance from the sound source at which the steady state direct and reverberant sound powers are equal.

Diffuse Sound Field – A sound field in which the time average of the mean-square sound pressure is the same everywhere and the flow of energy in all directions is equally probable (ANSI, 1976).

Directional Microphone – A microphone whose response varies significantly with the direction of sound incidence (ANSI, 1976).

Directivity Factor (of a microphone) – The directivity factor for an arbitrary frequency is the ratio of two electrical powers delivered to a resistor by two microphones located in a diffuse sound field: (1) The power delivered by the microphone under test, and (2) the power delivered by a non-directional microphone whose response is equal to the principle-axis response, typically 0 degrees or front, of the microphone under test. The measurement can be made at specified equally distributed test points (azimuths) in the horizontal plane (elevation equal to 0 degrees) or in both the horizontal and vertical planes. The number of test points is determined by the complexity of the directional response (Roberts and Schulein, 1997; ANSI S3.35, 2001).

Directivity Index (DI) – This measure takes the Directivity Factor and converts it to dB. Traditionally, the DI is derived by taking a complex set of measurements at an equal distance from the microphone at a variety of places around an imaginary sphere (Beranek, 1954). A three-dimensional polar pattern measure in a free field is used for the calculation (Ricketts & Mueller, 1999). The DI also may be calculated as the ratio of the root mean squared output pressure from a hearing aid in response to a signal originating directly in front of the microphone in a free field to the output in a diffuse field (usually a reverberation chamber). For hearing aid applications, the DI is often simplified by assuming symmetry in the vertical plane and performing the calculation based only on the two-dimensional polar pattern measured in the horizontal plane. Roberts and Schulein (1997) have shown that the calculated two-dimensional DI scores provide a reasonable approximation of true three-dimensional measures.

Front-to-Back Ratio (FBR) –This is the "simplest index for quantifying directionality electroacoustically" (Ricketts & Mueller, 1999). Measured in dB, it is the output of a hearing aid in response to a loud speaker placed at 180^0 azimuth, subtracted from the output of the hearing aid in response to a loud speaker placed at 0^0 azimuth. It can be measured in the clinic by using probe-microphone equipment, using the differences between the real ear aided responses taken from 0^0 and 180^0 azimuths. It also can be obtained in the laboratory, using a Zwislocki coupler mounted in a Knowles Electronics Manikin for Acoustic Research (KEMAR) (e.g., Hawkins & Yacullo, 1984: Madison & Hawkins, 1983). Hearing aid companies who sold cardioid microphones invented this measurement 25 years ago. Since the null for a cardioid polar response is at 180^0, the FBR measurement will look favorable. Unfortunately, FBR is a poor measurement for supercardioid and hypercardioid polar patterns that provide an improved directivity index overall, but have lobes instead of nulls at 180^0. This is why the Directivity Index has replaced the FBR.

Multi-Microphone Array Recordings: Room recordings made by employing an array (eight in the case of this dissertation) of interference tube (shotgun) microphones positioned at eight different horizontal azimuths surrounding KEMAR and fed to eight separate tracks of a DAT recorder.

Noise – An undesired sound within a useful frequency band, such as undesired electric waves in a transmission channel or device. If ambiguity exists as to the nature of the noise, the term "acoustic noise" or "electric noise" should be used (ANSI, 1973).

Omnidirectional Microphone – A microphone whose response, when tested in the free field, provides equal output for sounds originating from all directions (Ricketts & Mueller, 1999).

Polar Directivity Pattern (PDP or Polar Plot): A graphical plot of microphone (or hearing aid) output as a function of the angle of sound incidence. This plot is obtained by recording the microphone's or hearing aid's output level (or attenuation relative to that at 0^0 azimuth) at discrete angles as it is turned in a circle (horizontal plane) relative to the loudspeaker in an anechoic chamber. Concentric reference lines emanating from the center of the polar plot are graduated in dB. Shapes of polar patterns show how the gain of a hearing aid varies with position of sound source. The measures of directivity factor and directivity index are derived from polar patterns. For example, omnidirectional, bi-directional, cardioid, supercardiod, and hypercardioid are several examples of directionality. The numerical metrics for rating directionality are designed to take into account the noise-suppressing capabilities of a microphone in three dimensions. However, in practice, symmetry is usually assumed in the third dimension (i.e., the polar pattern in the vertical plane is assumed to be same as that in horizontal plane), and the calculations are frequently performed in only two dimensions (Ricketts & Mueller, 1999).

Traditionally, PDPs are measured in an anechoic chamber, although a room with a diffuse field may also be used. While PDPs obtained under anechoic conditions provide information about the noise rejection a microphone will have for sound arriving at specific angles of incidence under low reverberation listening conditions, PDPs obtained in a diffuse sound field will measure the noise rejection of a microphone for a worst case--high reverberation--listening environment (Roberts & Schulein, 1997).

A free-field PDP means that the measurement is performed with the hearing aid microphone(s) suspended in the test room using a stand or table and a mounting device that is acoustically small compared to the hearing aid assembly under test (dimensions of the holder being small compared to the shortest wavelength of sound being measured). Practically speaking the hearing aid assembly could be mounted on a 1/4 " diameter vertical rod that would be used for rotation of the microphone through the various test azimuths. Since free-field polar measures do not reflect the effects of head and body reflection and diffraction, the PDP also can be obtained by placing the hearing aid microphone(s) on KEMAR equipped with a Zwislocki coupler.

Data obtained from microphone polar measurements at each frequency in a horizontal plane can be used to calculate a single number approximating the directivity index (DI) (described previously) of the microphone at that frequency. The AI-DI (discussed previously) can be calculated by weighting the directional performance at each frequency by the number of speech cues contained in that frequency band (typically 1/3 octave). The *AI-weighted Polar Plot* is a newer metric: Twelve polar plots are obtained in the horizontal plane, one at each 1/3-octave frequency. The DI for each polar plot is multiplied by the appropriate AI weight for that frequency. The twelve products are then averaged to produce the AI polar plot -- the overall ability of the microphone to reject noise coming from a given angle (Etymotic Research, 2001).

Reverberation – The persistence of sound in an enclosed space, as the result of multiple reflections after the sound source has stopped (ANSI, 1973).

Reverberation Time – The time required for the mean-square sound pressure level originally in a steady state, to decrease by 60 dB after the source has stopped (ANSI, 1973).

Signal-to-Noise Ratio (SNR) - The ratio (in dB) between the desired signal (e.g. speech) and the competing signal (e.g. undesired speech or other background noise).

Simulated Sound Field – A sound field that is generated independently having the characteristics of the sound field being simulated (Levitt, 2000).

Sound Field - A region containing sound waves (ANSI, 1973).

Speech Recognition Threshold - A common way of evaluating a person's ability to follow conversational speech in noise is to hold one of the signals constant and to vary the level of the other signal until the listener achieves a 50 percent criterion score. Hearing aid microphones having various amounts of directivity can be compared in this way. As the hearing aid becomes more directional, the SNR required for 50 percent performance is decreased.

Unidirectional Index (UI) (in dB): This metric compares the output of signals originating from two different hemispheres. It is the ratio of the microphone output in response to sound sources placed in front of a listener (270 through 90 degrees azimuths) to microphone output in response to sound sources placed behind a listener (90 through 270 degrees azimuths). According to Ricketts and Mueller (1999), while this technique "may provide a more representative description of directivity than FBR, it is not ideal if it is assumed that the listener usually faces the talker directly and that noise may originate in the "front" hemisphere."

HOW DIRECTIONAL MICROPHONES WORK

Before the literature review is presented, the reader will be provided with an explanation of how directional microphones work, how they are classified, and how they are measured electroacoustically.

In general, when tested in a free field, omnidirectional microphones provide equal output for sounds originating from all directions. Directional microphones, on the other hand, are distinguished from omnidirectional microphones in that their output level is dependent upon the direction of sound origination.

Microphones can be classified in accordance with their directionality as opposed to, for example, their method of converting sound energy into electricity. All microphones have diaphragms that move back and forth in response to changes in the differential air pressure acting on the diaphragm. As the diaphragm moves back and forth, the transduction mechanism converts this movement to an output voltage that is proportional to the air pressure induced movement.

In an omnidirectional microphone, air is effectively trapped on one side of the diaphragm with the exception of a small air leak to allow for normal changes in atmospheric air pressure. This type of microphone has equal sensitivity for sounds arriving in all directions.

Conventional (Single Diaphragm) Directional Microphones

A conventional, or single diaphragm directional microphone responds to the pressure gradient or difference in sound pressure operating on either side (front and back) of its diaphragm. Two sound ports lead to a small two-chambered cavity divided by the diaphragm. An acoustical resistance at the back port, commonly fabricated from fine mesh cloth-like materials, acts in combination with the elasticity formed by the air in the rear cavity of the microphone to produce an acoustic delay or phase shift for sounds entering the rear port of the microphone. This time delay is determined as a part of the microphone design. As an example, if the value equals that of the transient time between front and rear sound ports, the resulting directional microphone will have an acoustical null, or minimum, for sounds arriving at 180 degrees. Front and rear sound waves acting on the diaphragm will have equal amplitude, but opposite polarity for that orientation. This is the classic cardioid pattern. By varying this time delay, the null angle can be adjusted to produce other pick-up patterns. Shorter delays, for example, will move the pattern through super cardioid toward bi-directional (which results when there is no internal time delay). The effect of this process is to reduce unwanted background noise relative to speech sounds arriving from in front of the microphone, thus allowing for an improvement in the SNR.

Additional ports have been used in directional microphone design to refine directional performance by reducing a phenomenon known as proximity effect (excessive bass build up as a sound moves closer to the microphone). In addition, higher order directional microphones have been built using this method. For example, the Sennheiser microphones used to record the restaurant noise in this study (commonly called "interference tube" or "shotgun" microphones) typically combine a multiple front sound port design with a conventional cardioid or super cardioid microphone element (Ricketts & Mueller, 1999; Schulein, 2000).

Multiple Microphone Techniques (Arrays)

The directional characteristics of hearing aids also can be improved using a spatial distribution of multiple microphones (also called "microphone elements"), each of which has a single diaphragm. Such techniques are used in other field such as astronomy, sonar, radar, and seismology. Because the sensors, and in some applications also the sources, are often placed in a straight line, the method is referred to as "array-techniques" or "array-processing." Many modern directional microphone hearing aids use at least two or more microphones and employ electronic (as opposed to mechanical) cancellation to form a directional pick-up pattern (Soede, 1990).

Dual Microphone Arrays

A dual microphone array may be constructed by combining the output voltages of two omnidirectional microphones whose sensitivities are matched in phase and amplitude and by using an electronic time delay to process the signal for sounds entering the rear microphone before the output is summed and sent to the pre- amplifier (the output signal from the designated "rear" microphone is passed through the time delay before it is electronically subtracted from the output from the "front" microphone). The hearing aid industry has applied array technology into their dual microphone systems and has marketed it as "dual microphone" technology. In actuality, it is the "simplest array-form (Soede, 2001)." Researchers in the array technology field call this type of microphone system a "Jacobi Array" (Weston, 1986).

Arrays with Three or More Microphones

Directional hearing aids using three or more microphones to achieve directionality are commonly called array microphones. As with the simpler dual microphone arrays, the microphones are electronically combined together and mechanically and electrically tuned (via weighting and delays) to provide increased directionality. Array microphones have been designed in various configurations. Only the Etymotic Research array microphone based on the work of Wim Soede (Soede, Berkhout, & Bilsen, 1993) will be discussed here, since it is one of the directional microphones used in this investigation. In 1993, Soede and colleagues reported the development and optimization of both a broadside and an endfire array microphone. In the broadside configuration, five miniature directional (cardioid) electret microphones are spaced over a length of 14 cm. In the endfire configuration, the five microphones are placed in a housing that can be placed alongside the head above the pinna. Thus, the broadside configuration can be attached or built into the front portion of eyeglasses, while the endfire configuration can be built into or attached to an eyeglass temple (or placed over the ear like a pencil, or worn as a barrette). The pencil-like array was used in this study. Microphone arrays with high directivity are sometimes referred to as beamforming arrays. The pickup "beam" can be fixed or can adapt with the environment (adaptive beamforming). The Soede array microphone is of the fixed variety.

Unless compensated for, a directional microphone will usually provide less gain in the low frequencies than the same microphone with an omnidirectional pattern. This occurs because such microphones respond to the sound pressure difference acting on a diaphragm with a fixed sound port spacing. At low frequencies where wavelengths are long (relative to the port spacing) the pressure differential is small. The similarity of pressure condensations and rarefactions at the two microphone ports prevents the microphone diaphragm from moving, thus canceling out sound and resulting in a reduction in low frequency gain (Preves, 1997). At higher frequencies the differential pressure is larger. This results in a rising frequency response. This rise in response is often compensated for by means of electronic equalization.

LITERATURE REVIEW

Directional Microphone Hearing Aids as a Means for Improving Speech Understanding

A major complaint of listeners with hearing loss is of increased difficulty understanding speech amidst background noise and/or reverberation (Kochkin, 1993; Kochkin, 1994). This is particularly the case in social situations such as family gatherings, meetings, parties, and restaurants where the background noise level can become quite high. Plomp (1978) has presented a quantitative model of sensorineural hearing loss. This model described cochlear damage in terms of a communicative handicap resulting from the sum of the (1) attenuation and (2) distortion of sounds. This distortion of sound results in decreased speech intelligibility in noise, despite compensation for the attenuation of sound by hearing aids. Thus, another solution must be found that will compensate for the reduced speech recognition in noise. This solution must be based on the reduction of undesired sounds that compete with the desired speech.

Several studies on speech intelligibility in noise have revealed that people with sensorineural hearing loss may require signal-to-noise ratio (SNR) improvements of 5-15 dB in order to perform at the same criterion level on a speech test as people with normal hearing (Dirks, Morgan, & Dubno, 1982; Killion, 1997a; Killion, 1997b; Soede et al., 1993; Tillman, Carhart, & Olsen, 1970). According to several investigations, every dB of signal-to-noise ratio (SNR) improvement translates into an 8.5% to 15% improvement in speech intelligibility, depending on the exact speech materials used (Killion & Villchur, 1993; Plomp & Mimpen, 1979; Soli & Nilsson, 1994; Voss, 1997).

Traditionally, hearing aids have been manufactured using omnidirectional microphones, thereby providing essentially equal amplification for sound arriving from any azimuth (modified of course by the microphone's position on the head). In a noisy environment, such a microphone will not only receive and transmit the desired speech (or other) signals, but also the disturbing background noise. Behind-the-ear (BTE) directional microphone hearing aids (DMHAs) were first introduced to the U.S. market in 1971 in order to facilitate improved hearing aid performance in noise (Agnew, 1999). By 1980, DMHAs comprised 20% of the hearing aids sold (Mueller, 1981). Over the next several years the use of DMHAs decreased steadily, due in part to the increasing popularity of custom in-the-ear (ITE) hearing aids. Another reason for the demise of DMHAs was the fact that many audiologists did not believe that these hearing aids significantly improved their clients' hearing in noise in every day life (Ricketts et al., 1999). Notwithstanding validation studies to the contrary (Mueller, Grimes, & Erdman, 1983), these audiologists may have been right. Studies have shown that DMHAs do not perform as well in real rooms characterized by reverberation. Several studies (Compton, 1974; Nielsen, 1973; Nielsen et al., 1978; Studebaker et al., 1980) showed that the directional microphone's advantage disappeared when worn in the field. Other studies also revealed that the directional microphone advantage decreased with increasing reverberation (Hawkins et al., 1984; Madison & Hawkins, 1983; Ricketts & Dhar, 1999a; Ricketts, 2000a).

In the past few years, DMHAs have enjoyed resurgence in their popularity due to the technological advances that have resulted in increased directivity.

Research suggests that today's DMHA can improve the SNR more effectively than ever before (Agnew & Block, 1997; Killion et al., 1998). Further, miniaturization of components has made it possible to manufacture directional ITE instruments. Not only are these instruments more popular than behind-the-ear directional instruments (due to more acceptable cosmetics), but they also have been reported to provide equal or greater success in SNR enhancement (Preves, 1997).

Methodological and Other Factors Affecting DMHA Performance

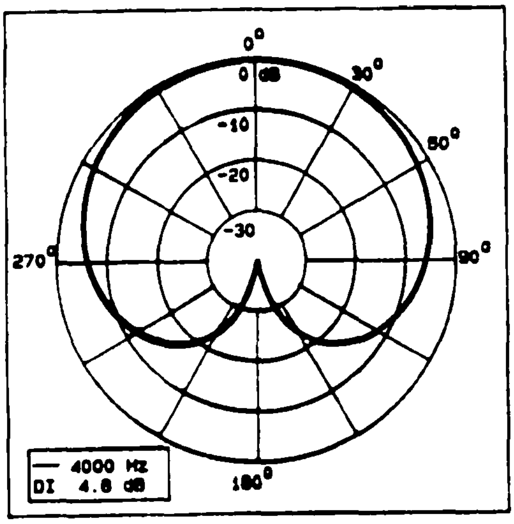
Studies researching the effectiveness of DMHAs in improving speech understanding in noise and reverberation have been carried out since the early 1970s. In general, the results of these investigations have supported the fact that DMHAs improve a person's ability to understand speech in noise as compared to traditional omnidirectional microphone hearing aids (OMHAs). However, several factors can affect a DMHA's performance. These factors include (1) placement of the microphone on the head, (2) speech materials and methodology used to determine performance, (3) test environment (reverberation, noise source configuration, and distance from the sound source(s), and (4) the characteristics of the microphone itself. Without judicious control of these factors, the results of clinical and laboratory assessments of performance can be confusing, misleading, and difficult to compare. In the next few paragraphs these parameters will be discussed in light of the research that has been done and how these factors led to the current investigation.

Placement of the Microphone on the Head

Various electroacoustic measurements of a microphone's directivity are usually done in an anechoic chamber in one of two ways: (1) The hearing aid can be attached to a 2cc coupler and mounted on a stand or (2) attached to a Knowles Electronic Manikin for Acoustic Research (KEMAR) manikin (Burkhard, M. D., & Sachs, R. M., 1975) equipped with a Zwislocki coupler. A graphical representation of the microphone's (hearing aid's) output as a function of angle of incidence is made by rotating the front microphone port of the hearing aid through several azimuths (angles), from 0^0 to 360^0, relative to the sound source. Rotation of the hearing aid can be done in free field (with the hearing aid attached to a 2cc coupler and mounted on a table) or in situ (with the hearing aid sitting on KEMAR's ear (while KEMAR is rotated). At each azimuth, the hearing aid's output level (or attenuation level) relative to that at 0^0 is recorded. This polar plot shows concentric reference lines emanating from the center of the polar plot and graduated in dB. Shapes of polar patterns show how the sensitivity of the microphone varies with the position of the sound source. For example, as seen in Figure 1, with the sound source at 180^0, the output of a cardioid microphone is greatly attenuated. Metrics including the Directivity Factor and the Directivity Index, and the Unidirectional Index are derived from the polar pattern.

Traditional polar measures (hearing aid or microphone in the free field) are not realistic because they do not reflect how the directional microphone will function when actually worn. When the hearing aid is placed on KEMAR, the effects of head and body reflections and diffraction of sound are taken into account. These measurements reveal all hearing aids, when worn, including those with omnidirectional microphones, have directional properties. The open ear has some directional characteristics.

Figure 1: Polar plot of cardiod microphone placed in the free-field (with permission from Etymotic Research, Inc).



People fit with omnidirectional, completely-in-the-canal (CIC) hearing aids also will enjoy a directional advantage thanks to pinna effects (Fortune, 1997). Omnidirectional custom ITE hearing aids also may provide a smaller, but significant directional advantage. However, it should be understood that the directional effects of omnidirectional ITE and CIC fittings, at best, equal those of an open ear canal (Roberts & Schulein, 1997).

It is very important to pay attention to all azimuths of a polar plot on KEMAR because the microphone may be of a type that does or does not provide the desired effect for the listener. In situ polar plots of BTE hearing aids with omnidirectional microphones suggest that this fitting causes a loss of the high frequency natural directivity of the open ear (Ricketts, 2000b). In fact, the in situ polar patterns of some BTE OMHAs may show the greatest sensitivity to sound from behind, rather than in front of the listener (Beck, 1983; Mueller & Hawkins, 1990; Ricketts, 2000b). Due to the influence of head reflections, BTE DMHAs may show the greatest sensitivity to signals arriving from about a 45^0 azimuth (Ricketts et al., 1999). The same is true for ITE directional and omnidirectional hearing aids (Fortune, 1997). This information is important when attempting to evaluate the performance of DMHAs. For example, a listener could be in a restaurant listening situation where noise, rather than the signal of interest, is present at a 45^0 azimuth.

Optimization of the polar pattern of a DMHA in the free field (without consideration of the impact of head and body reflection and diffraction) can result in poor performance. Such was the case for many of the DMHAs used in the 1970s and 80s (Hawkins et al., 1984; Madison et al., 1983). A more satisfactory approach is to quantify the in situ performance of DMHAs by measuring polar patterns on KEMAR. Preves (1976) demonstrated this technique over twenty years ago. More recently, investigators have developed methods of constructing polar plots using real ear measures (Fortune, 1997).

It should be noted that traditional polar plots (such as the one seen in Figure 1) are of the two- dimensional variety. That is, while they describe the directional response of a microphone on KEMAR to sound coming from various they do not take into account the microphone's response to sound coming from various elevations. As such, a 2-D polar plot can be used only as a crude measure of the directional response. Three-dimensional (3-D) polar plots determine directional performance at various azimuth and elevation angles. The test points are distributed over the surface of an imaginary sphere around the reference point of KEMAR. Therefore, a 3-D polar plot describes the directional response of the microphone to the sound, coming under various combinations of elevation and azimuth angles, presenting a complete measure of the microphone's three-dimensional pickup pattern. As with 2-D polar plots, 3-D polar plots can be measured in the free-field or on KEMAR.

An additional factor related to head placement is the microphone port orientation. Microphone port orientation in the horizontal plane can affect directivity. The magnitude of the reduction of directivity is most affected by the proximity of the microphone ports to the pinna, rather than the absolute angle of deviation from optimal. It is important to keep the microphone port orientation within 10% of horizontal to avoid affecting the directivity index (Ricketts, 2000b).

Materials and Methodology Used to Determine Performance

Two different protocols have been used to measure speech recognition performance in noise with directional microphone hearing aids: (1) The use of fixed signal-to-noise ratios (SNRs) and (2) the use of adaptive measures of SNRs. In the first approach, one or more preset SNRs is used to assess speech recognition performance (percent correct) using sentences, words, or syllables (Compton, 1974; Killion et al., 1998; Lentz, 1972; Mueller & Johnson, 1979; Nielsen, 1973; Preves, 1975; Ricketts & Dhar, 1999b; Voss, 1997). Usually the speech is presented at a constant level and the noise level is set to yield pre-determined signal-to-noise ratios. In the second approach, the SNR is varied to obtain threshold (Hawkins et al., 1984; Leeuw & Dreschler, 1991; Lurquin & Rafhay, 1996; Madison et al., 1983; Preves et al., 1999; Ricketts et al., 1999b; Ricketts, 2000a; Soede, Bilsen, & Berkhout, 1993; Valente & Fabry, 1995; Valente et al., 2000; Wouters, Litiere, & van Wieringen, 1999).

The advantage of the fixed SNR method is that it provides clear-cut information about how much speech recognition performance changes in a given test situation with the directional versus the omnidirectional microphone. The downside of this type of approach is that real world SNRs vary greatly, depending upon the environment and situation. In addition, the SNRs typically encountered by a client are not necessarily known by the clinician. Ricketts & Mueller (1999) point out that "if a high (favorable) SNR is selected, the benefit of even the best directional hearing aid will be minimized. Although the greatest benefit will be realized in the most difficult SNR conditions, if this SNR condition is worse than encountered in the real world by the patient, the amount of true directional hearing aid benefit will be overestimated."

Advantages of the adaptive or "up-down" procedures include higher efficiency and greater flexibility (Levitt, 1971). The method obviates the need to select an optimal signal-to-noise ratio for testing. SNR improvement can be related to percentage improvement in speech recognition if the transform for a particular speech material is known.

Test Environment

The test environment can have a major impact on DMHA performance. The environment can be defined in terms of its reverberation time, the distance of the sound source (and/or noise source(s)) from the listener, as well as the number and location of the noise source(s).

DMHA studies can be classified into two main groups: (1) studies in which the speech stimulus is presented from a single, frontally placed loudspeaker (0^0 azimuth) and the competing noise is presented from a single loudspeaker placed behind the listener (180^0 azimuth) and (2) studies in which the speech stimulus is presented from one loudspeaker placed at 0^0 azimuth, and the competing noise is presented from two or more loudspeakers, placed at various azimuths. The former studies can be referred to as *0^0/180^0 studies* while the latter studies can be referred to as *multiple noise source studies*. DMHA studies also can be secondarily classified into one of four groups: (1) Studies that hold the test environment as well as the noise configuration constant, (2) studies that hold the test environment constant and vary the noise source configuration, (3) studies that vary the test environment and hold the noise source configuration constant, and (4) studies that vary both the test environment as well as the noise source configuration. The following discussion will be divided into two sections, one addressing the *0^0/180^0* studies and the other the *multiple noise source studies*. Within each of these categories, the literature will be reviewed with regard to how the test environment and/or noise source(s) was/were varied and what impact this had upon the investigative findings. A separate section will detail the results of studies using distance as an independent variable. Finally, there are a couple of studies that cannot be grouped with the others due to the fact that they used unique speech and/or noise source configurations. As such, these studies will be discussed separately.

0^0/180^0 Studies

Several research studies have employed a *0^0/180^0* set-up (Hawkins et al., 1984; Lentz, 1972; Lurquin et al., 1996; Madison et al., 1983; Mueller et al., 1979; Valente et al., 1995). Although most of these investigations have shown directional microphone benefit, most of them were carried out in unrealistic settings -- in anechoic chambers or test booths having very low reverberation times. Because of this, it is unknown whether data obtained from these studies are indicative of real world benefit. In addition, with the exception of one study (Lentz, 1972) where speech and noise stimuli were presented at fixed levels, these studies used an adaptive procedure to present speech and noise, reporting DMHA performance in terms of dB SNR improvement.

In one of the earliest studies, (Lentz, 1972) CID W-22 words lists were presented at 62 dB SPL with white noise at SNRs of 0 and -6 dB. Testing was carried out in an IAC booth. Results showed a 16% to 24% improvement in monaural speech recognition for SNRs of 0 and -6 dB, respectively with the Maico M-100 cardiod microphone hearing aid.

Madison & Hawkins (1983) used a similar paradigm, but different speech materials. In an attempt to assess the effect of environment on the performance of normal hearing listeners wearing BTE Oticon E24V switchable omnidirectional/cardioid DMHAs, they presented NU-6 Lists at 65 dB SPL with multi-talker babble presented adaptively in two environments -- an anechoic chamber and a room having a reverberation time of 0.6 seconds. A DMHA advantage of 10.7 dB was found for the anechoic environment while a lesser advantage of 3.4 dB was found for the more realistic room environment.

In a follow-up study to evaluate performance with a DMHA (same make and model as used in 1983 study), Hawkins, et al (1984) tested listeners with normal hearing as well as with sensorineural hearing loss. Testing was carried out in three rooms having reverberation times of 0.3, 0.6 and 1.2 seconds, respectively. Subjects were asked to listen to KEMAR recordings of NU-6 lists that were recorded just beyond the critical distance for each room. A 3-4 dB DMHA advantage was seen for the two shorter reverberation times for both groups. For the 1.2-second reverberation time, subjects with normal hearing failed to show a statistically significant DMHA advantage. Data for subjects with hearing loss could not be analyzed since 5 of the 11 subjects could not obtain 50% word recognition in this condition. This finding, demonstrated that the DMHA advantage decreases as reverberation time increases and that the SNR is optimized when binaural DMHAs are used in rooms with short reverberation times.

Valente and his colleagues (1995) and Lurquin & Rafhay (1996) used adaptive procedures within sound attenuated booths to measure the SNR advantage provided to people with sensorineural hearing loss by a BTE DMHA.

While both studies used the $0^0/180^0$ paradigm and evaluated the same hearing aid (Phonak PICs AudioZoom (switchable omnidirectional/cardiod)), different methods were used. In the Valente (1995) study, the Hearing in Noise Test (HINT) was used with HINT noise fixed at 65 dBA . Lurquin & Rafhay (1996) used Fournier bisyllabic word lists along with cocktail noise. However, both studies found a similar SNR advantage of 7-8 dB with the directional microphone mode selected versus the omnidirectional mode. Further inspection of Table 1 below shows that the early studies revealed a DMHA improvement of 1 to 4 dB over the performance with OMHAs. On the other hand, the two studies done in the 1990s showed a SNR improvement of 7 to 8 dB. Because of differences in the speech materials used for evaluation and differences in the amounts of room reverberation in the various studies, it is difficult to make direct comparisons among studies. The directional microphones evaluated in recent studies probably do have increased directivity over the hearing aids used in the older studies, but the newer studies also employed easier speech materials (linguistically redundant) and performance was measured in a room having a low reverberation time. Thus, the results for the older and newer studies may really be more similar than they appear.

Table 1. Summary of $0^0/180^0$ Studies of DMHA Performance

Investigator(s)	Microphone Type(s)	Test Environment	SNR	Stimuli	Percentage or SNR Improvement (in dB) with DMHA(s)
Lentz, 1972	Separate hearing aids (Maico M-100) w/ omnidirectional & cardioid microphones.	Test booth	0; -6	CID-W22 Lists at 62 dB SPL/white noise	16 - 24%
Madison & Hawkins, 1983	BTE: Oticon E24V. Switchable omnidirectional/ cardioid	Anechoic & Rt - 0.6 sec.	Adaptive (50% criterion)	NU-6 Lists at 65 dB SPL/multi-talker babble	1.7 dB/3.4 dB
Hawkins & Yacullo, 1984	BTE: Oticon E24V. Switchable omnidirectional /cardioid	Rt - 0.3, 0.6 & 1.2 sec.	Adaptive (50% criterion)	NU-6 Lists at 65 dB SPL/12-talker babble	3-4 dB for Rt = 0.3 and 0.6; no DMHA advantage for Rt = 1.2 sec.
Valente, Fabry, & Potts, 1995	BTE: Phonak PICs Audio Zoom. Switchable omnidirectional/ cardioid. Unequalized frontal frequency responses.	Test Booth	Adaptive (50% criterion)	HINT/HINT noise at 65 dB SPL (A), spectrally & temporally matched to sentences	7-8 dB
Lurquin & Rafhay, 1996	BTE: Phonak PICs Audio Zoom. Switchable omnidirectional/ cardioid.	Test Booth	Adaptive (50% criterion)	Fournier bisyllabic word lists/cocktail noise	6.6 dB

However, the results from the studies cited in Table 1 should be interpreted with caution since the $0^0/180^0$ arrangement is contrived and engineered to show maximum benefit for cardioid microphones. Furthermore, the listening situations are not typical of real world conditions. Listening is usually not done in an anechoic chamber or in a sound attenuated booth and noise is rarely from a single source from behind. Thus, while measurements in an anechoic room or sound attenuated test booth serve as useful tools in designing DMHAs and in comparing one DMHA to another, they do not necessarily predict performance in the real world where room acoustics can influence performance.

The effect of reverberation on DMHA performance was clearly demonstrated in 1980 in a study that investigated the effect of five environments on the directional characteristics of three head-worn directional hearing aids mounted on KEMAR (Studebaker et al., 1980). Each hearing aid differed in its Front-to-Back Ratio (FBR). For example, hearing aid A's FBRs did not exceed 10 dB, whereas hearing aid B's FBRs were as large as 15 to 20 dB, depending on the frequency at which it was measured. The environments included an anechoic chamber, an audiometric test booth, a living room, a school classroom, and a church classroom. Frequency responses for each of the three hearing aids were obtained with broadband thermal noise presented from a signal loudspeaker as the manikin was rotated at 0, 90, 180, and 270^0. For each hearing aid, the frequency responses obtained at each azimuth became increasingly similar as reverberation time increased. When the reverberation time reached 0.8s or greater, little effective directionality remained. As the reverberation time increases and the reverberant sound field persists longer at a higher level, the reverberant sound incidence on the diaphragm will be more random in direction and level, and the relative polarity for phase of the signals is random and they will not cancel exactly. A truly random incidence sound field may not cancel at all in the microphone.

Multiple Noise Source Studies

Even as early as the 1970s, there was recognition of the importance of using multiple noise sources (e.g. Nielsen, 1973: Compton, 1974; Preves, 1975; Rumoshosky, 1976; Lentz, 1977). However, for whatever reason, the use of multiple noise sources never became popular in clinical evaluations. In addition, as mentioned in the previous section, some researchers continued to use only one noise source, even in the 1990s (Valente, et al, 1995; Lurquin & Rafhay, 1996). Recently, there has been a renewed interest in using two or more noise sources (Soede, 1993; Voss, 1997; Preves, 1999; Ricketts & Dhar, 1999). This is particularly important for two reasons. First, today's directional hearing aids contain not just cardioid microphones, but microphones having varying polar patterns and degrees of directivity. Second, multiple noise sources better simulate real world listening conditions. Like reverberation, the effect of noise source configuration is an important one since real world conversations take place with competing noise sources located at various positions, not always from one location directly behind the listener. While the studies shown in Table 2 may vary in methodology, they all have one thing in common: the investigators presented multiple noise sources in a single configuration. Only these more recent studies will be reviewed.

Table 2: Studies using multiple noise noises, but holding the noise source configuration constant. The test environment is either held constant or varied.

Investigator(s)	Microphone Type(s)	Test Environment	SNR	Stimuli	Speaker Configur-ation for Noise	Percentage and/or SNR Improvement (in dB) with DMHA(s)
Soede, et al., 1993	Broadside and endfire arrays	Anechoic diffuse field only	Adaptive; 50% criterion	Dutch sentences/ uncorrelated "diffuse" speech spectrum noise at 55 & 65 dB SPL	8 speakers in an imaginary "cube"	7 dB
Voss, T., 1997	BTE: Phonak PICS 232 AudioZoom (dual mic omnidirectional/cardioid)	Test booth & "Quasi free sound field"	0, -10, & -15 dB	Monosyllabic words at 65 dB SPL; Correlated, amplitude-modulated speech spectrum shaped babble.	45^0, 135^0, 225^0, 315^0	15% at -10; 29% at -15; no difference at 0 dB
Preves, D, et al., 1999	ITE: MicroTech Personal Choice dual mic omnidirectional/ supercardioid. Unequalized vs equalized frontal frequency responses	Test Booth only	Adaptive; 50% criterion	HINT/ uncorrelated HINT noise at 65 dB SPL (A)	115^0 & 245^0, just before nulls in microphone polar plot.	2.8 dB & 2.4 dB for unequalized and equalized microphone conditions, respectively.
Ricketts, T. & Dhar, S., 1999	BTE: Phonak Piconet AZ (dual mic; omni/hyper-cardioid), Siemens Prisma (dual mic; omni/hyper-cardioid); Widex Senso (separate omni/cardioid hearing aids)	Anechoic Room & living Room (Rt - 0.6 sec.)	Adaptive; 50% criterion	HINT/ uncorrelated, compressed & filtered cafeteria noise at 65 dB SPL (A)	90^0, 135^0, 180^0, 225^0, 270^0	Anechoic: 7.5 dB Living Room: 6.5 dB

In 1999, Preves and his colleagues evaluated binaural performance using ITE hearing aids that could be switched between omnidirectional and directional modes. HINT results using two uncorrelated noise sources indicated significant improvement in speech recognition for the directional versus the omnidirectional mode when the noise was presented at 115^0 and 245^0. Results showed HINT SNR improvement of 2.8 and 2.4 dB for unequalized and equalized microphone conditions. It is interesting to compare these results to those obtained by Valente, Fabry, & Potts (1995) who also used the same speech and noise materials (HINT) and a microphone of less directivity (cardiod), yet found a 6 to 8 dB improvement in the SNR (see Table 1). Upon first glance, one might conclude that the supercardioid microphone used in the study by Preves and colleagues should produce an even better result than that seen with the cardioid microphone used in the Valente study. However, the results can be explained by the fact that the Valente study used a single noise source from behind, thus favoring the polar response of the cardioid microphone. The Preves study attempted to approximate a more acoustically realistic setting by using uncorrelated noise sources that were positioned so that they were not aimed toward the nulls of the hearing aid microphones. The remaining studies shown in Table 2 employed four or more noise sources presented from the front, sides and back to create a more realistic, "diffuse" sound field.

In 1993, (Soede et al., 1993b) measured the performance of newly developed, highly directional broadside and endfire array microphones. Physical measurements were performed with KEMAR in a diffuse noise field created by arranging eight loudspeakers in an imaginary cube inside an anechoic chamber. KEMAR was placed in the center of the chamber, facing the front loudspeaker at a distance of 1 meter. Hearing aids having an omnidirectional, cardiod or array type microphone were affixed to KEMAR's right ear with a foam earplug. The measured attenuation of the diffuse noise field relative to the noise coming from the front speaker was recorded as a function of frequency in one-third octave bands (400-5000Hz). The mean level was computed from the one-third octave bands with equal weights. KEMAR recordings showed that an omnidirectional microphone amplified the diffuse sound field by 1 dB relative to a signal coming from the front. The hearing aid with one cardioid microphone attenuated the diffuse sound field for the lower frequencies with a mean of +1.5 dB. This translates to measured SNR improvement of +2.5 compared with the omnidirectional hearing aid. In comparison, each array microphone produced a SNR that improved by approximately 7 dB.

Behavioral measures also were obtained in the same diffuse noise field. Dutch sentences were presented adaptively amidst uncorrelated speech spectrum noise presented at two levels (55 and 65 dB SPL). Monaural listening with the array microphones produced a mean improvement in the SNR of 7 dB. Binaural listening with two endfire microphones produced improvement comparable to binaural improvement obtained by listening with two normal ears (1.7 dB) or two omnidirectional hearing aids (2.5 dB)

Sound-field testing was also carried out in a church hall. In this situation, a loudspeaker was placed in the front of the church and fed with a long-term speech spectrum noise. The SPL of the noise was measured as a function of the distance from the loudspeaker with a sound level meter and a 1-inch omnidirectional microphone. The same measurement was then carried out with two types of array microphones, keeping the main beam of the microphones in the direction of the loudspeaker. The microphone arrays doubled the effective reverberation distance of the hall. That is, the diffuse sound field was attenuated by 6 dB thus doubling the effective critical distance. The results of this study are quite encouraging because a 7 dB improvement in SNR performance in a cocktail party-like situation and a doubling of the critical distance in a church hall means that, with a monaurally-fitted array microphone, a hearing impaired person can potentially function about as well as a normal hearing person in the same acoustic conditions. In other words, to achieve 50% criterion on a Dutch sentence test, normal hearers and hearing impaired users of an array microphone require a SNR of around -8 dB whereas -1 dB is required for hearing impaired users of omnidirectional hearing aids. However, there is one problem with this study. Although a perfectly diffuse noise field may be easier to standardize than many real world environments, there is no evidence that directional benefit obtained in such an environment will correlate with that measured in real-world environments. A perfectly diffuse listening field can only be achieved in either a highly reverberant, irregularly shaped room (Beranek, 1954) or by using multiple speakers in an anechoic chamber (Veit, 2000). Hence, perfectly diffuse environments are rare in the real world and difficult to achieve in a clinical test environment due to space and financial limitations. In addition, highly diffuse environments generated via the use of high reverberation levels will severely diminish the directivity of the hearing aid being measured, thus preventing any useful comparisons of directional microphone hearing aid benefit (Ricketts et al., 1999).

Voss (1997) compared the performance of a dual microphone hearing aid set to cardioid versus omnidirectional mode. In this study, monosyllabic words were presented amidst correlated, amplitude-modulated speech spectrum shaped babble presented at SNRs of 0, -10 and -15 dB from four loudspeakers located at 45^0, 135^0, 225^0, and 315^0. Testing was carried out in two environments, a test booth and a "quasi free" sound field.[1] Thirteen adults (ages 29 to 82 years, median age of 58 years) with bilateral, mild to moderately-severe sensorineural hearing loss participated in this study. Each subject was an experienced hearing aid user. With the directional microphone setting, results showed speech recognition improvement of 15% at a SNR of -10 dB and 29% at a SNR of -15 dB -- or a SNR improvement of 1.5 and 3 dB, respectively. The 0 dB SNR test condition did not show improvement with the directional microphone setting (94%) as compared to the omnidirectional setting (92%). This study is interesting because it produced about one-half or less of the SNR improvement shown by Lurquin & Rafhay (1996) and Valente, Fabry, & Potts (1995) who tested the same model of hearing aid. However, it is important to note that the latter studies used only a single 180^0 sound source inside a test booth whereas the Voss (1997) study used a more realistic sound field that did not favor the

[1]According to the author, "the set up met the requirements of ISO 8253-2." That is, "the test room's reflecting surfaces had a moderate influence on the sound field."

cardioid pattern of the hearing aid. Another interesting finding is concerns the performance improvement observed with the directional microphone at SNRs of -10 and -15 dB, but not at 0dB. According to data reported by Pearsons, Bennett, and Fidell (1977), the average SNRs encountered in various face-to-face (speaker and listener separated by 1 meter) conversational situations (classrooms, homes, hospitals, departments stores) range between +4 and +21 dB. The least favorable SNRs were reported to occur in trains (-8 dB) and aircraft (-11 dB). Thus, while the improvement seen with the DMHA at -10 and -15 dB SNR is impressive, it is doubtful that listeners will encounter such adverse listening situations on a daily basis. The study's results would have been more impressive had improvement been seen at 0 dB SNR since that SNR will more likely be encountered on a daily basis.

In 1999, Ricketts and Dhar measured binaural HINT sentence performance of listeners with sensorineural hearing loss across the same three brands of BTE OMHAs and DMHAs in two different environments. Two of the hearing aids were of the switchable, dual microphone type (omnidirectional versus cardioid modes; Siemens Prisma and Phonak Piconet P2AZ). The third condition was tested using two separate omnidirectional and directional (cardioid) hearing aids (Widex Senso). In an effort to simulate a more realistic test environment, a unique type of competing noise was substituted for the commercially available HINT noise. Five sources of uncorrelated and compressed cafeteria noise that was filtered to match the long-term spectrum of the HINT sentences were used. The noise was fixed at 65 dBA. Speech recognition performance was also measured using the CUNY Nonsense Syllable Test (NST) (Resnick, Dubno, Hoffnung, & Levitt, 1975) presented amidst multi-talker babble at a +8 dB SNR. In both conditions, the noise was presented at 90^0, 135^0, 180^0, 225^0, and 270^0 in both an anechoic room and in a living room have a reverberation time of approximately 0.6 seconds. Results in the reverberant environment were 2-3 dB poorer than reported for the same hearing aids that were tested by Valente, et al in 1995. These differences might reflect differences in test environment (the Valente, 1995 study was done in a sound booth where reverberation time was not actually measured, but was assumed to be between 0.1 and 0.6 seconds). The differences also could be due to the fact that an uncorrelated diffuse sound field was used by Ricketts and Dhar, while a single loudspeaker from behind was used by Valente and colleagues.

Recently, some studies have begun to compare DMHA performance across various multiple noise source configurations - in a single or multiple test environments (Table 3).

A group of investigators at two sites (Valente et al., 2000) measured binaural performance on the HINT (HINT noise at 65 dB SPLA using dual microphone (omnidirectional versus hypercardioid) ITE hearing aids. Subjects with mild to severe sensorineural hearing loss were tested in a test booth with two noise configurations. The first was the "ideal laboratory condition", a

Table 2: Studies using multiple noise noises in which the configuration is varied (and the environment is either held constant or varied).

Investigator(s)	Test Environment	SNR	Stimuli	Speaker Configuration for Noise	Percentage and/or SNR Improvement (in dB) with DMHA(s)
Killion, et al., 1998	Various real world locations & test booth	0, +5, +10, & +15	SIN Test/Real world noise (unstandardized) & speech-enveloped modulated speech spectrum noise (test booth)	Real world (unstandardized) & IAC booth at 45^0, 135^0, 225^0, 315^0	4-5 dB
Ricketts, 2000a	Living Room (Rt = 0.6 sec; classroom (Rt = 1.1 sec.)	Adaptive; 50% criterion	HINT at 65 dB SPL with uncorrelated amplitude-modulated, filtered, & normalized cafeteria noise.	1. 0^0, 180^0 2. 90^0, 135^0, 180^0, 225^0, 270^0 3. 30^0, 105^0, 80^0, 225^0, 330^0 4. 30^0, 105^0, 180^0, 225^0, 330^0, 30^0, & 330^0 turned per-pendicular to listener	Living Room: 3.6 to 7.9 dB Classroom: 2 to 5.1 dB (depending on configuration of noise sources).
Valente, et al., 2000	Test Booth	Adaptive; 50% criterion	HINT at 65 dB SPL with correlated noise.	1. 180^0 2. 45^0, 135^0, 225^0, 315^0	Average of 3 to 4 dB

traditional *$0^0/180^0$* approach in which the noise was presented from a single loudspeaker located at 180^0 behind the listener. The second condition consisted of a multi-speaker sound field using correlated noise presented at four azimuths: 45^0, 135^0, 225^0, and 315^0. With the hearing aids in the directional mode, results revealed a SNR improvement of between 3.7 to 3.5 dB at Site 1 and between 3.2 and 2.7 dB at Site 2 for ideal and multi-speaker conditions, respectively. What is interesting is that while the average performance in the classic *$0^0/180^0$* condition was statistically significantly better (0.5 dB) than that obtained in the multi-speaker condition, the result was judged to be trivial and thus not clinically significant. As the authors explained, this small difference could have been be due to the fact that the noise used in the multi-speaker condition was correlated. Thus the improvement expected in a multi-speaker setting was not seen due to the absence of interaural time, intensity, and/or phase cues normally seen with uncorrelated noise sources.

In 2000, Ricketts investigated the impact of position of noise source(s) and reverberation on the directional benefit and performance of the same three commercially available directional hearing aids used in his earlier study (Ricketts & Dhar, 1999). The HINT test (with the same modified HINT noise used in the previous study) was used to evaluate binaural (a) directional performance (absolute scores) and (b) directional benefit (difference between HINT threshold SNRs for OMHA versus DMHA conditions for the same hearing aid brand) of participants with hearing impairment in two listening environments: (1) a "living room" having a reverberation time of 0.6 seconds and (2) a "classroom" environment having a reverberation time of 1.1 seconds. These environments were chosen to be representative of the reverberation present in average living rooms and classrooms as defined by Moncur and Dirks (1967) and Nabalek and Mason (1981) In addition to investigating DMHA performance in realistic listening environments, it was the first study to clinically investigate the effect of four different noise source configurations in each environment: (1) 0^0 and 180^0, (2) 5/B: 90^0,135^0, 180^0· 225^0, and 270^0, (3) 5/S: 30^0, 105^0, 180^0 , 225^0, and 330^0, (4) m5/S: 30^0, 105^0, 180^0, 225^0, and 330^0 (with the 30^0 and 330^0 loudspeakers turned perpendicular to the listener).

The first configuration was considered to be the traditionally ideal (yet contrived) laboratory set-up for directional microphone hearing aids. The second configuration consisted of noise sources concentrated behind and to the sides as might be experienced when listening in front of a class or in the theater. The third configuration was designed to imitate a meeting or a restaurant in which the competing noise is more diffuse. The fourth configuration was designed to reduce the output of the front speakers so that these noise sources might seem farther away, such as in a restaurant where the listener has positioned himself or herself so that the competing noise is mostly from behind and to the sides. A byproduct of this configuration is that it decreases the impact of competing noise in the front hemisphere for DMHAs with a large pickup "lobe" (the greatest sensitivity) in the 20 to 40^0 range. All sound sources were placed at a distance of 1.25 meters from the listener across all eight listening environments (four noise source arrangements X two reverberant environments).

This study yielded several interesting and important findings relevant to the present study. First, directional benefit across hearing aids tested in the living room environment ranged between 3.6 to 7.9 dB, depending upon noise source configuration. This benefit decreased to a range of 2 to 5.1 dB in the classroom setting. Directional benefit was significantly higher for the $0^0/180^0$ speaker configuration in comparison with all others and significantly less directional benefit was provided to listeners for the diffuse restaurant configuration (5/S) than the classroom (5/B) or restaurant configuration (m5/S) where the background noise at 30^0 and 330^0 is reduced 5 dB. Directional benefit also was significantly higher for the $0^0/180^0$ condition in the living room environment, whereas no other significant differences were seen for the same loudspeaker arrangement across the two environments. These results point out the fact that multiple noise sources do a better job of emulating real-world environments than does the $0^0/180^0$ test configuration. Second, an inverse relationship was noted between directional benefit/performance and reverberation time across all hearing aid brands. Although smaller in magnitude, this trend is in agreement with previous investigations (Hawkins & Yacullo, 1984; Madison & Hawkins, 1983; Studebaker et al., 1980). Specifically, the current study showed that increasing average reverberation time from 0.6 to 1 second resulted in a reduction of directional benefit of 0.93 dB (m5/S configuration) to 2.7 dB ($0^0/180^0$ configuration). However, Hawkins & Yacullo (1984) found directional benefit to decrease approximately 9 dB as reverberation was increased by a similar amount (0.6s to 1.2s). While these studies cannot be compared directly due to differences in test materials and methodologies, the relatively small impact of reverberation on the measured directional benefit in Ricketts (2000) investigation may be partially explained by the interaction between source distance and reverberation. The Hawkins & Yacullo (1984) study used distances of 2.2 to 3.3 meters between the sound (speech) source (just beyond the critical distance) and the listener whereas Ricketts (2000) used a fixed distance that was smaller (1.5 meters) (within the direct field). The reduction in distance greatly improves the ratio of direct to reflected sound energy, thus decreasing the negative impact of reverberation on directivity. According to

Ricketts (2000), a distance of 1.5 meters (4.9 feet) is more realistic, supported by data collected by Pearsons, Bennett, & Fidell (1970) who reported that people listening in "home measurement situations" voluntarily selected a speaker-to-listener distance of approximately 1 meter. For the present study, an even closer speaker-to-listener distance of 0.61 meters (24 inches) will be employed. It is felt that this shorter speaker-to-listener distance is more realistic for a one-to-one communication situation in a noisy restaurant than would be the distance of 1.5 meters.

In an approach designed to provide increased face validity, some researchers (Killion et al., 1998) have made binaural head-worn (using real people) recordings of prototypical ITE hearing aids equipped with both omnidirectional and directional (D-MIC™) microphones. Several pairs of ITE hearing aids were equipped with D-MIC™ cartridges and sub-miniature Microtronic four-pin connectors. One pin was connected to the omnidirectional microphone output and one to the directional microphone output. The directional microphone output was equalized to produce the same frequency response (flat) as the omnidirectional microphone. Cables were connected to permit each of the two stereo microphone outputs--directional and omnidirectional--to be connected to a hand-held digital analog tape (DAT) recorder. The individual, acting as a "recording dummy," wore two custom ITE hearing aids attached to the recording instrumentation described above. Each DAT recorder was carried in a small belt pack. Outputs of the omnidirectional and directional microphones were recorded simultaneously, thereby permitting later comparison of the two microphone outputs under identical conditions. A modification of the Speech in Noise (SIN) Test was used to make the recordings.

Each talker was provided with a small boom microphone connected to a sound level meter (SLM). This arrangement allowed the talker to monitor his or her own voice level while reading the sentences. For each block of sentences, the talker established a "suitable level for the highest vocal effort", producing an estimated +10 dB to +15 dB SNR according to a precision SLM held near the listener's ear. While the talker maintained that effort, the mouth-to-boom-mic distance was adjusted until the monitor SLM read peaks of 85 dBA (FAST scale). The talker then read the first five of the SIN sentences while attempting to produce frequent peaks of 85 dB on the SLM. Three more sets of five sentences were presented using vocal efforts of 80, 75, and 70 dB A. This procedure resulted in a sequence of sentence blocks modeled after the SIN (Speech in Noise) Test (Killion and Villchur, 1993; Fikret-Pasa, 1993). The omnidirectional recordings presented speech and noise at SNRs of +10, +5, 0, and -5 dB while the directional microphone recordings presented SNRs of +15, +10, +5, and 0 dB. Recordings were made in various noisy real-world environments: a crowded street party (90-95 dBA), two restaurants (70-80 dBA and 60-65 dBA) and museum party and a classroom party simulation (80-85 dBA).

Subjects with normal hearing and sensorineural hearing loss listened to the recordings under insert earphones using presentation levels of 70 dB HL and "loud but OK" (Hawkins, D. B., Walden, B.E., Montgomery, A. & Prosek, R. A., 1987), respectively. In general, indoor results showed that the directional microphone provided improvement in the SNR threshold for 50% correct consistent with AI-DI expectations (4.4 dB).

Because the methodology of the experimental design was not standardized, it is difficult to compare the study's results with past and future investigations of other directional hearing aids. However, this study is important for several reasons. First, it points out the need for a test environment that approximates real-world reverberation and noise conditions and is easily replicable in the laboratory. Second, it showed that the theoretical AI-DI values of the hearing aids (0.3 dB for the omnidirectional microphone and 4.7 for the directional microphone) could be used to predict improvement in the effective SNR for the SIN test when measured indoors. Third, the noise levels measured in the real restaurants are similar to those used in past laboratory studies. And, one of the restaurants used in the study is the exact same restaurant that is being used in the present study. Knowing the level of the ambient noise in the restaurants is helpful since this information is not available directly from the Pearsons, et al. (1977) study. Finally, results measured outdoors (street party recordings) showed that individuals with hearing loss could understand better (9 dB improvement) with the DMHAs than when they listened to the DMHAs recordings made in the restaurants and museum. This is to be expected because the street party situation is a free field situation where the listener is in the direct sound path of the primary talker. In the other listening environments, the listener has to contend with room reverberation. The effect of room reverberation is related to the next factor affecting DMHA performance -

distance. This factor will be discussed following the discussion of an additional study.

A Unique Study

Another study is worthy of discussion. It cannot be grouped with the former investigations due to the fact that the investigators used a noise source configuration different from the other studies.

In an attempt to develop a method of evaluating DMHAs in the clinic without establishing a bias for the DMHA, Mueller and Sweetow (1978) suggested presenting the competing signal from a single speaker located at a 90^0 overhead location. The authors explained that this location was cost-effective because it did not require the use of several loudspeakers to produce a diffuse sound field. In 1979 Mueller and Johnson (Mueller et al., 1979) employed this method in an experiment to evaluate the effects of varying the front-to-back ratio (FBR) of a DMHA on speech perception for a group of listeners with sensorineural hearing loss. Each of the 24 subjects was tested with four DMHAs that differed only in their FBR (The FBR at 1000Hz ranged from 6 dB for hearing aid 1 to 20 dB for hearing aid 4). All experimental testing was conducted in a double-walled test suite. Sentences from the Synthetic Sentence Identification (SSI) test were presented at 55 dB SPL from a loudspeaker located 6 feet from the subject at a 0^0 azimuth. The competing message (single-talker discourse) was presented directly overhead, approximately 2.5 feet from the ear of the listener, at three SNRs (0 dB, -10 dB, and –20 dB). Results showed the following: As the FBR (directivity) of the hearing aid increased, a systematic improvement of 8 to 12 percent was seen for all three aided conditions. Again, while an advantage was seen for the DMHAs, testing was done in an audiometric test booth. Since this environment is relatively non-reverberant, it does not say much about how a DMHA will perform in a more real world environment. However, this study was unique in that it has been the only one to employ an overhead speaker location in an attempt to create a more diffuse sound field that does not necessarily favor the cardioid microphone.

Distance from the Sound Source

The impact of reverberation upon directional benefit is also dependent on the distances of the sound source of interest (e.g. the talker) and the competing stimuli (e.g. room noise) relative to the listener (Ricketts and Mueller, 1999). The reason for this is that the proportion of reflected to direct sound energy has been shown to increase in reverberant environments as the distance from the sound source is increased (Ballou, 1991; Peutz, 1971). This proportion will continue to increase up to a "critical distance", the location in the room where the proportion of direct and reflected energy is the same. At distances beyond the critical distance, the reflected sound energy begins to dominate the total sound energy around the listener's head. When the listener is beyond the critical distance (e.g. in a very large room like a lecture hall), the impact of reverberation on microphone directivity will be the greatest (Studebaker, et al, 1980). However, when the listener is close to the talker (within the near-field), the amount of direct to reflected sound energy is increased, decreasing the negative impact of reverberation on directivity. As such, the signal-to-noise advantage provided by directional microphones will be most evident in situations where the talker and the listener are close to each other. An example of this would be a face-to-face conversation across a small table in a restaurant. So, a more accurate way to interpret the results of Studebaker and his colleagues is to understand that it was not that the directional microphone hearing aids lacked innate directivity, but that their effective directivity was adversely effected by presentation of the test signal beyond the critical distance, causing it to became "buried" within the reflective field. Had the experiment been repeated at a close distance, the test signal would not have been a source of reverberant masking and the directional microphones would have performed more satisfactorily. Nevertheless, the main point is that multiple reflections do impact on the ability of directional microphones to improve the SNR.

RATIONALE FOR THE CURRENT INVESTIGATION

Several conclusions can be made from the literature review. First, the majority of studies evaluating DMHAs have used artificiaily contrived settings. That is, they have been carried out in environments having little or no reverberation and with the speech stimulus being presented from the front and the noise from the back, thus favoring the polar response of many directional microphones. Second, due to differences in test materials and noise source configurations, it is rather difficult to compare and contrast results across studies.

Nevertheless, all studies have shown that directional microphones provide better speech recognition performance than omnidirectional microphones for speech in noise. Finally, recent investigations (Preves, et al., 1999; Ricketts & Dhar, 1999; Ricketts, 2000a; Valente, M., et al, 2000) do appear to be converging on the use of a single test material (the commercially available HINT test or at least the use of HINT sentences with the investigator's own competing noise) along with multiple uncorrelated noise sources (except for Valente, et al, 2000 who used correlated noise). And at least two of these recent studies (Ricketts & Dhar, 1999; Ricketts, 2000a) have collected data in more than one environment in an attempt to gain a better idea of how well DMHAs work in real world environments. However, what continues to be needed is a convenient clinical method for predicting how DMHAs will perform in the real world. This method should allow the investigator to conveniently measure the performance of directional and omnidirectional hearing aids in an environment capable of simulating a variety of listening situations, thus avoiding the need to perform testing in more than one room.

The current investigation focused on comparing speech recognition performance amidst background noise when the talker and the listener are in close proximity to each other - e.g. an intimate restaurant setting. A method of reproducing and simulating the environment (called the "R-SPACE®") was developed to reproduce/simulate the sound field found in a typical restaurant. Consideration was devoted to deciding whether the sound field acoustics of various restaurants should be sampled before the selection of one restaurant was made. Instead, one particular restaurant was selected as the recording venue for two reasons: (1) its large size and vaulted ceiling produces acoustics thought to sound typical of a busy, neighborhood restaurant and (2) its management agreed to allow recordings to be made on the premises. Noise measurements made on several different occasions revealed the average SNR (C-scale) to be approximately +5 to +10 dB. Subjectively, the nature of the noise was judged to be rather diffuse -- i.e., it was difficult to single out any one particular person's speech over another's.

If several people are having dinner together, usually only one of the diners is directly across from the listener while the others are located at varying azimuths in relation to the listener. For this investigation, we decided to evaluate the listener's (each subject's) performance with speech being presented at a 0^0 azimuth (the location of one of the dining companions) and noise (of other dining companions) at various azimuths. Presentation of the primary signal at a 0^0 azimuth also makes it possible to compare the results of this study with those of the majority of previous studies. (Later studies may focus on the effect of speech being presented at different azimuths.) In two of the conditions, uncorrelated noise stimuli were presented at several azimuths (45^0, 90^0, 135^0, 180^0, 225^0, 270^0, 315^0 in the horizontal plane since this combination has been used by other researchers (Soede et al, 1993b) and is known to produce a diffuse noise field of the type that would be found in a restaurant where several people are talking simultaneously. This study concentrated on situations where DMHAs are helpful as compared to OMHAs so that a contrast between OMHAs and DMHAs could be made. This means that speech was always presented at a 0^0 azimuth to the microphone (as worn on the head) and noise was presented as various azimuths (except for 0^0). The close-range conversation-in-noise situation was selected for evaluation first because it was reasoned that it should provide a good test of the usefulness of the recording and evaluation techniques to differentiate among hearing aids varying in amounts of directionality. It was also felt that it would be possible to demonstrate a clear advantage between an omnidirectional and directional hearing aid, and one would expect to differentiate between highly directional and less directional hearing aids.

METHODOLOGICAL ISSUES

Rationale for Stimulus Recordings and Instrumentation

A goal of the current study was to determine the amount of agreement among different methods of evaluating the effect of hearing aid directionality and a measure of performance in the "real world." A specific real-world situation was recorded and served as the standard for comparison: a restaurant setting with the listener and the talker at close range (within the critical distance). Since an additional goal of the investigation was to reproduce the sound field of a real restaurant within the R-SPACE simulator, a method had to be developed that allowed participants to compare the sound field of the real restaurant with that reproduced in the simulator. Logic would dictate that the most meaningful way of accomplishing this would be to ask listeners to compare the original background sounds in the restaurant to the same sounds reproduced within the R-SPACE.

Although this method has been used in the design and demonstration of loudspeakers (Villchur, 1964), it was impractical to carry out for the following reasons: First, the simulator would have had to be set up in the restaurant. Each subject would have been required to listen to the background noise in the restaurant, stand up and move into the simulator, and then listen again to the played back recording of the restaurant (while everyone in the restaurant was asked to keep quiet). Not only would this have required an inordinate amount of time to accomplish, but also restaurant patrons would probably have become so engrossed in the process that the natural background noise would have ceased or changed in some fashion - or everyone would have lost patience and vacated the restaurant.

Beginning with Villchur in 1962, many investigators have employed the use of tape recordings for judging the quality of sound systems. For example, a stereo recording of live source material can be made and subsequently played backed through several pairs of loudspeakers having a range of fidelity ratings. Those recordings can then be re-recorded (dubbed) and played back under earphones while a listener is asked to judge the fidelity of each recording. Thus, fidelity ratings of each pair of loudspeakers can be made in one sitting without the listener having to move from one loudspeaker set-up to another.

A requirement for the present study was a well-controlled, efficient design that permitted measurement of each participant's performance as a function of listening through hearing aids having varying amounts of directivity in four restaurant environments: (1) A real restaurant environment; (2) a simulated version of the real restaurant; (3) the real restaurant as reproduced in an IAC booth using a single loudspeaker located behind (180^0) the listener as the restaurant noise source; and (4) the real restaurant as reproduced in an IAC booth using a single speaker at 90^0 overhead as the restaurant noise source. The first two conditions were included to provide information regarding how well the first condition could be simulated in the second. Both of these conditions presented competing noise that was uncorrelated and delivered from multiple azimuths around the listener. The third condition was included because, although it is contrived and favors cardioids, it has been a common clinical method of measuring their performance. Although it has been reported on only once in the literature, the fourth condition was included in order to determine whether a single noise source placed overhead is a valid and cost-effective means of simulating a diffuse, realistic sound field.

Several binaural KEMAR tape recordings were developed and used to compare speech recognition amidst noise produced in the real restaurant with that played back in the simulator and in the two IAC booth conditions under three binaural hearing aid microphone conditions: (1) ITE omnidirectional; (2) ITE supercardioid microphone (D-Mic™); (3) an endfire array microphone (Soede, W., Berkhout, A.J., & Bilsen, 1993a). These three microphones were selected because they represent a broad range of directivity, their AI-DI values being approximately 0.3, 4.7 and 7 respectively. That is, each microphone's polar pattern was judged to be different enough to reflect the acoustic differences in each environmental condition.

To create the primary speech signal for all four environments, recordings of commercially available HINT (Nilsson, et al., 1994) were played from a single loudspeaker placed in front of KEMAR (0^0) in the IAC booth. To create the competing noise stimulus, simultaneous binaural recordings were made through the three types of hearing aid microphones fitted to KEMAR in each of the four environments. Test tokens were created by taking the binaural speech recordings made through each set of hearing aids microphones in the IAC booth and presenting them along with the binaural noise recordings made through each set of hearing aid microphones in each of the four environments. For example, the HINT sentences recorded through the array microphones in each IAC booth were mixed with the restaurant noise recorded through the array microphones in all four environments. This method was chosen over recording the sentences simultaneously in each environment at the various SNRs since the former method provided more control over the stimulus parameters. Attempting to record in each environment would have been particularly time-consuming and would have precluded the use of an adaptive test procedure as recordings at several SNRs would have been required. In addition, it would have been difficult to control for room noise and external noise from motor vehicle and air traffic.

The use of KEMAR recordings controlled for the variable of head movement and the use of binaural recordings made directly through three different hearing aid microphones avoided the influence of additional hearing aid circuitry such as signal processing and frequency shaping. Instead, the only variable was microphone directivity.[2] A pilot study was carried out to test the method. Results indicated that KEMAR recordings are an acceptable way of comparing the microphones and test environments. Details of the pilot study may be found in Appendix A.

Subjects

Twelve subjects with normal hearing served as listeners for this study after a pilot study showed that subjects with mild to moderate mid to high frequency sloping sensorineural hearing loss could not detect the noise stimulus when listening to the high-pass frequency response of the array microphone recordings (see Appendix A for details).

Stimuli

Listeners were presented with binaural KEMAR recordings of the HINT sentences and restaurant noise recorded through omnidirectional ITE, D-MIC™, and array hearing aid microphones fitted simultaneously to KEMAR. The binaural target-speech recordings were made by playing the sentences through a single Tannoy Arena loudspeaker placed at a 0^0 azimuth at a distance of two feet from the front of KEMAR in an IAC booth. The binaural KEMAR noise recordings were made under four test conditions: (1) in the actual restaurant, (2) in the R-SPACE (located in an acoustically treated recording room at Etymotic Research, (3) in an IAC test booth (located at Northwestern University) with the noise being presented from a single loudspeaker located 180^0 behind KEMAR, and (4) in the

[2] It was discovered later that the array microphone which had more of a high-pass frontal frequency response. This will be discussed later.

same IAC test booth with the noise being presented from a single loudspeaker located overhead.

Twelve 20-sentence long HINT sentence lists were presented to each participant. Each list was matched with a restaurant noise sample (henceforth referred to as a noise "segment") to produce a test token that represented one of 12 microphone/test environment listening conditions. For each of the 12 conditions, a unique combination of one HINT sentence list and one noise segment was presented to each subject. This design was used to spread the error variance possibly created by using unfiltered and uncompressed natural restaurant noise along with the sentences. Presentation of the 12 conditions was randomized for each subject.

The HINT procedure was chosen for several reasons. First, it was specifically designed for measuring the SRT in noise for linguistically redundant (and therefore realistic) speech materials. Second, it has been used in many other studies of directional microphone hearing aids. Third, the HINT materials have been digitally recorded for standardized presentation. Finally, a sufficient number of lists were available for the design of this study.

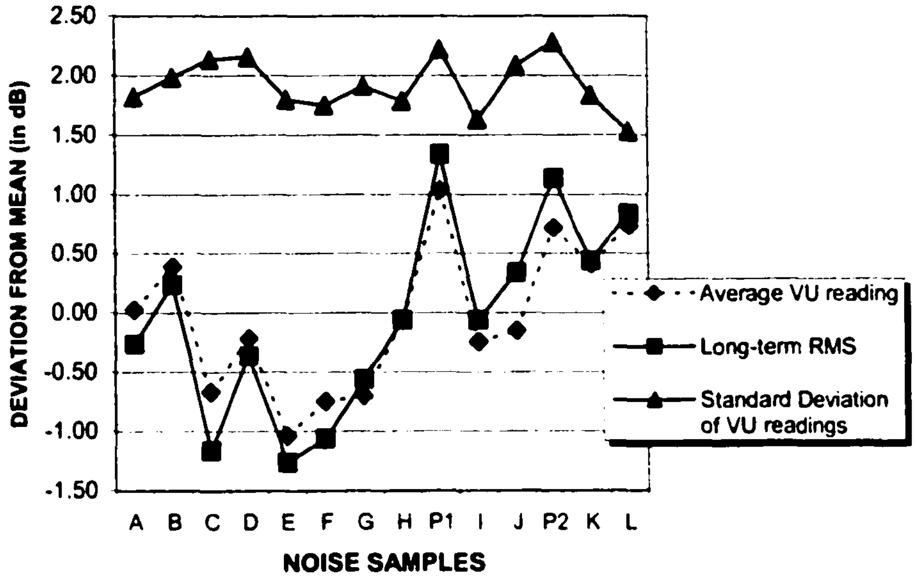
The HINT (Nilsson, Soli, & Sullivan, 1994) was developed as a tool for measuring speech intelligibility in noise with and without spatial separation from the speech source. The test consists of 250 sentences (25 lists of 10 sentences each) read by a male talker. The sentences are of approximately equal length (six to eight syllables) and difficulty (first-grade reading level). The listener's task is to repeat sentences spoken by a male talker in the presence of speech-shaped noise presented at a fixed level of 65 dB SPL. An adaptive procedure is used to estimate the SNR at which the sentences, presented in background noise, can be repeated correctly 50 percent of the time. Correct identification of each sentence is based on proper repetition of all words of each sentence, with the exception of certain articles where substitution is allowed (e.g. "a" for "the"). Scoring is done for either one or two 10-sentence blocks. The presentation level of the sentences is adaptively adjusted depending on the listener's response. An incorrect response raises the speech presentation level and a correct response lowers the speech presentation level for the next trial. The level of the sentence stimuli is varied in 4-dB steps for the first five trials and in 2-dB steps for all remaining trials. The test is meant to be used with a speech-shaped noise whose spectrum matches that of the talker (supplied with the HINT recordings). In this study, brief segments of noise recorded in the real restaurant were substituted. The use of such noise was determined to be suitable in a pilot study described later.

Normative data on the HINT test are available for listeners with normal hearing as well as sensorineural hearing loss (Nilsson, Gellnet, & Sullivan, 1992; Nilsson, Soli, & Surmida, 1994). In 1999 (b), Ricketts & Dhar used a modified HINT test. They replaced the standard speech spectrum-shaped noise with amplitude compressed cafeteria noise. The peak-to-valley ratios observed with a 10 msec time window did not exceed 7 dB after amplitude compression. The noise sources were then filtered to provide a long-term average spectral shape identical to that of the test stimuli. The modified HINT (one speech stimulus and five competing noise tracks) was presented from compact disk via three CD players and three stereo amplifiers. Two 10-sentence blocks were used for each test condition. A 95% confidence interval of 1.86 dB was obtained for this modified HINT test. This was found to be in fair agreement with the 1.20 to 1.51 dB (depending on noise azimuth) reported for the unmodified HINT (Nilsson et al., 1992).

For the present investigation, the HINT sentences were re-recorded in a quiet room (IAC booth) in order to control for variability across environmental conditions. Although it would have been more realistic to record the sentences simultaneously with the noise in the restaurant, it would have been very difficult to control extraneous noise and the natural ebb and flow of restaurant noise. In addition, the recording of several SNRs would have been too time-consuming. Since a goal of the current study was develop a procedure that will predict the "real world" performance of hearing aids, we felt it important to use actual, unmodified, restaurant noise in place of the commercially available HINT noise.

Prior to collecting pilot data, a study was done to compare HINT reliability with the commercially available HINT noise versus the restaurant noise recorded in the current investigation. First, the recording of the restaurant noise was divided into 14 approximately 2.5-minute segments. Using SoundForge 4.5, the long-term rms of the restaurant noise was measured for each segment. An average VU reading also was calculated for each segment by using a metronome and noting the peak VU reading every 0.75 seconds. For complete details of the sampling procedure, see the calibration section. Figure 2 below

Figure 2. Comparison of VU sampling versus overall rms calculation for restaurant noise.

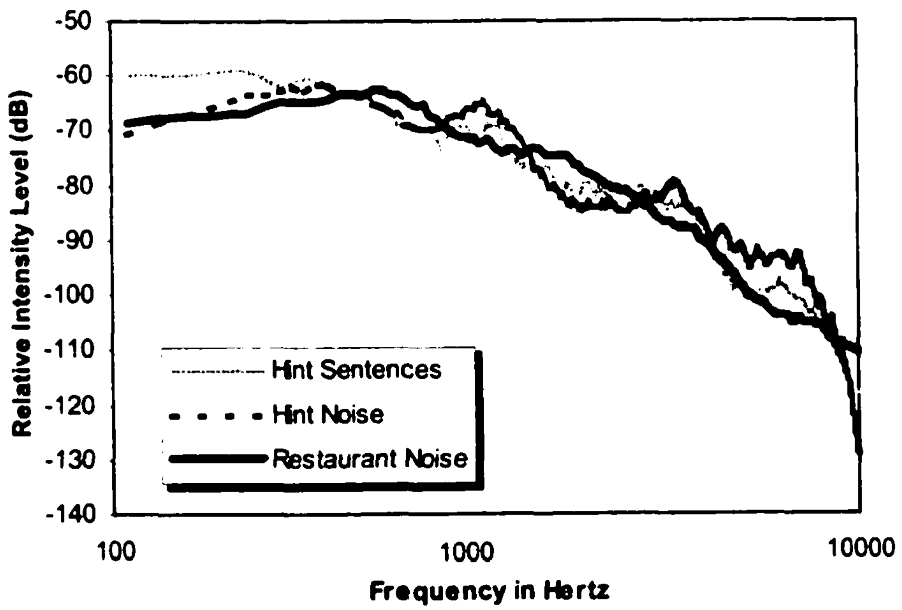


compares the rms versus peak reading calculations for each noise segment.

For the reliability study, segments 12 and 14 were selected as stimuli. Noise segment 12 was selected because it showed the highest variability while noise segment 14 was chosen because it showed the lowest variability. Five normal hearing subjects were used as listeners. Four 20-sentence blocks of the HINT sentences were presented in a counterbalanced manner against the four counterbalanced segments of noise: (1) two identical segments of the commercially available HINT noise and (2) segments 12 and 14 of the restaurant noise. Statistical analysis revealed a 95% critical difference for the HINT noise of 1.7 dB. Recall that Nilsson et al (1994) had obtained a 95% critical difference of 1.20 to 1.51 dB for 20 item lists with listeners having normal hearing.

For the restaurant noise, a 95% critical difference of 0.8 dB was obtained. This result indicates that the use of the restaurant noise segments does not have a negative effect on test-retest reliability. Interestingly, the long-term speech spectrum of the restaurant noise used in the study was almost identical to that of the speech-shaped HINT noise (Figure 3).

Figure 3. Frequency spectra of HINT sentences, HINT noise, and restaurant noise as measured by Sound Forge 4.5



PRELIMINARY ACOUSTIC MEASUREMENTS

Room Characteristics

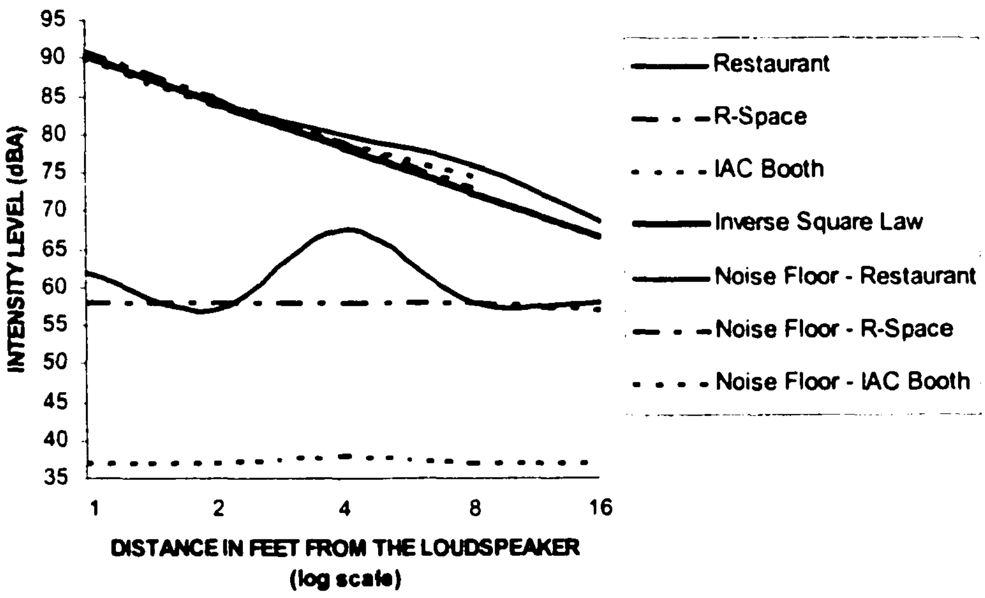
The recordings were made in three different rooms: (1) the restaurant (2) a recording room (which housed the R-SPACE), and (3) an IAC booth. Approximate room dimensions are shown in Table 4 below:

Table 4: Linear dimensions and volume of the three recording venues.

Venue	Dimensions (feet)	Volume (cubic feet)
Restaurant	36L X 36W X 8.5 to 17.5 H (sloping roofline)	22,000
R-Space Recording Room	19.4L X 17W X 7.0H	2613
IAC Booth	12L X 9.4W X 7.5 H	830.7

Estimations of the critical distance were obtained in each recording environment by measuring the level of pink noise at several distances from the same Tannoy Arena loudspeaker used for delivering the pink noise calibration signal (as well as the HINT sentences) to the KEMAR manikin located at 0^0 azimuth, 24 inches away. Measurements were made under acoustic conditions similar to those present during recording sessions. For each of the recording venues, Figure 4 shows the amount of attenuation (A-scale) seen as distance was doubled. In the restaurant and in the IAC booth, the break in the attenuation slope between 4' and 8' suggests a semi-diffuse field beyond that point. In the R-Space recording studio, the critical distance is thought to be somewhere beyond 8 feet. Most importantly, the direct-to-reverberation ratio at 24 inches (the "listening" distance between the loudspeaker and KEMAR) was found to be larger than 10 dB for the restaurant and the IAC booth and at least 15 dB for the R-Space recording studio. Thus, masking effects due to room reverberation did not contaminate free-field SNRs of KEMAR-recorded speech and noise presented to the subjects.

Figure 4. Attenuation of a pink noise signal as a function of distance for the three recording venues (A-scale).



Ambient noise levels measured at two feet in the restaurant (57 dBA), the R-Space recording studio (less than 57 dBA), and in the IAC booth (37 dBA) were well below the stimuli presentation levels and remained consistent, with the exception of a several dB rise at 4 feet in the restaurant, throughout each room. Therefore, the only masking effects present would be those caused by the level of the recorded noise itself, following processing through the hearing aid microphones and mixing with the sentence stimuli.

Microphone Specifications

Specifications for the array microphones used in making the initial recording of the restaurant noise may be found in Appendix B. In situ AI-DI values for the hearing aid microphones used in this study may be found in Table 5 below. These values were obtained with all hearing aids affixed to KEMAR. Polar directional patterns and directivity by frequency graphs for each of the hearing aid microphones may be found in Appendix C.

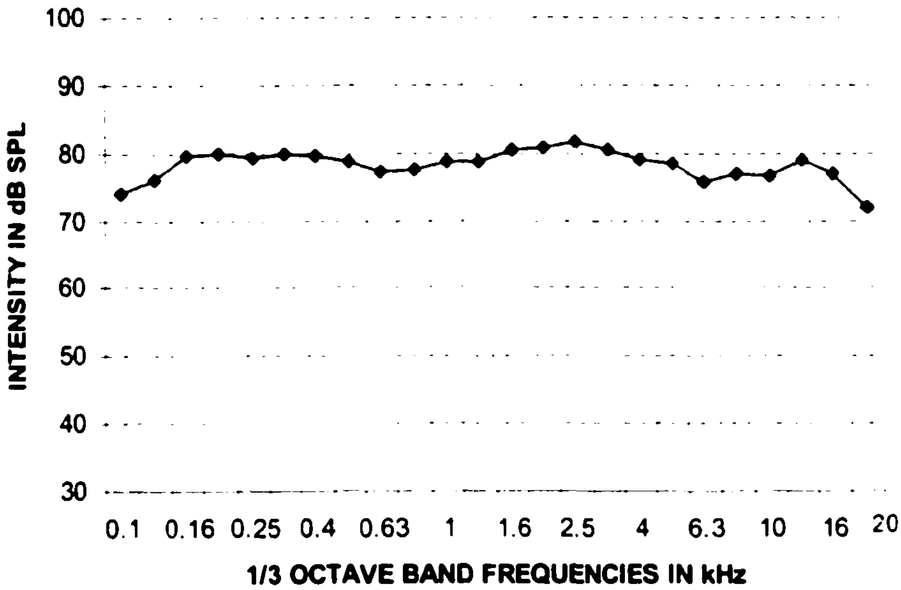
Table 5. AI-DI values for hearing aid microphones as measured on KEMAR under anechoic and diffuse conditions.

KEMAR ~ Anechoic Chamber			
	Right Ear	**Left Ear**	**Average**
ITE Omnidirectional	1.3	0.36	0.83
D-MIC™	4.7	4.32	4.51
Array	8.03	7.33	7.68
KEMAR Diffuse Field Directivity			
	Right Ear	**Left Ear**	**Average**
ITE Omnidirectional	0.96	0.37	0.66
D-MIC™	5.25	4.65	4.95
Array	9.99	9.93	9.96

Sentence Loudspeaker Frequency Response

Spectrum analysis was carried out to determine the frequency response of the loudspeaker used to calibrate all test environments and to present the HINT sentences. A pink noise generator was used to send a signal to a Carver equalizer and Samson amplifier led to the loudspeaker. The level of the noise measured at 9 inches from the loudspeaker was 90dB SPL (A-weighted). The equalized loudspeaker 1/3-octave band response was flat within 3 dB from 160Hz to 16,000Hz (Figure 5). Using the same procedures, spectrum analysis

Figure 5. One-third octave spectrum analysis obtained 9 inches from the loudspeaker, 0^0 azimuth.



also was carried out alongside each of the manikin's ear canals, with and without the hearing aid microphones in place. Results were similar for both ears and both conditions and followed the curve shown in Figure 5.

INSTRUMENTATION, CALIBRATION, RECORDING, AND PREPARATION OF TEST TOKENS

Rationale for Calibration and Recording Procedure

The goal of the calibration and recording process was to relate the free-field SNR at KEMAR's reference point (KRP) (the location at the entrance to KEMAR's ear canal with KEMAR removed from the room) to the SNRs of the audiometer settings used to present the binaural tracks of speech and noise. Ideally, a one-to-one relationship between them should be seen. In order to compare performance of the hearing aid microphones in the various noise environments, an intricate calibration and recording process was required for the preparation of the test tokens. To facilitate comprehension of the details of process (described later), the reader is first referred to the outline below:

1. Five sets of noise recordings were made of the restaurant simultaneously:
    a. A multi-microphone array recording of the restaurant noise (8 tracks).
    b. Three binaural recordings of the same restaurant noise through three binaural sets of hearing aid microphones mounted on KEMAR (6 tracks).
    c. 1 ER-11 overhead omnidirectional calibration microphone track.

2. Each track of the multi-microphone array recording of the restaurant noise was then played back through each of the eight R-SPACE loudspeakers and simultaneously into:
    a. Three binaural sets of hearing aid microphones mounted on KEMAR (six tracks) and
    b. 1 ER-11 overhead calibration microphone (one track).

3. The ER-11 calibration track recorded in the real restaurant was played back through a single loudspeaker placed in front (0^0) of KEMAR in an IAC booth and simultaneously recorded through:
    a. Three binaural sets of hearing aid microphones mounted on KEMAR (six tracks) and
    b. 1 ER-11 overhead calibration microphone (one track).

4. The ER-11 calibration track recorded in the real restaurant was played back through a single loudspeaker placed directly overhead (90^0) of KEMAR in an IAC booth and recorded simultaneously through:
    a. Three binaural sets of hearing aid microphones mounted on KEMAR (six tracks) and
    b. 1 ER-11 overhead calibration microphone (one track).

5. A tape recording of the HINT sentences was played through the target sentence loudspeaker located in front (0^0) of KEMAR. Simultaneous recordings were made of:
    a. Three binaural sets of hearing aid microphones mounted on

KEMAR (six tracks) and
b. 1 ER-11 overhead calibration microphone (one track).

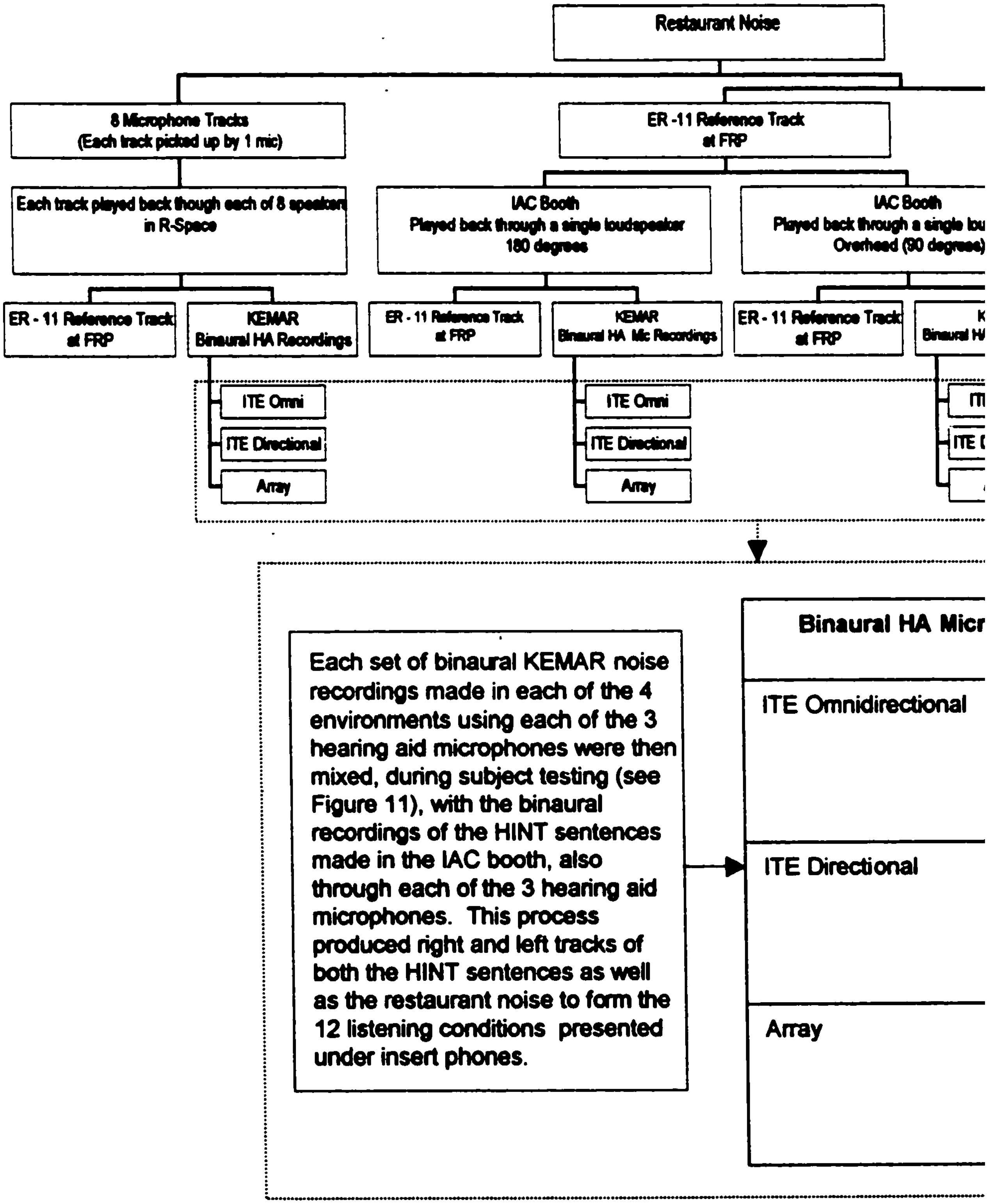
6. Following the recording sessions, portions of the sentence and noise recordings were recorded onto new tapes as test tokens to produce a 3 X 4 design in which binaural hearing aid recordings of the HINT sentences presented at 0^0 azimuth in the IAC booth were to be presented with binaural recordings of restaurant noise as recorded through the same hearing aid microphones under four test conditions:
   a. Live: Eight sources of uncorrelated real restaurant noise.
   b. R-SPACE: Eight sources of uncorrelated restaurant noise played back from the R-SPACE loudspeakers.
   c. IAC booth: Restaurant noise recorded from the ER-11 overhead microphone and played back from a single loudspeaker located directly behind KEMAR (180^0)
   d. IAC booth: Restaurant noise recorded from the ER-11 overhead microphone and played back from a single loudspeaker located directly above (90^0) KEMAR.

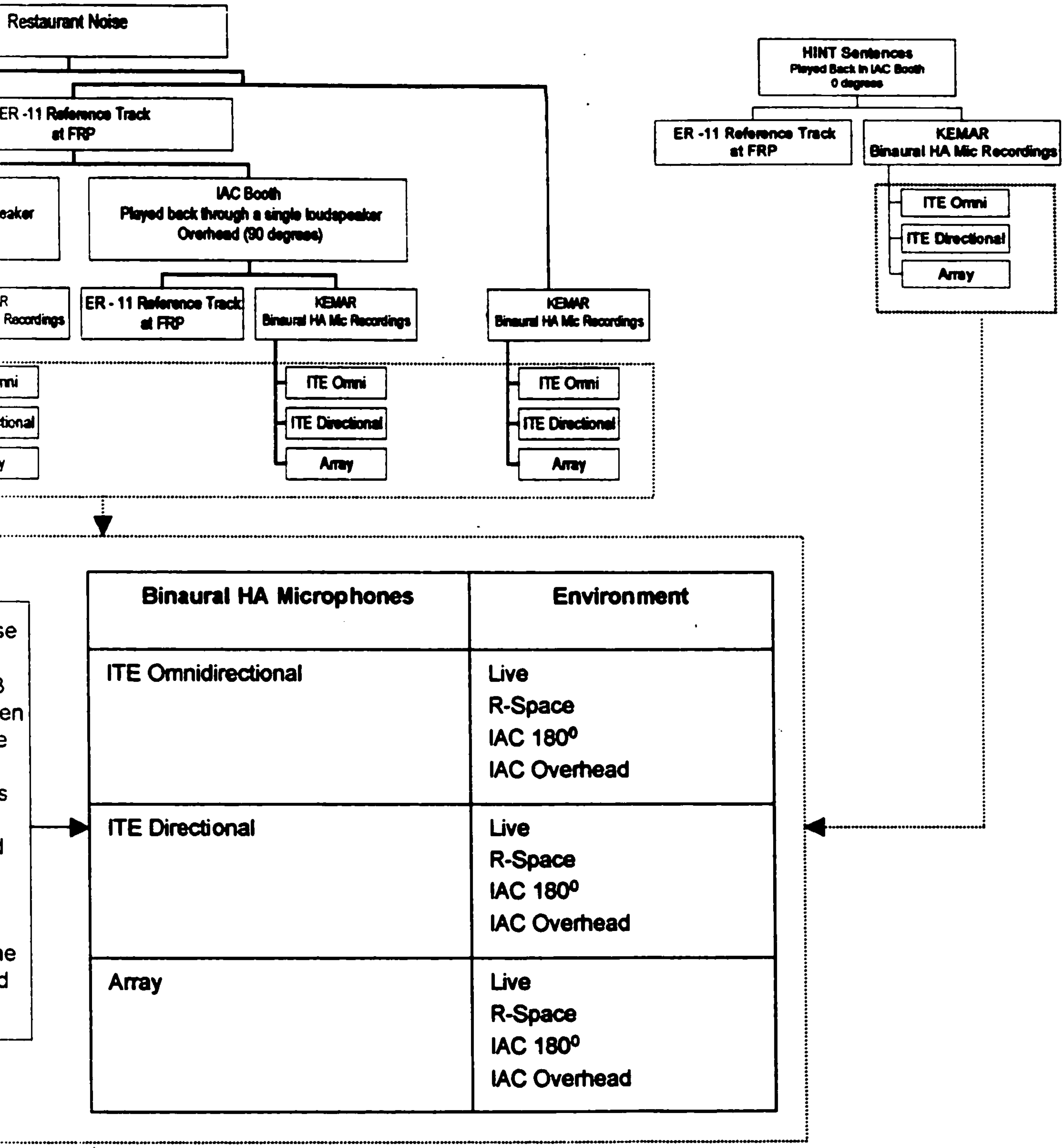
These steps are graphically illustrated in Figure 6 on the next page. The calibration process is described in detail on the following pages and is also outlined in Appendix D.

Instrumentation, Calibration and Recording of Restaurant Noise in the Real Restaurant

Three sets of recordings were made simultaneously in the restaurant environment: (1) Binaural recordings made through hearing aids fitted to KEMAR, (2) a multi-channel recording made from an array of microphones placed around KEMAR, and (3) a calibration track recorded through an ER-11 microphone place above the manikin's head.

Figure 6. Recording process for preparation of test stimuli.




Restaurant Noise
ER -11 Reference Track at FRP
IAC Booth Played back through a single loudspeaker Overhead (90 degrees)
ER - 11 Reference Track at FRP
KEMAR Binaural HA Mic Recordings
ITE Omni
ITE Directional
Array
KEMAR Binaural HA Mic Recordings
ITE Omni
ITE Directional
Array
HINT Sentences Played Back in IAC Booth 0 degrees
ER -11 Reference Track at FRP
KEMAR Binaural HA Mic Recordings
ITE Omni
ITE Directional
Array
Binaural HA Microphones
Environment
ITE Omnidirectional
Live
R-Space
IAC 180°
IAC Overhead
ITE Directional
Live
R-Space
IAC 180°
IAC Overhead
Array
Live
R-Space
IAC 180°
IAC Overhead

The multi-track digital tape recording system (DTRS) consisted of two Tascam DA-38 8-track recorders that were used to record the binaural hearing aid microphone tracks and reference signals, and a third recorder (DA-98) which was used to record the signals for the R-SPACE simulation. Each DTRS unit uses a helical scan head system to record up to 8 tracks of 16-bit, linear pulse-code-modulated digitized audio on a Hi8-size cassette tape. A sampling rate of 48 kHz was used. The three DTRS units were driven in synchrony by the same clock, that of the DA-98. Thus, 24 synchronized tracks were available.

KEMAR was positioned in the middle of the main dining room of the restaurant in a location normally used for a small dining table. This position was nestled among many other nearby tables occupied by diners. The manikin was oriented so that it did not directly face any of the walls of the restaurant, but instead was at an angle. A foam "hairpiece" was affixed to the top of KEMAR's head to reduce the reflections from the head to the reference microphone. Five pairs of binaural hearing aid microphones also were affixed to the manikin: (1) Omnidirectional ITE; (2) ITE D-MICs™ ; (3) omnidirectional BTE; (4) directional BTE microphones, and (5) a prototype pair of ultra-directional, 5-element, endfire array microphones (placed about an inch above each ear and facing slightly downward so as to point toward the center of the ear-level loudspeaker used for calibration and voice recordings). The frontal frequency responses of the ITE and BTE omnidirectional and directional microphones were equalized whereas the array microphones had more of a high-pass response. The effect this had on the results of the experiment will be addressed later. The ITE and BTE omnidirectional and directional microphones were housed inside plastic hearing aid-like shells and inserted and attached to each of KEMAR's ears. Both the ITE and BTE cases contained omnidirectional and directional microphones that functioned simultaneously. This arrangement allowed a direct electrical connection to a properly positioned ITE omni and directional microphone independent of any other hearing aid amplifier or receiver processing. An illustration of the ITE and BTE microphone set-ups may be found in Appendix E. Figure 7 shows the positioning of the hearing aid microphones, the ER-11 calibration microphone, and the foam "hairpiece" on KEMAR's head.

Figure 7. Close-up of hearing aid microphones positioned on KEMAR's head.

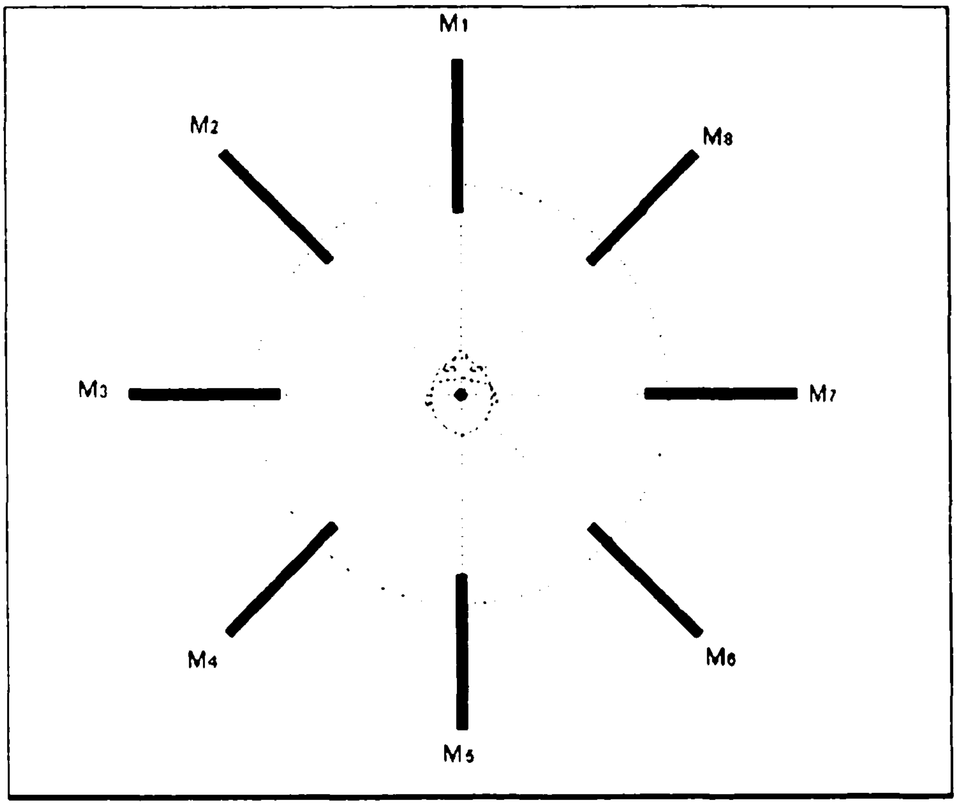
The outputs of the hearing aid microphones were connected to custom-made preamp boxes. The outputs of the preamp boxes were connected to the line inputs of both a Mackie board and a Soundcraft Spirit Folio Notepad mixer before connection to separate recording tracks of the multi-track tape recorders via a submaster mix of outputs from both mixers. In each case, channel separation for left and right signals was accomplished via full-left and -right settings of pan pots on the mixers. Since only restaurant noise recordings from the omnidirectional and directional ITE and the array microphones were used in this study, recordings made with the BTE microphones will not be discussed further.

For the ER-11 calibration microphone, pre-amplification was supplied first by the impedance-matching preamp box supplied with the microphone, and then by an Etymotic Research 1180 line amplifier, which fed an additional input to the multi-track recording system.

To record the restaurant noise to be used as one of the test conditions (live) as well as for source material for the sound field simulator (R-SPACE), eight Sennheiser MKH-70 interference-tube[3] microphones were placed around KEMAR in a circular, horizontal array (Figure 8), from here on referred to as the "multi-microphone array." Each microphone pointed outward from KEMAR in equally distributed, 45-degree angular increments, beginning with one microphone facing directly forward (0^0). Each microphone's directivity was high enough that an informal "walk-around test" in playback revealed focused sources that appeared at the positions stated during the walk-around procedure. The output of each microphone was fed into a separate track of the DTRS. Three recordings of restaurant noise were made simultaneously.

[3] Commonly called "shotgun" microphones, these highly directional microphones typically combine a multiple front sound port design with a conventional cardioid or super cardioid microphone element.

Figure 8. Multi-microphone array (surrounding KEMAR) used to record restaurant background noise for two listening conditions: (1) Live and (2) R-SPACE.



The first set of recordings consisted of 6 tracks of restaurant noise recorded through the three hearing aid microphones binaurally affixed to KEMAR. These recordings were used to prepare the competing noise portion of the test tokens to be used for the restaurant (live) condition.

A second, simultaneous recording consisted of the noise track recorded from the ER-11 calibration microphone placed above the manikin's head. This recording was later played back in the sound booth (and re-recorded through the hearing aid microphones on KEMAR) in order to prepare the two IAC booth listening conditions: (1) Noise behind (180^0) and (2) noise directly overhead

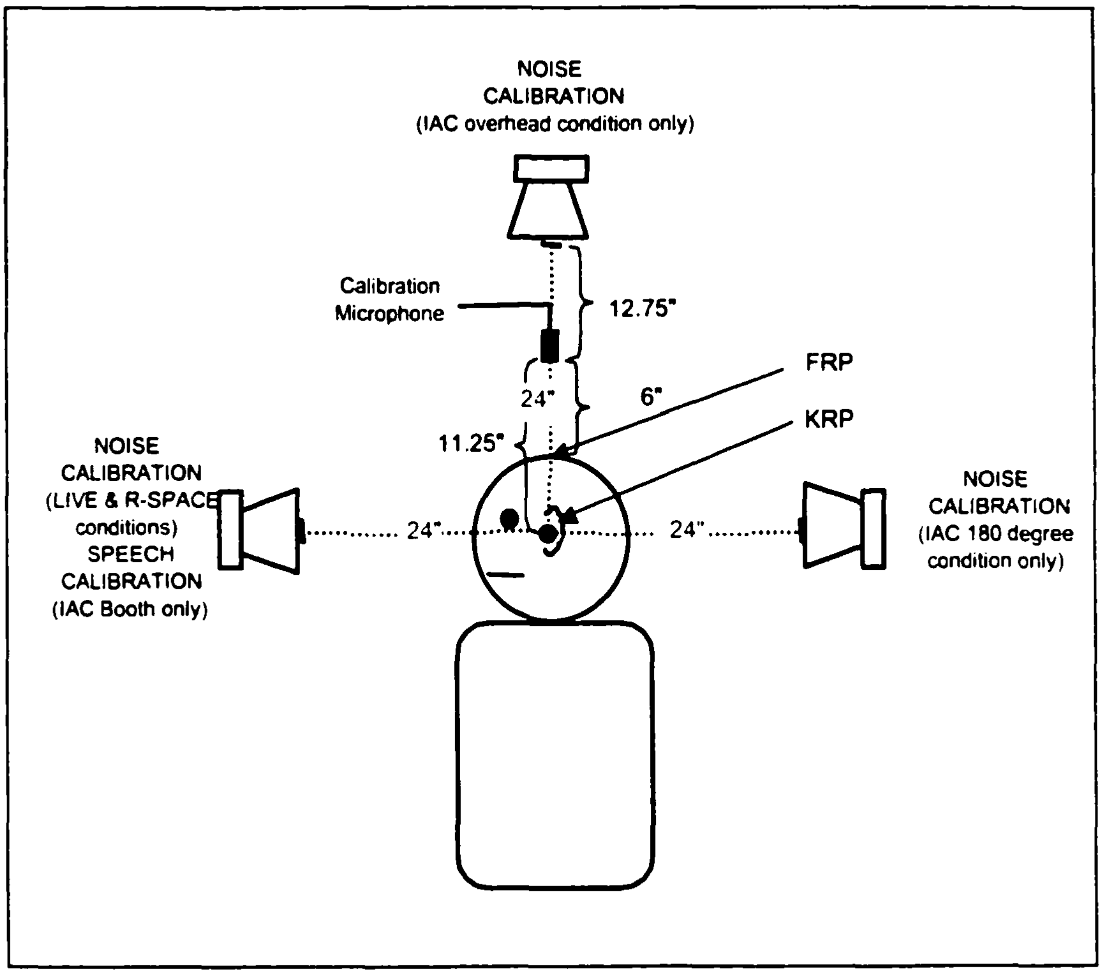
(90^0). The rationale for the ER-11 microphone recording for this purpose will be presented in the first section devoted to the IAC conditions.

A third set of recordings consisted of 8 tracks of restaurant noise as recorded through the multi-microphone array. This set of recordings was later played back through the R-SPACE (and re-recorded through the hearing aid microphones on KEMAR) in order to prepare the competing noise portion of the test tokens representing hearing aid microphone listening in the R-SPACE condition.

A Tannoy Arena loudspeaker, equalized for flat response +/- 3 dB for the 1/3-octave bands centered at 160 Hz to 16 kHz, delivered pink noise from in front of KEMAR, with the front surface of the loudspeaker enclosure 24 inches from KEMAR's reference point (KRP) (ANSI S3.35, 1985). The level of the calibration signal was 84 dB SPL (all SPLs were measured with C-weighting, except where indicated otherwise) at the chosen field reference point (FRP), six inches above the apex of KEMAR. Figure 9 illustrates the calibration set-up for not only the restaurant recording sessions, but also the recording sessions that took place in the simulator and in the IAC booth (which will be discussed later).

This FRP was chosen to permit recording of an ongoing reference signal using the omnidirectional microphone (Etymotic Research ER-11) at the FRP, simultaneously with binaural KEMAR recordings of the restaurant noise using the head-worn microphones. Since the parameter of interest in this study was the level of the noise in the free field (the restaurant), the ER-11 omnidirectional microphone was selected as the reference microphone. Although the normal

Figure 9. Calibration set-up for field reference point (FRP).



placement for this microphone in the free field would be at the center of the manikin's head with the manikin removed from the room (the KRP), the practical constraints of recording in a real restaurant prevented this method of calibration. The 84 dB SPL pink noise was then recorded through each head-worn microphone on its own track at a peak level of -16 dB FS (re full scale) on the DTRS to allow levels as high as 100 dB SPL to be recorded without overload. The audio signal level that produced -16 dB FS was +4 dBu (0 dBu = 0.775 volts), which was at least 18 dB below the clipping level of the microphone preamp systems (Mackie and Soundcraft mixers) used to deliver the signals to the recorders. The calibration signal from the ER-11 reference microphone was recorded at a somewhat lower level, owing to the fixed choices of gain available with the separate preamplifier (Etymotic ER-80) used to deliver the ER-11 signal to the recorder.

In short, in the real restaurant, pink noise was presented from a frontal loudspeaker 24 inches away from the KRP and measured to be 84 dB SPL, 6 inches above KEMAR's head (FRP). This calibration signal was recorded simultaneously on separate tracks of the DTRS through each hearing aid microphone, as well as through the ER-11 omnidirectional reference microphone located at the FRP. Recordings of the live restaurant noise (the sounds of a breakfast party of approximately 42 persons) were made with all recording gains fixed. Later, target HINT sentences were recorded using the same microphones fixed to KEMAR in an IAC booth. Thus, if in the target talker recordings the pink calibration noise is set to produce the same playback level as that recorded in the restaurant, then the SNR at the FRP should be determined solely by directional noise reduction, not level errors. Post-hoc rms analysis using SoundForge 4.5 revealed the restaurant noise to be 9 dB below the pink noise calibration signal, or 75 dB SPL at the FRP. Table 6 lists the microphones and types of recordings made in the restaurant as well as in the other three environments.

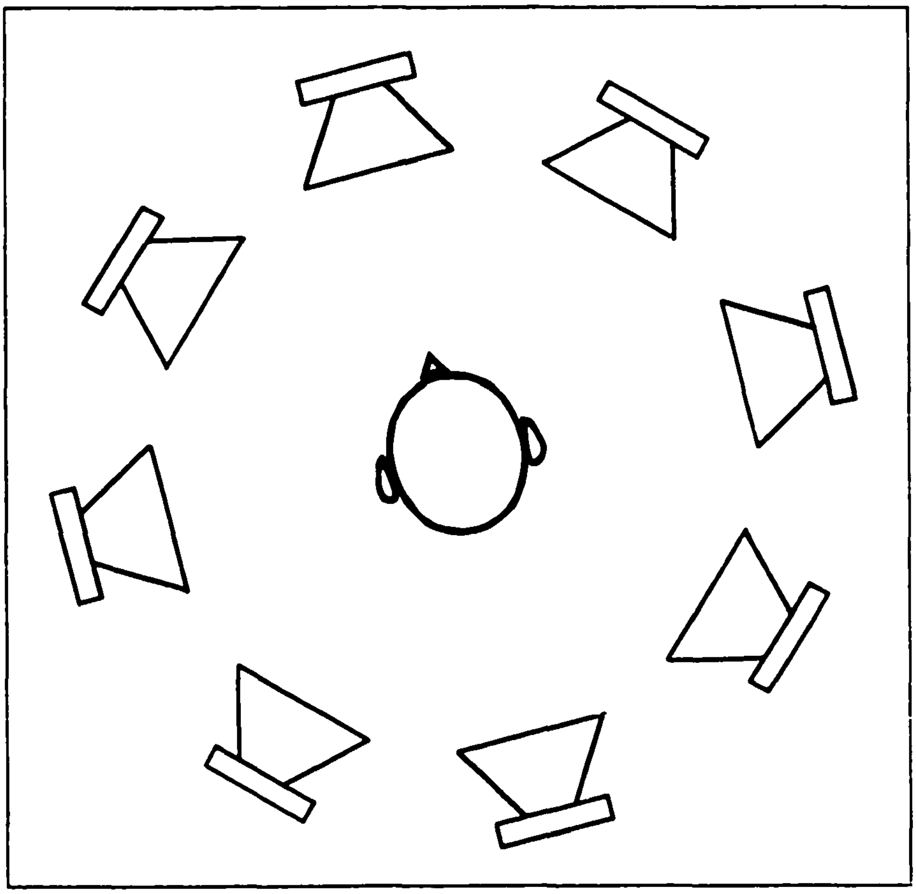
Table 6. Microphones and recordings made in the four test environments.

RECORDING TYPE	MICROPHONE
Calibration Reference for All Four Listening Environments/Source for Noise Conditions Used in IAC Booth	ER 11
KEMAR (Hearing Aids) (All Listening Environments)	Right omnidirectional ITE Left omnidirectional ITE Right directional ITE Left directional ITE Right array Left array
Multi-Microphone Array (Restaurant Only)	0^0 azimuth 45^0 azimuth 90^0 azimuth 135^0 azimuth 180^0 azimuth 225^0 azimuth 270^0 azimuth 315^0 azimuth

Instrumentation, Calibration and Recording of the Restaurant Noise in the R-SPACE Sound-field Simulation System

The R-SPACE consists of an octagonal array of equally spaced, ear-level Tannoy Arena loudspeakers positioned around KEMAR with the emanating surface of each loudspeaker 24 inches from the KRP (Figure 10). The same calibration loudspeaker, KEMAR manikin, head-worn microphones, and ER-11 reference microphone that were used for the previous recordings were employed in this condition. As in the real restaurant, the orientation of the simulator and the manikin were such that neither was directly facing any walls in the simulation room.

Figure 10. R-SPACE playback/recording system.



Calibration procedures for the binaural recordings in the R-SPACE simulator were similar to those for the recordings in the real restaurant. First, using the pink noise previously recorded in the restaurant at 84 dB SPL at each of the eight shotgun microphones, the gain of each corresponding simulator loudspeaker was adjusted to produce 84 dB SPL at the FRP, with one exception. The front-center microphone noise track was reproduced as a phantom image, by splitting the signal between the left-front and right-front loudspeakers. Specifically, the noise track recorded with the shotgun microphone positioned 24 inches in front of KEMAR in the restaurant was split equally between the left-front and right-front speakers such the pink noise from the left-front speaker and the right-front speaker produced about 78 dB SPL at the FRP, respectively. Assuming the two signals added in phase, the total was approximately 84 dB SPL at the FRP.

Next, a pink noise signal was introduced through the equalized sentence loudspeaker, positioned at the KRP 24" in front of KEMAR (Figure 9). The signal was adjusted to produce 84 dB SPL as observed at the FRP. This signal was recorded through each binaural hearing aid microphone as well as through the ER-11 reference microphone with each microphone output connected to a separate track of two DA-38 recorders sync-locked to third recorder, a DA-98 containing the tape of the noise recordings made in the real restaurant. Then, the DA-98 and -38 recorders were started in synchrony. The DA-98 supplied playback of the restaurant noise to the simulator, while the DA-38s recorded the simulation binaurally through KEMAR head-worn hearing aid microphones and through the ER-11 reference microphone located 6 inches above KEMAR's head. Table 6 lists the various tracks recorded in this condition.

Instrumentation, Calibration, and Recording of HINT Sentences in the IAC Booth

Initial plans called for the token sentences for the live condition to be recorded in the same restaurant that was used for recording the live restaurant noise, while additional sentence tokens were to be made in the remaining test environments (R-SPACE and the two IAC conditions). However, noise from a nearby four-lane truck route made it impossible to record speech without distortion and with an acceptable signal-to-noise ratio (SNR). Thus sentences to be used for all test conditions were recorded in a sound-treated IAC booth, with the assumption that the differences among the acoustic conditions in the restaurant and in the various simulation conditions, versus those in the IAC booth, would not significantly affect the relative intelligibility of the sentences across conditions. This assumption was made because of the close distance chosen for delivery of the sentence materials (within the critical distance for each condition). As for the other conditions, the goal of the calibration and recording procedure to be described was to be able to equate the live restaurant calibration to that of the HINT sentences.

The same manikin used for the restaurant and R-SPACE recordings, using the same head-worn and reference microphones, was placed nearly centrally in a double-walled 9'4" X 12' X 7'5" Industrial Acoustics Company, Inc. (IAC) test booth facing at a slight angle relative to the opposing wall and, again, 24 inches from the loudspeaker at 0^0 azimuth (Figure 9). The loudspeaker was located approximately four feet from either wall. Calibration procedures were similar to those described in the previous sections with additional steps being added.

The first step was to establish a sound-field level for the sentences that would cause neither excessive overload nor an unacceptably poor SNR in the array microphones (whose early prototype pre-amplifiers had been subject to both clipping and a fairly high amount of internal noise). This sound-field level was observed, after listening to the sentences through the array microphones, to be such that the speech-shaped calibration noise from the sentence track of the HINT CD registered 73 dB SPL at the FRP. Thus, it was assumed that typical HINT sentences were 11 dB below the 84 dB pink noise calibration signal (as opposed to 9 dB softer than the noise in the live restaurant).

For the second step, the gain of each binaural microphone channel, as well as that of the reference microphone, was adjusted such that the peak sentence level would not overload the respective DTRS track. (It should be noted that the HINT calibration noise and sentences were delivered from a pre-recorded track of a tape being played by the DA-98 recorder that was part of the recording system. This recorder also supplied the clock signal to the two DA-38 recorders used to record the binaural and the reference signals, as in the real restaurant.) Then, pink noise at 84 dB SPL (at the FRP) was recorded through each channel, as in the real restaurant. The "HINT noise" (supplied as part of the HINT test) from the HINT sentence track was recorded from the frontal loudspeaker using the same system gain as that for the sentences, with respect to the original HINT CD. Following the recording of the HINT calibration noise, the HINT sentences were recorded onto the same respective tracks of the DA-38 recorders, as were the binaural signals in the restaurant, but on a separate set of tapes. Again, Table 6 lists the recordings made with the reference and hearing aid microphones.

Instrumentation, Calibration, and Recording of the Restaurant noise in the IAC Booth - Noise at Rear (180^0)

Using exactly the same setup as for the recording of the HINT sentences (described in the previous section), KEMAR was rotated 180^0, so that the loudspeaker that had delivered the sentences from in front would now deliver the restaurant noise from the rear. The tape in the DA-98 that had delivered the HINT sentences also contained, on a separate track, the sound-field reference signal that had been recorded in the real restaurant. Recall that this signal had been recorded via the omnidirectional ER-11 microphone placed at the FRP, 6 inches above KEMAR's head in the restaurant. Since the IAC booth listening conditions were supposed to represent traditional clinical monaural presentation of competing noise, the ER-11 track was deemed to be the signal of choice as it was a single track and its omnidirectional pickup pattern captured sounds from all directions equally.

In order to re-establish the same sound-field SPL that had been present in the restaurant, the 84 dB SPL pink-noise signal that had been recorded via the omnidirectional reference microphone in the restaurant was re-introduced into the loudspeaker in the IAC booth, now to the rear of KEMAR. The drive to the loudspeaker was adjusted such that the pre-recorded pink-noise signal once again registered 84 dB SPL at the FRP, but now in the IAC booth. Then, the entire reference recording made in the real restaurant was played through the rear loudspeaker and recorded through all binaural hearing aid microphones as well as the reference microphone (Table 6) without any further changes to the recording system.

Instrumentation, Calibration, and Recording of the Restaurant Noise in the IAC Booth - Noise Overhead

The loudspeaker was then moved from its ear-level position 24 inches behind the KRP to a position 24 inches directly above the KRP (Figure 9). KEMAR, head-worn microphones, and the reference microphone were placed at the positions and orientations used for all previous recordings. Without any changes to the playback system, the signal that had produced 84 dB SPL at the FRP (from the loudspeaker placed 24 inches in front and then behind the KRP), was re-introduced to the loudspeaker that was now at the same distance, but directly overhead. This signal now produced 85 dB SPL at the KRP (The significance of this will be discussed in a separate section entitled "Post-Hoc Calibration Factors" which can be found in the Results Section).

Before recording the restaurant noise from the overhead loudspeaker, the gain of the ER-11 reference microphone channel had to be lowered (the FRP was now closer to the loudspeaker than before, thus producing a much higher SPL). Following this and without further adjustments to the system, the original omnidirectional reference track of the restaurant breakfast party noise was introduced into the overhead loudspeaker and re-recorded through the binaural and reference microphones onto the corresponding tracks of a separate set of DTRS tapes (Table 6).

Following this recording, the same signal that had produced 84 dB SPL from ear-level was once again introduced, and was observed to be 92 dB SPL at the FRP. The extra 8 dB at the FRP is attributable to the much closer loudspeaker-to-FRP distance with the loudspeaker overhead. This signal was recorded via the ER-11 onto the reference track, for later ease of relating the current reference recordings, delivered from overhead, to the 84-dB SPL calibration signal recorded for the ear-level conditions.

Preparation of Binaural Test Token Tapes

A total of 45 separate tracks were recorded for use in this study (Table 7).

Table 7. Tracks recorded in the various venues. The shaded column shows the 30 tracks used to create the binaurally delivered test tokens.

	KEMAR		ER 11			Multi-Microphone Array
	Sentence Tracks	Restaurant Noise Tracks	Sentence Calibration Track	Restaurant Noise Calibration Track	Restaurant Noise Track for IAC Conditions	Noise Tracks for Live and R-Space
LIVE		6*		1		8
R-SPACE		6*		1		
IAC 0^0	6*		1			
IAC 180^0		6*		1	1	
IAC 90^0		6*		1	1	
OVERHEAD						

*Two tracks (right/left) for each of the three hearing aid microphones.

Following the initial recording sessions, portions of the sentence and noise recordings were re-recorded onto new tapes as test tokens to produce a 3 X 4 design in which binaural recordings of HINT sentences as recorded through the three binaural hearing aid microphone pairs at 0^0 azimuth in the IAC booth, would be presented with binaural recordings of restaurant noise as recorded through the same hearing aid microphones under four test conditions: (1) noise from real restaurant, (2) noise from R-SPACE simulator, (3) IAC booth, noise from single speaker at 180^0 azimuth, and (4) IAC booth, noise from a single overhead speaker.

To restate, the goal of this final re-recording process was to equate the live restaurant noise calibration to that of the HINT sentences so that the test conditions would reflect what the various microphones would have provided to the listener had the sentences and noise been recorded simultaneously.

Preliminary Measurements and Other Preparations for Recording of Test Tokens

Prior to the re-recording process, additional measurements were made on the HINT sentence and restaurant noise recordings.

Measurement of HINT Sentences. The recording of the HINT sentences made in the IAC booth was re-recorded so that all sentences were concatenated, deleting any waveform 40 dB or more below the instantaneous peak level. Using Sound Forge™ 4.5 (2000), the long-term amplitude spectrum also was measured and can be seen in Figure 3. The relative rms levels of the concatenated sentences, calibration noise, and calibration tone from the HINT CD can be seen in Table 8.

Table 8. Relative RMS and peak values for concatenated sentences, calibration noise, and calibration tone from the HINT CD.

	HINT SENTENCES	HINT NOISE	HINT 1kHz CALIBRATION TONE
RMS	-27.2 dB FS	-26.4 dB FS	-15.1 dB FS
PEAK	-3.5 dB FS	-14.2 dB FS	-12.1 dB FS

Measurement of Restaurant Noise. The ER-11 reference microphone recording of the restaurant noise was divided into 14 approximately 2.5-minute long segments. Using SoundForge 4.5, the long-term rms of the noise was measured for each sample. An average VU reading also was calculated for each sample by using a metronome and noting the peak VU reading every 0.75 seconds, using the following procedure devised by Mead Killion (2000).

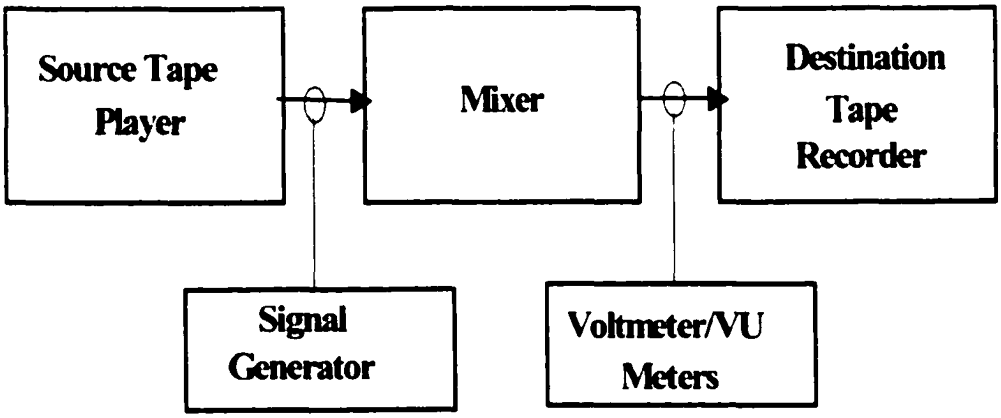
1. The metronome was set to its slowest speed, once every 1.5 seconds.
2. The metronome was turned on and the tape of the noise segments was started.
3. One person listened to the metronome and watched the VU meter (Larry Revit) while the other (this investigator) recorded the VU meter readings.
4. Each time the metronome clicked, the first person announced the VU reading while the second person recorded it.
5. The same tape was played again, but this time about 0.75 seconds later than the first time.
6. Readings were taken again.
7. This resulted in VU readings for every 0.75 seconds for each noise sample.

Sound Forge™ 4.5 (2000) also was used to determine the rms level for each noise sample, relative to the mean values of the metronome readings. The long-term rms values, average VU meter readings, and standard deviations may be found in Figure 2. Note that the long-term amplitude spectrum of the restaurant noise was measured previously, also using Sound Forge™ 4.5 (2000) (Figure 3). Noise segments 9 and 12 were later recorded as practice tokens while the remaining segments were recorded as test tokens.

Equipment setup for preparation of test materials

The following instrumentation was used for preparation of both the sentence and the noise tokens: a Mackie 1604-VLZ mixer, Tascam DA-98 and DA-38 DTRS digital recorders (one each), a Beckman Industrial (Model FG2A) signal generator, two Simpson (model-544) VU meters mounted in an Etymotic Research-built driver box, and a Fluke (model-8060-A) "true-RMS" digital voltmeter (Figure 11).

Figure 11. Recording system for preparation of test tokens



The mixer was set up to route line-level signals from the DA-98 or from the function generator, via pan pots, to "left" and/or "right" outputs which were controlled simultaneously by a dual master fader. The left and right outputs were connected via balanced lines to line inputs of separate tracks of the DA-38 recorder: tracks 1 and 2, respectively, for the sentences, tracks 3 and 4, respectively, for the noise segments. The VU meters received signals from the left and right output channels via a parallel, buffered, unbalanced set of outputs. The voltmeter was moved between the unbalanced inputs of each VU meter circuit as needed.

Preparation of Master Binaural Test Token Tapes for the Live Restaurant Condition

Calibration was carried out with the goal of equating recordings of the live restaurant and the HINT sentences. The 6 tracks of restaurant noise and the 6 tracks of HINT sentences (3 pairs of hearing aid microphones each) were played back and re-recorded onto four tracks of the DRTS, thereby producing stereo recordings of the HINT sentences and restaurant noise as "heard" by KEMAR under each of the three microphone conditions. The mixer was adjusted so 0 VU = -20.7 dB re: the digital signal clipping on record and a 1 kHz calibration tone was recorded.

For the noise, the record gain was set so that the 84 dB SPL pink noise signal was 6 dB above 0 VU. This was derived by the 9 dB average difference between the 84 dB SPL pink noise calibration signal and the 75 dB SPL live restaurant noise, plus a -3 dB safety factor to prevent audiometer VU pegging. Since the original noise level in the restaurant was typically 75 dB SPL, with bursts to 81 dB SPL, setting the 0 VU to equal 78 dB SPL would result in the highest bursts reaching +3 dB on the VU meter. The voltmeter was set so that 0 dB = +6 VU (for the purpose of adjusting each noise segment to equate rms). Thus, the 1 kHz calibration tone was recorded at 78 dB SPL so that -3 dB on the

VU meter would equal 75 dB SPL at the FRP, the average level of the background noise recorded in the restaurant.

For the HINT sentences, the recording level was set so that 84 dB SPL equaled 8 dB above 0 VU (derived from the 84-73 = 11 dB SPL difference, plus a safety factor of 3 dB, or 8 dB). Thus, the 1 kHz calibration tone equaled 78 dB SPL so that -3 dB VU equaled 75 dB SPL as the average level of recorded HINT sentences at the FRP. *Thus, for both the live restaurant noise and the HINT sentences recorded in the IAC booth, 0 VU equaled approximately 78 dB SPL at the FRP.* Table 9 summarizes the relationship between SPL and VU Readings for the sentences and noise.

Table 9. Summary of relationship between SPL and VU Readings for binaural test tokens of HINT sentences and live restaurant noise.

Relationship Between dB SPL and VU Reading	
HINT Sentences	Restaurant Noise
84 dB SPL = +8 VU	84 dB SPL = +6 VU
78 dB SPL = 0 VU	78 dB SPL = 0 VU
75 dB SPL = -3 VU	75 dB SPL = -3 VU

Before each restaurant noise sample was recorded onto tracks 3 and 4 of the DTRS recorder, the overall system gain was adjusted according to the deviations from the mean rms level shown in Figure 2. These adjustments were made to achieve equal rms level for each noise sample and are listed in Table 10.

Table 10: Adjustments made to equate the rms level of each noise segment.

Noise Segment	dB Correction to equate rms
A	+0.26
B	-0.24
C	+1.16
D	+0.36
E	+1.26
F	+1.06
G	+0.56
H	+0.06
I	+0.06
J	-0.34
K	-0.44
L	-0.84

Preparation of Master Binaural Test Token Tapes for the R-SPACE Condition

Again, calibration was carried out with the goal of equating recordings of the restaurant noise played back through the simulator and the HINT sentences. First the system was set such that 0 VU equaled -20 .7 dB FS. The record gain was then raised 14 dB so that the pink noise calibration signal equaled 70 dB SPL (84 - 14 = 70) for all microphones. Mixer equalization was adjusted as follows to reduce a low frequency "bump" that had been discovered previously: (1) The tone generator was set for 0 VU at 1K; (2) An 18dB/octave cut (F_c at 75 Hz) was added; (3) the low frequency shelving cut was adjusted so that 100 Hz equaled -5 dB for all three pairs of hearing aid microphones; (4) It was verified that the pink noise calibration signal was unaffected by these changes.

The record gain was then decreased by 14 dB and then increased by 6 dB so that the 84 dB SPL pink noise calibration signal equaled 6 dB above 0 VU. This was derived by the 9 dB average difference between the 84 dB SPL pink noise calibration signal and the 75 dB SPL restaurant noise, plus a -3 dB safety factor to prevent audiometer VU meter pegging. Thus, the 1 kHz calibration tone equaled 78 dB SPL so that -3 dB on the VU meter equaled 75 dB SPL at the FRP (the level of the background noise recorded in the restaurant).

The voltmeter was set such that 0 dB equaled +6 VU (for the purposed of adjusting each noise segment to equate rms (as shown in table 10 above).

Preparation of Master Binaural Test Token Tapes for the IAC Booth Conditions

Calibration was carried out with the goal of equating recordings of the restaurant noise played back through a single speaker (front and then overhead) in the IAC booth and the HINT sentences. The procedures for calibration and recording of the noise tokens recorded in the IAC booth were precisely the same as for the real restaurant condition and the R-Space condition (minus the equalization procedure).

Summary of Preparation of Master Binaural Test Token Tapes

In summary, calibration was carried out with the goal of equating recordings of the restaurant noise played back through the various speaker arrangements in the four environments and the HINT sentences. Thus, when the test tokens were presented to the subjects, setting the test equipment so that 0 VU equaled 63 dB HL for each of the four tracks (binaural restaurant noise and binaural HINT sentences) meant that the highest bursts of the stimuli were being presented at a level equivalent to 78 dB SPL at the FRP (with -3VU equaling about 75 dB SPL, the average level of the original restaurant noise).

Final Organization, Recording, and Coding of Test Tokens

In order to prepare the test tokens, two 8-track DTRS players and one 8-track DTRS recorder were required. For each hearing aid microphone pair, the appropriate restaurant noise tracks were mixed with the HINT sentence IAC booth recordings using the following DTR recordings: (1) One playback of binaurally recorded noise segments from the real restaurant; (2) One playback of binaurally recorded noise segments from the R-Space simulator; (3) One playback of binaurally recorded noise segments from the frontal speaker in the IAC booth; (4) One playback of binaurally recorded noise segments from the overhead speaker in the IAC booth. This mixing process resulted in a DTR 4-track recording of HINT sentences (tracks 1 and 2; left/right) and noise segments (tracks 3 and 4; left/right). Each 20-sentence list was matched to one of 12 possible noise segments using a Hyper Greco Latin Square Design. Table 11 shows the codes used in the design matrix to denote each test condition. Table 12 shows the study's design as well as how each test token was crafted.

Table 11. Codes used to denote test condition.

Code	Condition
I	Omnidirectional ~ noise at 180^0, IAC booth
II	Omnidirectional ~ real restaurant noise
III	Array ~ noise at 180^0, IAC booth
IV	Array ~ real restaurant noise
V	Omnidirectional ~ noise overhead, IAC booth
VI	Omnidirectional ~ R-SPACE noise
VII	Array ~ noise overhead, IAC booth
VIII	Array ~ R-SPACE noise
IX	D-MIC ~ noise at 180^0, IAC booth
X	D-MIC ~real restaurant noise
XI	D-MIC ~ noise overhead, IAC booth
XII	D-MIC ~ R-SPACE noise

Table 12. Hyper-Greco Latin Square Design: For each of the 12 conditions (defined in Table 13), each 20-sentence list is paired with a noise sample. Each sentence/noise combination is unique.

SUBJECT	I	II	III	IV	V	VI	VII	VIII	IX	X	XI	XII
A	1a	2b	3c	4d	5e	6f	7g	8h	9i	10j	11k	12l
B	2d	1c	4b	3a	6h	5g	8f	7e	10l	9k	12j	11i
C	3b	4a	1d	2c	7f	8e	5h	6g	11j	12i	9l	10k
D	4c	3d	2a	1b	8g	7h	6e	1f	12k	11l	10i	9j
E	9e	10f	11g	12h	1i	2j	3k	4l	5a	6b	7c	8d
F	10h	9g	12f	11e	2d	1k	4j	3i	6d	5c	8b	7a
G	11f	12e	9h	10g	3j	4i	1l	2k	7b	8a	5d	6c
H	12g	11h	10e	9f	4k	3l	2i	1j	8c	7d	6a	5b
I	5i	6j	7k	8l	9a	10b	11c	12d	1e	2f	3g	4h
J	6l	5k	8j	7i	10d	9c	12b	11a	2h	1g	4f	3e
K	7j	8i	5l	6k	11b	12a	9d	10c	3f	4e	1h	2g
L	8k	7l	6i	5j	12c	11d	10a	9b	4g	3h	2e	1f

The 12 conditions are listed across the top in the first row. Each cell of the following 12 rows contains a number and a letter. The number corresponds to the sentence list recorded for that condition while the letter corresponds to the noise sample recorded for that condition. It can be seen that each sentence and noise segment forms a unique pair. The rationale for this design will be discussed in the design section. Each sentence list is composed of 20 sentences (List 1 is comprised of original HINT sentence Lists 1 and 2, List 2 is composed of sentence lists 3 and 4, etc). Each letter corresponds to one of the 12 noise segments. Table 13 shows the letter codes used to specify which original noise segments (Figure 2) were used as test versus practice tokens. HINT sentence List 25 (10 sentences only) was paired with the first half of noise segment P1. Since List 25 was used only for checking presentation levels, it seemed reasonable to pair it with noise segment P1 (which has a large standard deviation and high rms, compared to the other noise segments). HINT sentence list "Practice 1" was paired with the second half of noise segment P1, HINT list "Practice 2" was paired with the first half of noise segment P2, and HINT list "Practice 3" (not used) was paired with the second half of noise segment P2.

Table 13. Codes used to delineate which original noise segments are being used as test and practice tokens.

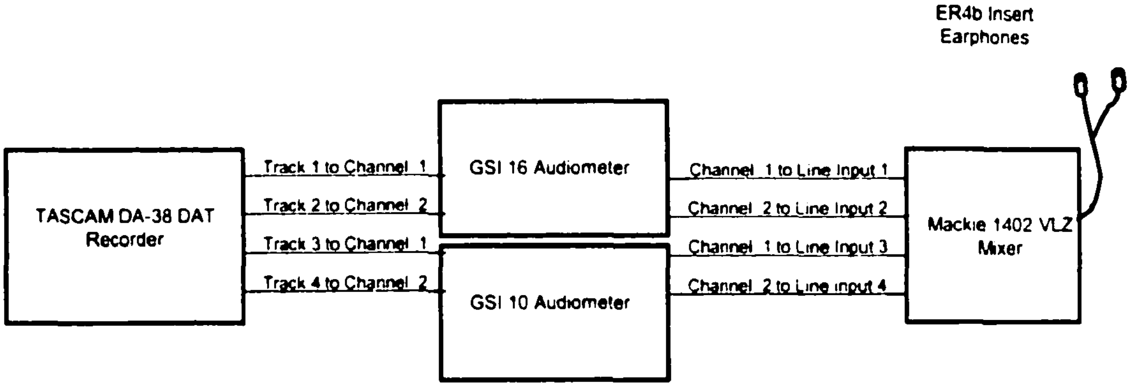
	Test Tokens												Practice Tokens	
Token Letter	A	B	C	D	E	F	G	H	I	J	K	L	P1	P2
Original Noise Sample Number	1	2	3	4	5	6	7	8	10	11	13	14	9	12

DATA COLLECTION

Instrumentation

Figure 12 illustrates the instrumentation employed for token presentation. A Tascam DA-38 Digital Audio Tape Recorder was used to play the test stimuli to the listeners. The DA-38 was connected to two 2-channel audiometers.

Figure 12. Instrumentation used for presentation of test tokens to subjects.



Tracks 1 and 2 of the DA-38 DATR were used to deliver the HINT sentences to channels 1 and 2 of a Grason-Stadler 16 audiometer. The audiometer outputs of channels 1 and 2 were led to line inputs 1 and 2 of a Mackie 1402 VLZ Mixer. Tracks 3 and 4 of the DA-38 DATR were used to deliver the noise recordings to channels 1 and 2 of a Grason-Stadler 10 audiometer. The outputs of channels 1 and 2 were then led to line inputs 3 and 4 of the mixer. A pair of ER4b insert earphones was used to deliver the binaural sentence and noise tracks to the subjects who sat inside an IAC test booth.

Calibration

Channels 1 and 2 of both audiometers were adjusted to read 0 VU using the 1000Hz sinusoid calibration tone described previously and attenuator settings of 63 dB HL. One-by-one, for each of the four channels, as the 1000Hz calibration tone was played, the mixer settings were adjusted to produce 78 dB SPL rms from each ER4b insert into a 2cc coupler connected to a Frye 6500 real ear analyzer (with the reading initially verified by a Larson-Davis 800B Sound Level Meter). Recall that 0 VU equaled 78 dB SPL and that -3 VU equaled 75 dB SPL, the average level of the original restaurant noise and HINT sentences.

Test Procedure

Prior to testing, oral as well as the following written instructions were provided to each participant:

> This is a test of your ability to hear soft speech in a restaurant setting. You will first hear some restaurant noise. After a few seconds, you will also hear a man reading a sentence. The loudness level of the man's voice will change during the testing. Sometimes it will be very faint. Repeat everything you hear the man say, even if you have to guess. I will stop after each sentence to allow you to repeat what you heard. Again, please repeat anything you hear, even if it is only a part of the sentence or a part of a word. You are encouraged to guess!
>
> Some sentences will be repeated a little bit louder each time until you repeat them correctly. Most sentences will be presented only once. The loudness levels of the sentences are chosen to be soft on purpose so do not be discouraged if you cannot understand the entire sentence every time. No one can! Again, make sure you repeat everything you hear and do not be afraid to guess.
>
> I will begin by giving you some practice sentences so you can become familiar with the man's voice and the various noises you will be hearing. Please feel free to ask questions or make comments while you are practicing. There will be two practice sections of 10 sentences each and twelve test sections of 20 sentences each. If you need a break, let me know as we can stop between sections. Testing can be completed at another time if you become fatigued.

Following the instructional process, each participant was comfortably seated in the test booth with the ER-4b insert earphones inserted into each ear. Sentence List 25 was presented along with the noise tracks at 63 dB HL in each ear. The sentences were increased to 76 dB HL to ensure the presentation levels were comfortable and heard at the midline of the subject's head. Practice Lists 1 and 3 (ten sentences each) were then presented.

The listener's task was to repeat the sentences spoken by the male talker in the presence of the restaurant noise presented at a fixed level of 63 dB HL in each ear. The procedure estimates the SNR at which the sentences, embedded in noise, can be repeated correctly 50 percent of the time. Correct identification of each sentence is based on proper repetition of all words of each sentence, with the exception of certain articles where substitution was allowed (e.g. "a" for "the"). The presentation level of the sentences was kept the same in each ear and was adaptively adjusted depending on the listener's response. An incorrect response resulted in the speech presentation level being raised and a correct response resulted in the speech presentation level being lowered for the next trial. The level of the sentence stimuli presented to each ear was varied in 4-dB steps for trials 1 through 4 and in 2-dB steps for trials 5 through 20. If a correct response was noted for the 20th (and last) trial, then a hypothetical 21st trial would occur 2 dB lower. If an incorrect response was noted for the 20th trial, then a hypothetical 21st trial would occur 2 dB higher. To calculate the SNR, the attenuator settings (presentation levels for the sentences) for trials 5 through 21 were averaged and, from this result, the audiometer dial setting for the noise (63 dB HL) was subtracted. It should be noted that instructions for the standard HINT test call for playing the initial sentence 10 dB below the noise level if the speech and noise originate from different speakers. Pilot work for this investigation revealed the need for presenting the first sentence at significantly softer levels due to the noise rejection capabilities of the directional microphones. In fact, initial presentation levels were often at least 20 dB below noise levels for the array overhead condition (IAC booth). Presentations of listening conditions were randomized.

EXPERIMENTAL DESIGN

Each subject was presented with 12 sets of sentence/noise sample tokens representative of KEMAR recordings made using the 3 binaural sets of hearing aid microphones in the 4 noise environments. A 12 x 12 Hyper-Greco Latin Square design was employed (Table 12) such that each condition consisted of a unique sentence/noise sample combination. This design was selected because it is efficient when the factors of interest have more than two levels and no (or negligible) interactions are expected between factors. For example, to examine the effect of three hearing aid microphones, four noise environments, 12 sentence lists, and 12 noise segments on HINT sentence performance, a full 3 x 4 x 12 x 12 factorial design, resulting in 1728 experimental runs would need to be done. Since the HINT sentences have been standardized to their own speech spectrum ("HINT" noise) and a pilot of the restaurant noise revealed good reliability with more than one noise sample, it was not expected that interactions between the sentence lists and noise segments would occur nor should they interact with the other factors of interest. Of main concern was the estimation of main effects of hearing aid microphone type and noise environment and perhaps an interaction between these two factors. The Hyper-Greco Latin Square design spreads the error variance of the sentence lists and noise segments so that the main effects of microphone and environment may be seen.

RESULTS

Raw Data

Table 14 shows the mean across-subject performance for the three hearing aid microphones and four noise environments along with the standard deviations for these conditions. Scores are reported as SNRs for 50% correct performance for the HINT sentences. For the analysis, post-hoc calibration corrections were applied to the live condition. Corrected values are shown in italics. The rationale for this is described in the following section.

Table 14. SNR as a function of microphone and noise environment. Across subject means and standard deviations are shown.

Condition	Omni		D-MIC		Array	
	Mean	s.d.	Mean	s.d.	Mean	s.d.
IAC 180	-4.1	1.4	-12.7	1.3	-13.7	1.5
IAC 90	-6.1	1.2	-11.4	1.5	-20.8	1.1
R-Space	-6.2	0.9	-9.8	1.6	-12.0	1.1
Live	-6.7	1.2	-11.3	1.4	-12.8	1.2
Live Corrected	***-5.7***	***1.2***	***-10.3***	***1.4***	***-11.7***	***1.2***

Post-Hoc Calibration Corrections

Recall that the goal of the calibration and recording procedure was to relate the free-field SNRs at the KRP to the SNRs of the audiometer settings used to present the binaural tracks of speech and noise. The calibration process resulted in a 0 VU setting achieving peak sentence noise levels of 78 dB SPL at the FRP. Table 15 shows the resulting KRP calibration levels for the various loudspeaker locations for the four listening environments.

Table 15: Calibrated levels (at the KRP) for sentence and noise tokens for various loudspeaker locations in the four test venues.

Loudspeaker Location(s)	Restaurant	R-SPACE	IAC Booth
Speech 0^0 in front: 24" from apex of KEMAR's head, horizontal plane	79 dB SPL	79 dB SPL	79 dB SPL
Noise 180^0 behind: 24" from apex of the KEMAR's head, horizontal plane			79 dB SPL
Noise 90^0 overhead: 12.75" from KEMAR's apex, vertical plane			79 dB SPL
Live restaurant noise surrounding KEMAR	78 dB SPL	79 dB SPL	
Noise presented via an 7-loudspeaker array with phantom frontal image		79 dB SPL	

Post-Hoc Correction Factors for Live Condition

As seen above, for all but the live condition, the level of speech and competing noise at KEMAR's ears was determined to be 79 dB SPL. Unlike the other conditions, competing noise in the live condition remained at 78 dB SPL at the KRP because it was picked up and recorded from a distance of at least 10 feet away (see Appendix F). As a result, the actual free-field SNRs presented to the subjects were 1 dB better than calibrated for and all raw scores for the live condition were adjusted to 1 dB worse (less favorable SNRs) prior to data analysis.

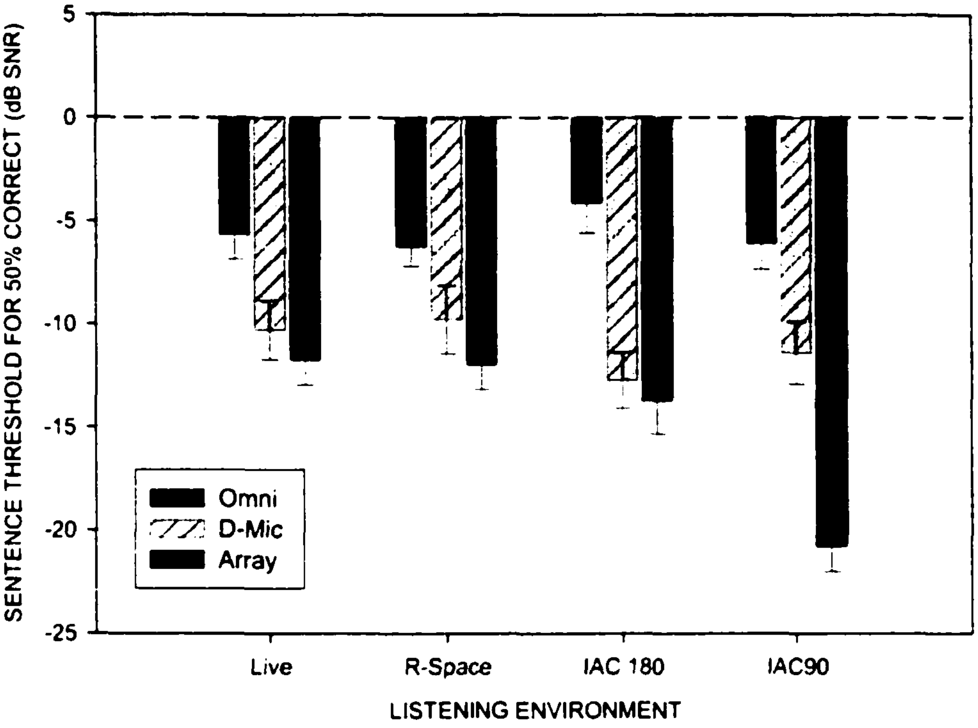
Post-Hoc Correction Factor for Live Array Condition Only

An additional small correction factor also was applied to the live array microphone scores <u>only</u> due to slight clipping of the array tracks. Appendix G details an experiment in which it was discovered that equalized pink noise was 0.5 dB higher than array noise (the ratio of the pink calibration noise to the HINT sentences was 0.5 dB less for the array microphones). Because of this, the level of the sentences presented to the subjects would have been 0.4 higher for the array microphones. To compensate for this, each subject's score in the live array condition was made 0.5 dB better. This resulted in a net correction for the live array scores of +0.5 dB [(+1) + (-0.5) = +0.5], meaning that all live array scores were made worse by 0.5 dB prior to data analysis.

Corrected Results

Mean subject performance for the three hearing aid microphones and four noise environments is shown in Figure 13 (post-hoc correction factors were applied). To analyze these results, a two-way (2 within; 0 between), fixed-effects analysis of variance (ANOVA) was performed. The within-subjects factors were microphone type (3 levels) and noise environment (4 levels). The dependent variable was the HINT sentence threshold for 50% correct across the levels of the two factors. Results revealed that as microphone directivity increased, SNR thresholds improved significantly ($p < 0.01$) across environments. A main effect also was seen for noise environment ($p < 0.01$). Most importantly, a significant two-way interaction also was found between microphone type and noise environment ($p < 0.01$). The error variance of the distribution was 1.4 dB, showing good test-retest variability. The ANOVA summary table is shown in Table 16.

Figure 13. Field-referred SNRs required for HINT sentences to be repeated correctly 50% of the time across three hearing aid microphone conditions and four noise delivery environments. Means and standard deviations are shown.



A

Table 16. ANOVA Summary Table

Effect	df Effect	MS Effect	df Error	MS Error	F	p-level
Microphone	2	1024.947	22	1.21782708	841.6195068	0.000000
Environment	3	92.447617	33	1.39750445	66.15193176	0.000000
Interaction	6	71.74942	66	1.41554821	50.686684	0.000000

The purpose of the investigation was to test the hypothesis that a listener's threshold for speech in noise with a particular type of hearing aid microphone would be similar, whether listening to binaural recordings made in the simulator or in the real restaurant. It also was theorized that the simulator technique would provide more accurate information about performance in the real world (live) than would either the common clinical ($0^0/180^0$) or the unique (noise overhead) IAC booth techniques. For that reason, the live condition was chosen as the "gold standard" to which subject performance in each of the other three environments would be compared, across microphone types. Nine separate a priori contrasts were made. To determine whether significant differences existed for the means, the Bonferroni method of multiple comparisons (Dunn, 1961) was employed. Tables 17 and 18 summarize the results of this analysis.

Table 17. Calculation of the Critical difference between two means using the Bonferroni Method.

Error Variance of the Distribution	Variance of the Mean (MS Error/12 Ss)	Standard Deviation of the Mean	Standard Error of the Difference Between Two Means	Critical Difference
1.416	0.118	0.343	0.486	1.39
Microphone x environment	1.416/12 Ss	SQRT of 0.118	0.343 * SQRT of 2	t of 2.86 * 0.486

Table 18. Determination of significant differences in mean microphone performance between the live condition versus performance in the simulator and in the two IAC booth conditions.

Listening Condition Contrasted Against Live Condition	Microphone	Difference in Mean Score in Question and Live Condition	Significant Difference
Simulator	Omni	0.5	No
	D-MIC	0.5	
	Array	0.3	
IAC Booth - 180^0	Omni	**1.6**	**Yes**
	D-MIC	**2.4**	
	Array	**2.0**	
IAC Booth - Overhead (90^0)	Omni	0.4	No
	D-MIC	1.1	No
	Array	**9.1**	**Yes**

DISCUSSION

Comparison of Microphone Performance in the Live Versus Simulator Environments

Results of the analysis showed that performance was similar, for each microphone condition, whether subjects listened to noise recordings made in the simulator or in the real restaurant. In fact, there was only a 0.3 to 0.5 difference between each of the three microphones, compared across the two listening environments. These results strongly support the hypothesis that the simulation replicated the acoustic conditions of the restaurant, to the extent that: (1) The simulation sounded close to real, this conclusion based on informal (but skilled) listening checks of localization and a formal pilot study (Revit, Schulein, Compton, & Killion, 2000); (2) For all practical and statistical purposes, subjects listening through hearing aid microphones of differing directivity performed the same on tests of speech intelligibility in noise in the simulator as they did in the restaurant.

The improvement seen with the D-MICs as compared to the omnidirectional microphones was between 3.6 and 4.6 dB and is in rough agreement with the AI-DI values for the directional microphones of 3.3 (anechoic) to 4.3 (diffuse). Improvement seen (over the omnidirectional microphones) with the array microphones was 5.8 dB for the simulator and 6.6 dB for the live condition. This fell somewhat short of what was expected, i.e., AI-DIs of 6.9 (anechoic) and 9.3 (diffuse). This discrepancy could be due to the fact that the array microphone had more of a high-pass frontal response than the other two microphones (discussed later). This also would explain the small improvement (two dB) seen between the D-MIC and array conditions in both the simulator and live environments, when an improvement of 3 to 5 dB was expected (based on average AI-DI values for the microphones used in this study).

Comparison of Microphone Performance in the Live Versus Traditional Listening Environments

The second hypothesis of this study was that the simulator technique would provide more accurate information about performance in the real world than would the more traditional (competing) techniques of evaluation. Inspection of Figure 13 clearly shows a discrepancy in the pattern of performance for the microphones in the IAC 180^0 and 90^0 conditions as compared to the live and simulator conditions. This was verified by the Bonferroni analysis (Tables 17 and 18), which found significant differences between the live condition and the 180^0 condition for all three microphones and between the live condition and the 90^0 condition for the array microphone only. Results for each environmental condition will be discussed in more detail under separate headings.

Comparison of Microphone Performance in the Live Versus $0^0/180^0$ Listening Environments

As discussed in the literature review, presenting speech from in front with a single competing noise directly behind the listener has been a commonly accepted way of measuring omnidirectional and directional microphone performance. However, as shown in Figure 13 and Table 18, presenting a single noise from behind resulted in a significantly different pattern of microphone performance as compared to that seen in the real restaurant environment. Subjects listening to recordings made through the D-MIC™s and the array microphones achieved *better* SNRs (2.4 dB and 2.0 dB, respectively) in the 180^0 condition versus the live condition. These results can be explained, in part, by examining the relationship between the spatial characteristics of the sound sources and the directivity of the microphones. Figures 14 and 15 show the in situ polar directional patterns of the right D-MIC™ and array microphone used in this study. Both microphones have noticeable rear lobes. The D-MIC™ has a very large pick-up pattern in the front hemisphere and its nulls are located approximately at 120^0 and 265^0. The array microphone has a narrower pick-up pattern in the front hemisphere and deeper nulls at 90^0 and 270^0. According to research by Ricketts (2000), "in an environment without reverberation, and given a particular hearing aid's polar directivity pattern with a signal of interest directly in front of the listener, the SNR from a pair of directional hearing aids will be dependent on the relative intensity level of the competing noise integrated over *all angles* of the polar pattern." Thus, it seems reasonable to assume that, in the case of the $0^0/180^0$ condition, better performance for both directional microphones occurred because noise not present in the front hemisphere as it was in the live and simulator conditions.

Figure 14. In situ polar plot of right D-MIC (all microphones on KEMAR), testing carried out 24 inches from loudspeaker.

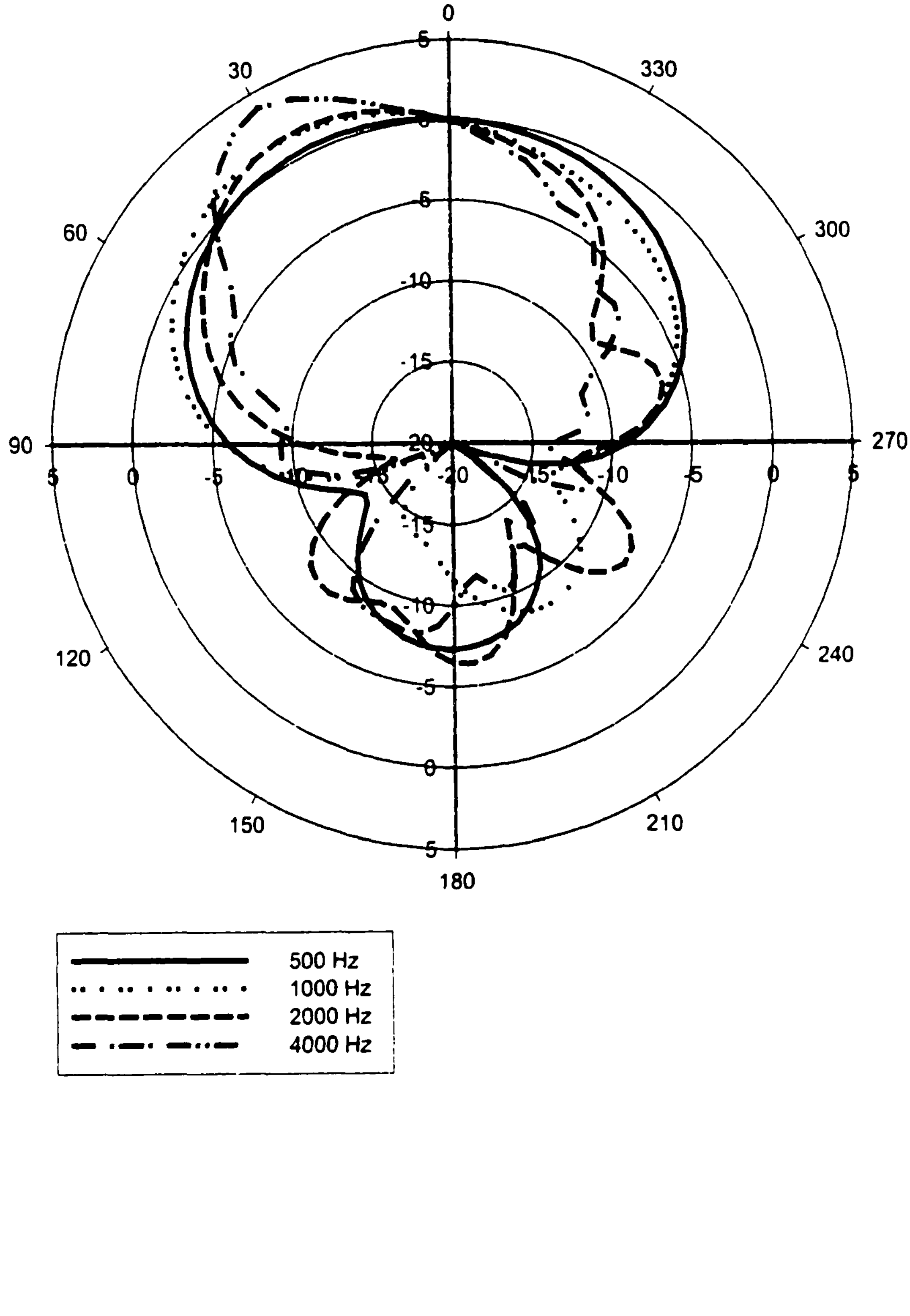
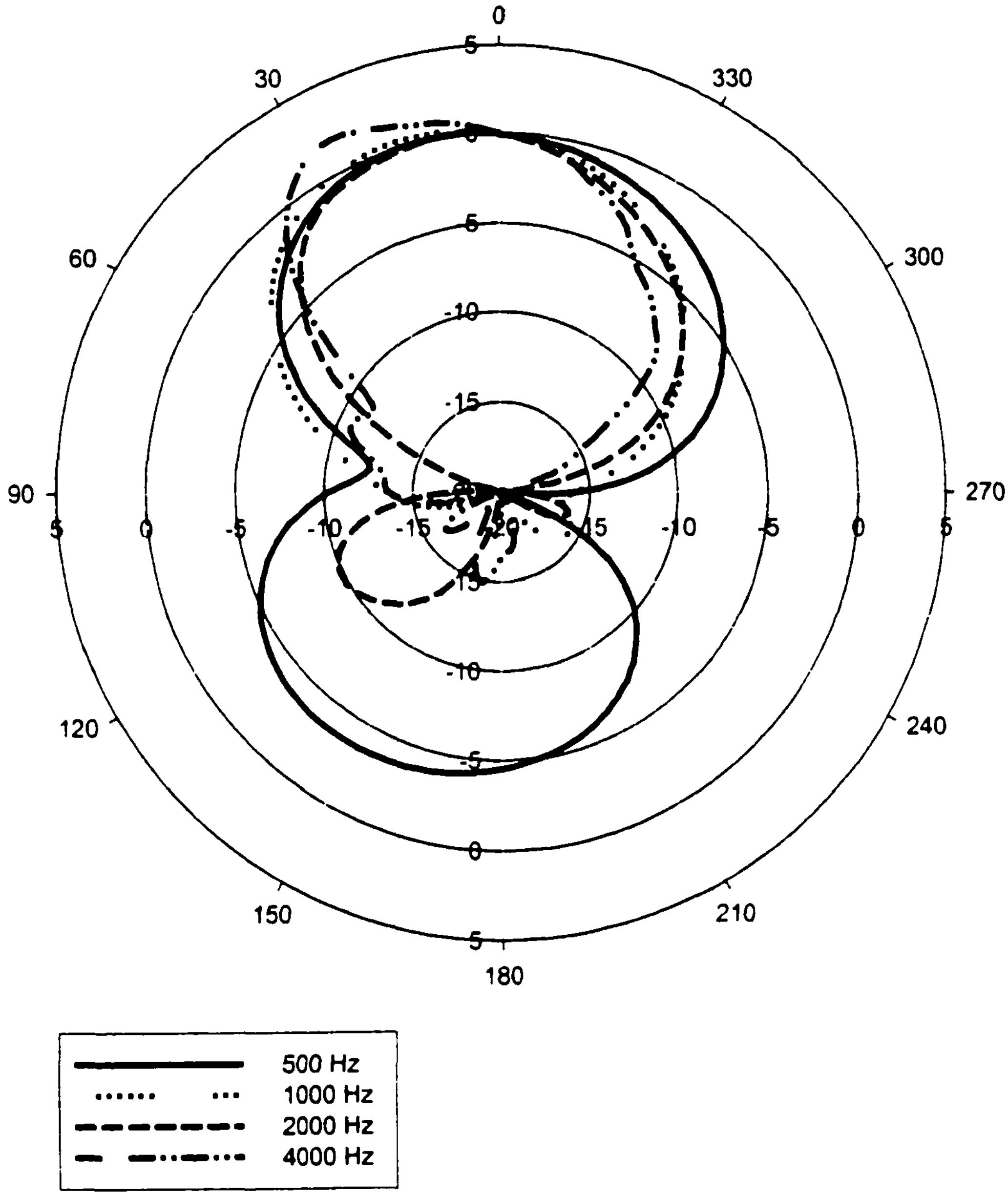


Figure 15. In situ polar plot of right array microphone (all microphones on KEMAR), testing carried out 24 inches from loudspeaker.



Interestingly, in the $0^0/180^0$ condition, omnidirectional microphone performance was in the opposite direction. In fact, it was about 2 dB worse than performance was for all three of the other listening environments. A detailed explanation of these contradictory results can be found after the discussion of Bonferroni comparisons for the fourth and final (90^0) listening environment.

Comparison of Microphone Performance in the Live Versus Overhead (90^0) Listening Environments

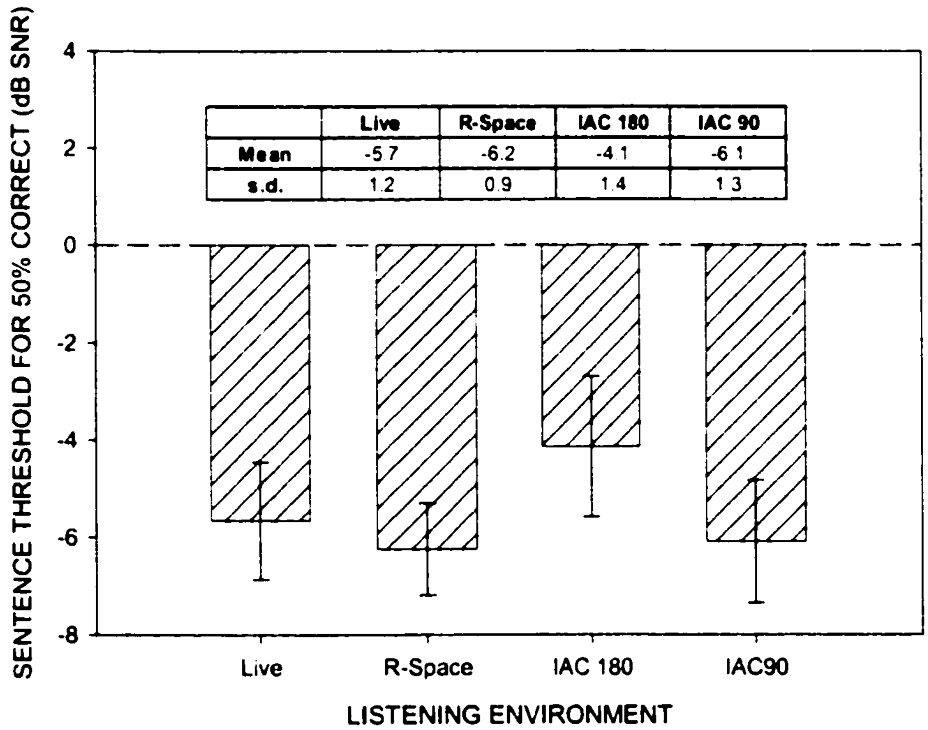
Inspection of Figure 13 and Table 18 shows that, with the exception of the array microphones in the IAC 90^0 condition, a similar pattern of performance was seen for the microphones in both the IAC 90^0 and simulator/live conditions. In the IAC 90^0 condition, omnidirectional microphone performance was not significantly different from that seen in the live condition. This was anticipated in view of the polar pick-up patterns of the omnidirectional microphones. While the 1.1 dB better performance for the D-MIC™ in the overhead condition was not found to be statistically significant, if not due to chance, then perhaps the 1.1 dB better performance is due to the fact that the polar pattern of the D-MIC shows a significant reduction in output beginning at its mid-line (90^0 and 270^0). If the null is continuous to the vertical plane, then this attenuation pattern would have fallen almost directly under the overhead noise source (Figure 14). Even better performance in this noise condition might have been seen had the area of most rejection fallen at the midline rather than at 110^0 and 255^0. The most important result in the $0^0/90^0$ condition was the impressively large improvement in SNR achieved when subjects listened to recordings made through the array microphones. This improvement was 9.1 dB better than that achieved for the array microphones in the live environment. The reason for this improvement is due to the location of the single noise source in relation to the array microphones' polar patterns. The loudspeaker was positioned directly above a wide, deep null existing at the mid-line of each array microphone (Figure 15). A listening check revealed dramatic signal attenuation in both the horizontal and vertical planes of the microphones' midlines. The conclusion that can be drawn from this result is that the $0^0/90^0$ listening condition provides a contrived advantage for the array microphones used in this study.

This type of test arrangement would be a good one to use to demonstrate array superiority in conditions where noise arises from a single source overhead -- for example, overhead ventilation noise in an office. If it weren't for the contrived array performance, it could be said that the $0^0/90^0$ condition produced similar results to those produced in a diffuse noise field (live condition). Thus, if one wanted to use a single noise source to test the performance of microphones having various degrees of directivity, this might be a viable (and less expensive) option, *provided that the microphones tested where known to be devoid of midline nulls in the vertical plane.*

Omnidirectional Performance: Some Contradictory Results

Although not an initial focus of the study, we decided to formally investigate subject performance when listening to omnidirectional microphone recordings across the four noise environments. Inspection of Figure 16 below reveals a 0.5 dB difference between the R-Space and Live conditions, but quite a noticeable 2 dB difference between the two IAC booth conditions with performance in the $0^0/180^0$ condition being the worst.

Figure 16. Field-referred SNRs required for HINT sentences to be repeated correctly 50% of the time for omnidirectional ITE microphones across four noise delivery environments. Means and standard deviations are shown.



The poorer performance in the $0^0/180^0$ as opposed to the $0^0/90^0$ condition was initially puzzling since, for the particular omnidirectional ITE microphones used in this study, <u>better</u> performance in IAC 180 condition was expected. Manufacturer's specifications (in situ AI-weighted polar plot) for the D-MIC™ showed that, on the average, it should provide more noise rejection from directly behind (1.5 dB) as compared to 90^0 in the horizontal plane (0 dB). Since Roberts and Schulein (1997) have shown that the calculated two-dimensional DI scores provide a reasonable approximation of true three-dimensional measures, we assumed that the omnidirectional microphone would show no noise rejection overhead. But, just the opposite occurred.

To shed some light on this finding, several measurements were performed. First, Syntrillium Cool Edit Pro 1.2™ (2001) multi-channel signal editing software was used to perform a spectral comparison of the noise tokens recorded through the omnidirectional microphones fitted to KEMAR and presented in the IAC booth. Identical noise segments were selected for analysis, the only difference being whether the noise segment was recorded from the loudspeaker behind (180^0) or above (90^0) KEMAR. A comparison of the relative RMS amplitudes for each of the two loudspeaker locations was made for both the right and left channels (ears). Results showed that the noise segment presented in the overhead condition averaged approximately 1 dB softer than that presented from behind, for both channels. Figure 17 provides a visual comparison of the differences between the right channel noise spectra for the two presentation azimuths.

Second, *in situ* anechoic polar frequency responses were performed on the right omnidirectional microphone using a chirp stimulus and with the test loudspeaker positioned 24" (same distance used for calibration and recording) above (90^0) and behind (180^0) KEMAR. Two frequency response curves were obtained with all microphones in place, including the BTE microphones whose recordings were not used for this study. AI-weights from the Speech Intelligibility Index (ANSI S3.5-1997) were applied at 1/3-octave center frequencies to the frequency responses. The Resulting AI-weighted frequency responses seen for each loudspeaker condition (Figure 18) indicated that performance should have been be about 1 dB better in the overhead condition versus behind noise condition. Recall that the omnidirectional scores for the IAC 180 and IAC 90 conditions were -4.1 and -6.1 dB respectively. Since the investigation above revealed a 1dB softer presentation level for the overhead condition as compared to the behind condition, 1 dB of the 2 dB discrepancy is now accounted for. The additional 1 dB difference is felt to be due to test-retest variability or some other unaccounted for factor.

Figure 17. Comparison of spectra for omnidirectional ITE noise tokens as recorded through KEMAR at 90 versus 180 degrees.

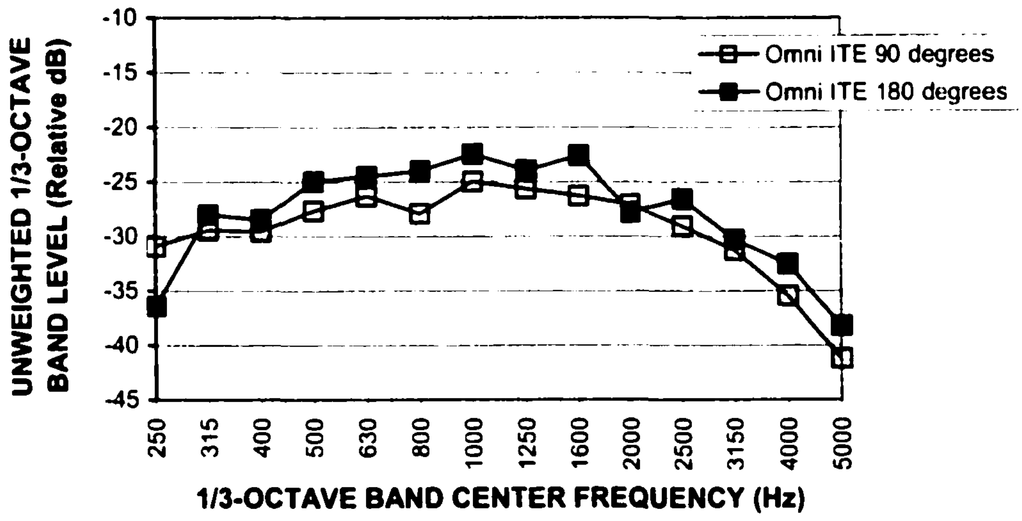
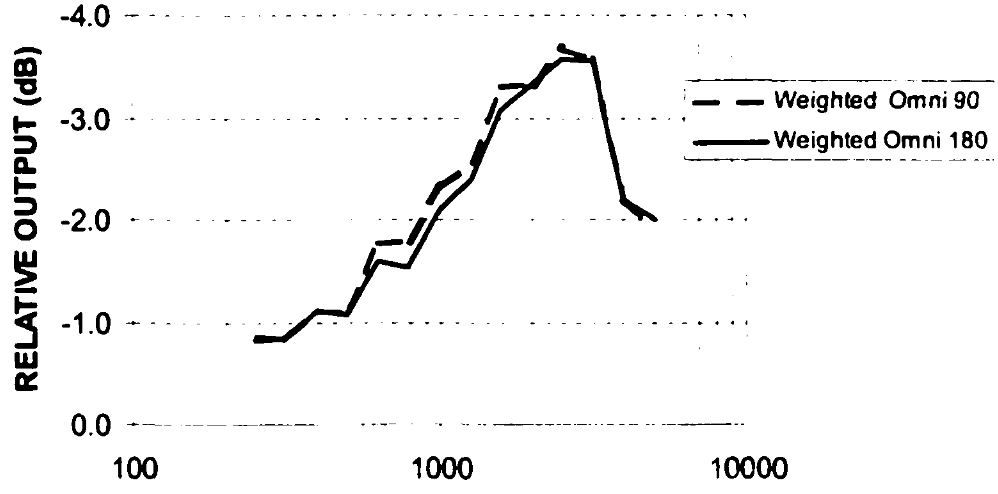


Figure 18. AI-weighted one-third octave band polar responses for the right omnidirectional ITE microphone at 90⁰ and 180⁰ (IAC booth).



A Flaw and Some Conjecture: Why the Array Microphones Did Not Fair as Well as They Might Have as Compared to the D-MICs™

It was assumed that all microphones were matched in their frontal frequency responses. After the recording process, this was found not to be the case. Re-recording was not carried out since it would have been too expensive and time-consuming. It was theorized that although the high-pass response might possibly compromise the array performance, performance would nonetheless be consistent across noise conditions. To ensure audibility of all stimuli (a problem in the pilot study which used subjects with hearing loss) subjects with normal hearing were employed. During the mixing process, a 1000Hz calibration tone was recorded which represented the –3 dB VU freq peaks of the HINT sentences as recorded via the ITE omnidirectional microphone track. Playback of the other microphone tracks revealed that it also represented generally the same level for the other microphone tracks since the HINT noise levels were matched for all microphone tracks. However, the array sentence tracks were somewhat different from the rest, because of their high frequency emphasis. Frequently, the VU peaks were lower than -3 dB, where peak speech levels occurred mostly in the low frequency phonemes. However, higher peaks (as high as +3 VU) occurred where high frequency phonemes predominated. It is possible that the lower VU peaks for the array sentences, coupled with the better than expected directivity of the D-MICs may help to explain the relatively poorer performance than expected for the array microphones. In spite of this, as mentioned previously, subject performance was similar to that obtained by Soede (1990) with normal listeners. In his study, SNRs of normal listeners improved about 5 dB with the array microphone compared to performance with the omnidirectional microphone (monaural study). In this study, the improvement was about 6 dB in the simulator and live conditions. The additional 1 dB here is probably due to binaural listening.

SUMMARY AND CONCLUSIONS

The purpose of this investigation was to test the hypothesis that the real-world performance of hearing aid microphones with varying amounts of directivity can be predicted using a simulation technique. Further, it was theorized that that the simulation technique would provide more accurate information about performance in noisy, real world environment than would the more conventional techniques of evaluation: Speech presented from the front and noise competing either (1) from a single loudspeaker behind or (2) from a single loudspeaker overhead.

To test these hypotheses, binaural master recordings of live restaurant noise were made through omnidirectional, super-cardioid, and array microphones affixed to KEMAR, while, simultaneously, a separate recording of the live restaurant noise was made using several shotgun microphones surrounding KEMAR. The latter recording was later played back through multiple loudspeakers of a simulator designed to simulate the acoustics of the live restaurant noise, while, simultaneously, the played-back noise was recorded binaurally using the same KEMAR manikin (fitted with the same hearing aid microphones) situated in the middle of the simulator's loudspeakers. To create the two conventional test situations, one channel (from the overhead calibration microphone) of the restaurant noise recording also was played back (on separate occasions) through a single loudspeaker placed overhead (IAC 90^0) and behind (IAC 180^0), again, using the same KEMAR manikin, fitted with the same microphones in an IAC booth. To provide the target stimulus for all four test conditions, binaural KEMAR recordings of HINT sentences were made by playing a recording of the sentences through a single loudspeaker located in the same IAC booth in front (0^0) of the manikin.

Stereo recordings of both KEMAR (1) noise recordings (made in the restaurant, the simulator, and in the two loudspeakers positions in the IAC booth) and (2) HINT sentence recordings were then mixed and presented, under insert earphones, to subjects with normal hearing.

Using the adaptive HINT procedure, sentence reception thresholds were obtained for the three microphones in the four noise environments. Result showed essentially identical performance for all hearing aid microphones in both diffuse noise environments (simulator and live conditions). The rank ordering of hearing aid microphone performance in the IAC 90^0 condition was similar to the diffuse environments, with the exception of significantly better performance for the array microphones. Performance in the IAC 180^0 condition compared to the other three conditions was slightly worse for the omnidirectional microphones, similar for the super-cardiod microphones (D-MIC) and significantly worse for the array microphones.

Several conclusions follow from these results. If one's goal is to assess hearing aid performance under realistic conditions where speech is directly in front (0^0) of the listener and ambient noise is more or less diffuse and surrounds the listener (e.g. in a restaurant or other social situation), then the results of this investigation demonstrate the simulator technique comes closer to imitating such a situation than do the other conditions – and does so using real, unaltered restaurant noise. The efficacy of assessing hearing aid performance by placing the target speech signal directly in front (0^0) of the listener and using a single competing noise source either behind (180^0) or directly above (90^0) is unrealistic. Most listening situations are characterized by the presence of multiple noise sources at various azimuths. A single noise source placed either directly behind or above the listener can produce a contrived performance advantage or deficit, depending on the polar pattern of the microphone. The key finding of this study is that the major strength of the simulator system is that a realistic estimate of the in-noise performance of any microphone can be assessed without having detailed information about its polar pattern. While the results of this investigation are promising in terms of their applications to clinical and research work in the area of aided speech perception in real rooms, several weaknesses are present that should be addressed in future research.

First, because the noise recordings employed in the simulation technique are specific to a particular restaurant, the findings of this investigation cannot be generalized to other restaurants. In order to assess hearing aid microphone performance in other restaurants (as well as in other noisy settings), additional noise recordings would be needed. This is not an easy task. The preparation of the noise tokens was extremely difficult and time-consuming. Future research is needed to develop a more efficient means of developing alternative listening environments. This research might include the sampling of a variety of restaurants to determine the characteristics of the "average" quiet, semi-quiet, or noisy restaurants. A recorded collection of various restaurants could then be produced (e.g. "Average noisy neighborhood chain restaurant"; "Quietly elegant formal restaurant"; etc.) Validation procedures could be carried out that show how various microphones perform in these restaurants and correction factors could be developed that would necessitate testing only in one restaurant setting.

Second, additional validation procedures are needed if alternative speech materials are to be used. The frequency importance information of the HINT is more low frequency dependent, for example than that of nonsense syllables. Thus, repeating the experiment with nonsense syllables might yield a different pattern of results.

Third, the sentence recordings were made well within the critical distance, thus ruling out the effects of reverberation on speech perception in noise. Additional study is needed to determine the pattern of test results obtained in other environments characterized by increased reverberation time (e.g. places if worship, lecture halls).

Fourth, the investigation used subjects with normal hearing. Additional study using subjects with sensorineural hearing loss is needed to determine if test results would yield the same patterns.

Fifth, it would be interesting to know if the array microphones would have performed better had their front frequency responses been matched to those of the other microphones. While this variable was consistent across noise environments and thus yielded the same degree of performance in each environment, an increase in low frequency information may have resulted in increased net benefit from the array microphones versus that of the others.

In summary, although the results of this study cannot easily generalized, to other noisy situations, microphones, or listeners with hearing loss, the effort nevertheless makes an important scientific contribution because it shows that It is possible to obtain speech recognition performance that approximates real world performance by carefully recording real noise and reproducing it in the lab.

APPENDIX A

PILOT STUDY:
Hearing Aid Performance as a Function Of Hearing Aid Microphone and Test Environment

OBJECTIVE

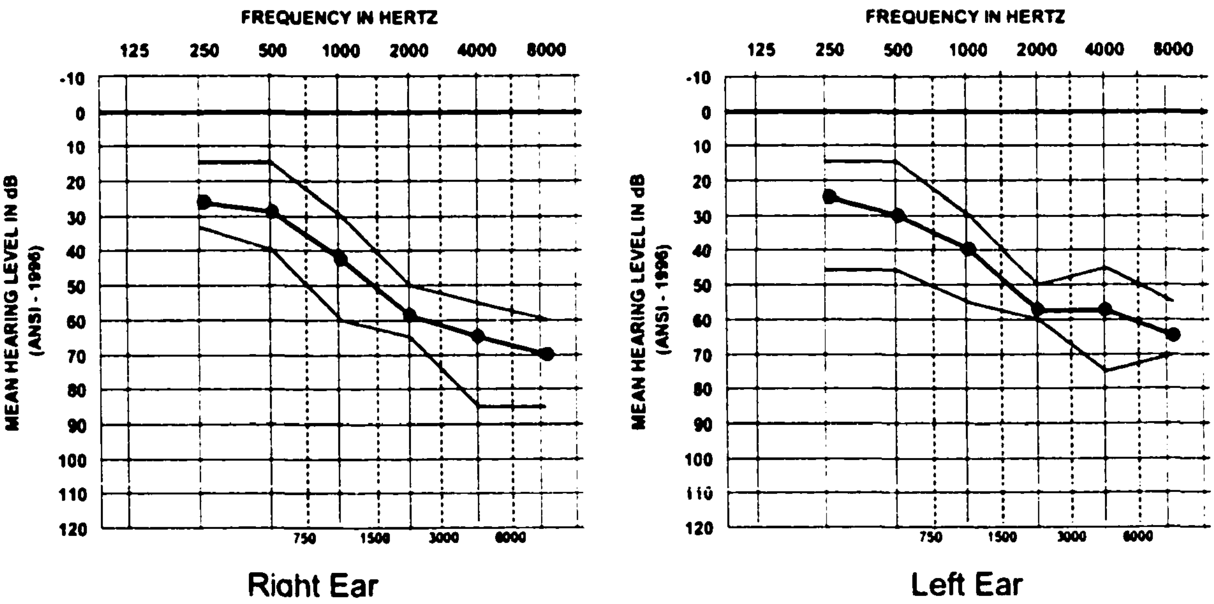
Using binaural recordings obtained on a Knowles Electronics Manikin for Auditory Research (KEMAR), the purpose of this study was to compare the performance of two hearing aids, with vastly different levels of directivity, in two test environments. The first microphone was omnidirectional and was encased in an ITE shell whereas the second microphone was an array prototype, fixed horizontally onto the KEMAR's temporal lobe and powered by an external pre-amplifier. The first environment was that of a real restaurant where the restaurant noise was spatially separated around the KEMAR. This environment was included as an attempt to determine whether a more realistic listening situation could be used to evaluate hearing aid performance, and thus provide more real world validity in the process. The second environment consisted of an IAC booth where the restaurant noise "heard" in the first environment was recorded and played back to the KEMAR from a single loudspeaker located at a 180^0 azimuth. This testing environment has been traditionally used in the clinic for measuring the performance of directional hearing aid microphones (DMHAs), even though the noise is not spatially separated and is located unrealistically directly behind the listener. Although additional microphones and conditions will be used in the main study, the pilot study was carried out to test the soundness of the stimuli and methodology.

METHOD

Subjects

Four experienced hearing aid users with symmetrical mild to moderately sloping sensorineural hearing loss and absence of middle ear pathology served as listeners for this study. Figure 1 shows the mean and range of audiometric thresholds for the right and left ears of the experimental group.

Figure 1. Mean and range of audiometric thresholds for the left and right ears of the four pilot subjects.



Stimuli, instrumentation, calibration, and data collection procedures are discussed in detail the dissertation and thus will not be addressed here.

DESIGN

The experimental design consisted of a 4 X 4 Hyper-Greco-Latin Square in which each 20-sentence HINT list (1, 2, 3, and 4) is paired with a noise sample (a, b, c, and d) for each of the four conditions listed in the first row of Table1 below. Each sentence/noise combination is unique. The twelve test tokens shown in Table 1 and used in the current pilot study will form a portion of the entire 12 X 12 design described in the dissertation.

Table 1. Hyper-Greco Latin Square Design: For each of the four conditions listed in the first row, each 20-sentence list is paired with a noise sample.

SUBJECT	Omni 180^0	Omni "Live"	Array 180^0	Omni "Live"
A	1a	2b	3c	4d
B	2d	1c	4b	3a
C	3b	4a	1d	2c
D	4c	3d	2a	1b

A fixed effects analysis of variance (with subject as a factor) was carried out. The independent variables are microphone type (2 levels: omnidirectional and array) and test environment (2 levels: IAC booth with a single noise source at the rear (180^0) and live restaurant with multiple noise sources). The single dependent variable is SNR for 100% correct repetition of a HINT sentence 50% of the time.

RESULTS

Mean HINT SNRs were obtained for the four subjects and can be seen in Table 2. The HINT thresholds were determined by noting the audiometer dial setting (in dB HL) where each subject obtained his or her HINT threshold.

Table 2. Mean SNRs obtained using recordings of HINT sentences presented along with (1) restaurant noise in IAC booth (single noise; 180^0) and in (2) live restaurant. SNRs are based upon audiometer attenuator settings (dB HL).

Audiometer SNRs (dB HL)				
	Omni ITE Rear	Omni ITE Live	Array Rear	Array Live
Subject 1	3.41	1.65	-3.18	-6.24
Subject 2	5.88	2.00	-2.82	-3.76
Subject 3	2.00	-0.35	-7.88	-7.53
Subject 4	0.71	-0.12	-8.24	-7.06
Mean	3.00	0.80	-5.53	-6.15

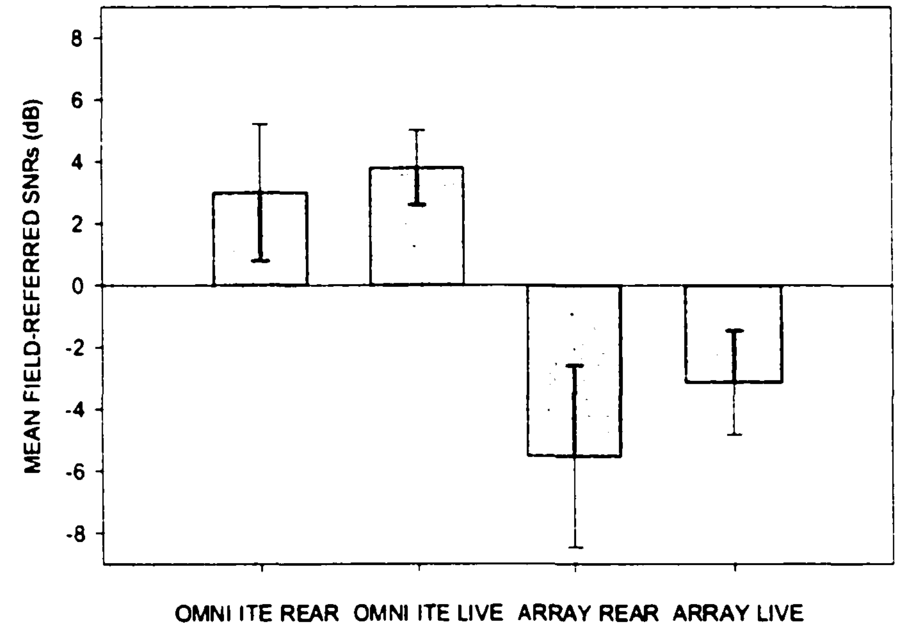
Prior to statistical analysis, correction factors were applied to each subject's HINT score to account for the fact that the calibration signal was measured to be 3 dB higher (on the VU meter) at the KEMAR reference point for both microphone conditions in the live restaurant setting. The corrected scores shown in Table 2 provide accurate relative field SNR data across conditions.

Table 3. Relative free-field SNRs obtained using recordings of HINT sentences presented along with (1) restaurant noise in IAC booth (single noise source; 180^0) and in (2) a live restaurant.

Relative Free-Field SNRs				
	Omni ITE Rear	Omni ITE "Live"	Array Rear	Array "Live"
Subject 1	3.41	4.65	-3.18	-3.24
Subject 2	5.88	5.00	-2.82	-0.76
Subject 3	2.00	2.65	-7.88	-4.53
Subject 4	0.71	2.88	-8.24	-4.06
Mean	3.00	3.80	-5.53	-3.15
Correction Factor Applied to Each Cell	0	3	0	3

Figure 2 shows the field-referred SNRs required for sentences to be repeated correctly, 50% of the time, across the three hearing aid microphone conditions and for the two noise-delivery environments. Means and standard deviations for each microphone condition can be seen.

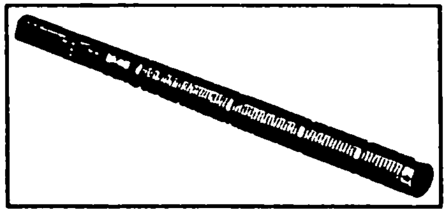
Figure 2. Field-referred signal-to-noise ratios (SNRs) required for sentences to be repeated correctly, 50% of the time, across three hearing aid microphone conditions and for two noise-delivery environments. Means and standard deviations are shown.

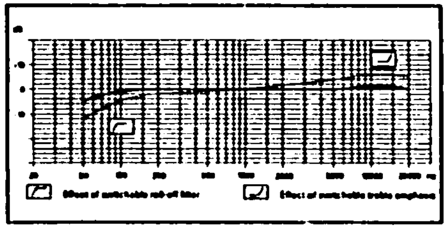


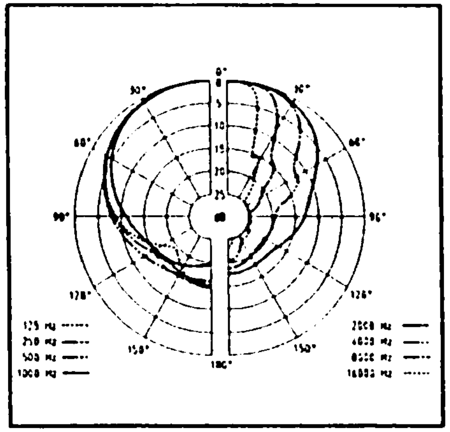
Fixed effects ANOVA (repeated measures with subject as a factor) revealed a main effect ($p = 0.0006$) for microphone with some evidence of main effects for environment ($p = 0.048$) as well as subject ($p = 0.025$). No interactions were noted.

DISCUSSION/CONCLUSION

Results indicate that the research design is sufficiently sensitive to be used for the main study. It should be noted that although the array overhead condition was not a part of the pilot study, data was collected using that condition. Preliminary results show that the noise tracks were at least 20 dB below the 0 dB VU level set for the audiometer, due to the directional treatment effects of the array microphone. This resulted in the subjects hearing the HINT sentences only at low levels, thus resulting in a threshold for sentence repetition only, not performance at a specific SNR. This occurred because the subjects had sloping hearing losses and frequency shaping was not used. In addition, the array microphone had high pass frontal frequency response, compared to the other microphones in the study, which were more flat in their frontal frequency responses. This difference caused the speech stimuli (as well as some of the noise conditions) to be presented at intensities reduced from where they would have been had all of the microphones had the similar frontal frequency responses. Since the investigator is interested in the relative performance among microphones and environments, no changes will be made to the array microphones. However, normal hearing subjects will be used in place of subjects with hearing loss in the main study to avoid the possibility of the subjects not hearing the stimuli recorded through the array microphones.







MKH 70 RF Condenser Microphone

The MKH 70 is a lightweight long gun microphone. Its excellent directivity is particularly suited to applications undertaken in difficult conditions, such as high background noise and distance microphone positioning. Its frequency-independent directivity prevents sound coloration from off-axis sound sources.

Features

- Exceptionally low inherent self-noise
- Transformerless and fully floating balanced output
- Infra-sonic cut-off filter
- Symmetrical transducer technology ensures extremely low distortion
- Switchable pre-attenuation, switchable roll-off filter and switchable treble emphasis
- Rugged and weather-proof
- Black, anodised light metal body
- RF phase discriminator circuit provides immunity to humidity

Technical Data

Pick-up pattern	super-cardioid/lobar
Frequency response	50–20,000 Hz
Sensitivity (free field, no load) (1 kHz)	50 (15) mV/Pa
Nominal impedance	150 Ω
Min. terminating impedance	1000 Ω
Equivalent noise level	
A-weighted (DIN IEC 651)	5 (13) dB
CCIR-weighted (CCIR 468-3)	16 (24) dB
Max. sound pressure level	124 (132) dB at 1 kHz
Power supply	phantom 48 ± 4 V
Supply current	2 mA
Dimensions in inches	.98 x 16.14
Weight	6.34 oz

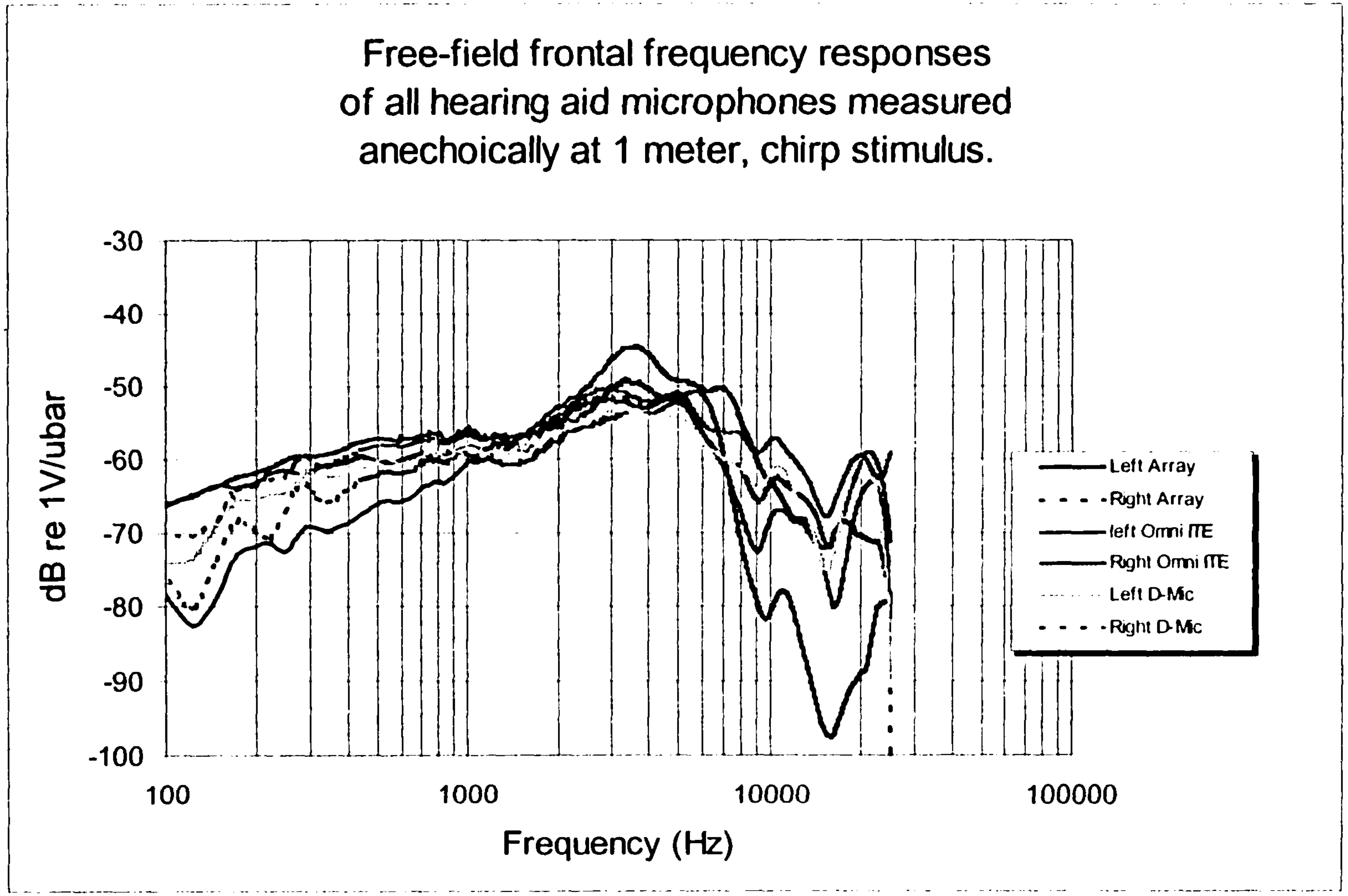
Optional Accessories

Shock mount	MZS 40
Stand adapter	MZQ 40
Windscreen	MZQW 40
Phantom battery power supply unit	MZA 14 P 48 U
Foam windshield	MZW 41
Shock mount	MZS 20-1
Basket windshield	MZW 20-1
Long hair windcoat	MZH 20-1
AC powered phantom supply for 2 mics	N 481-2

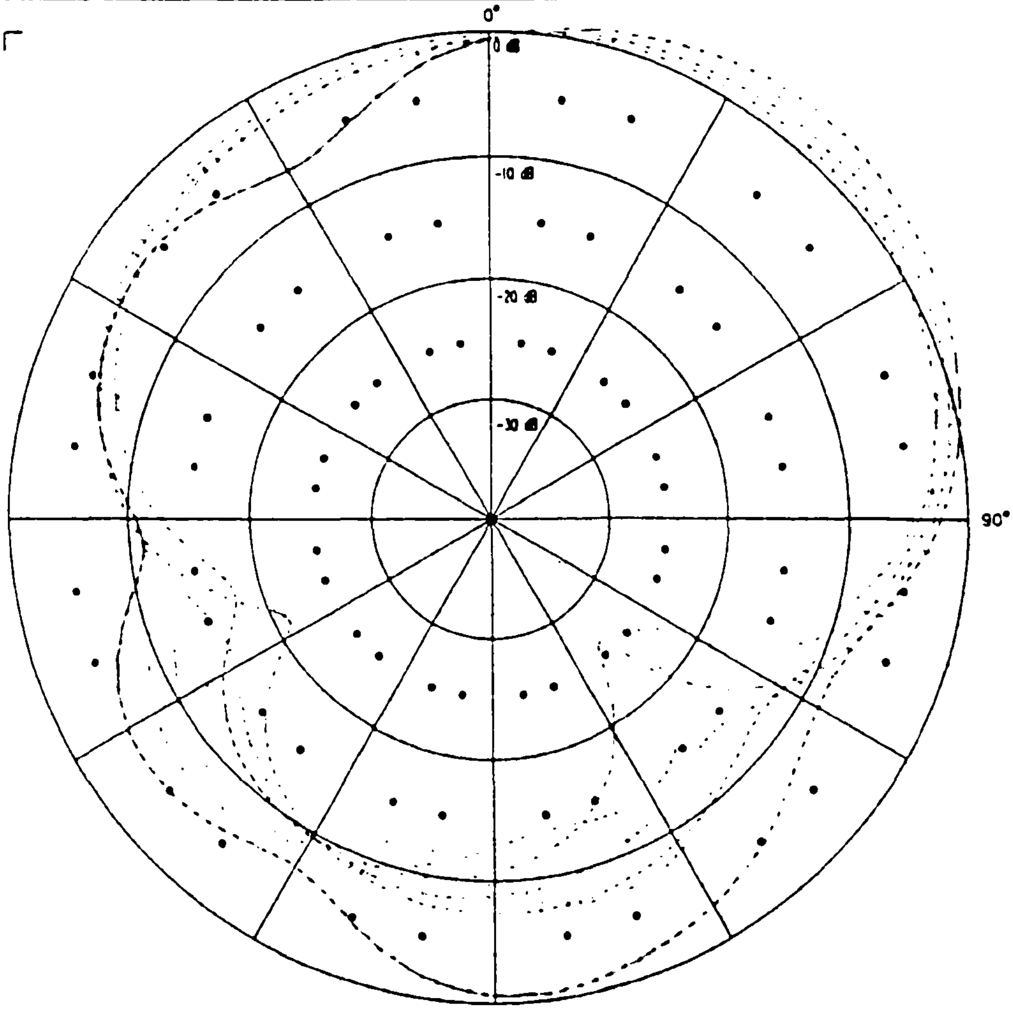
Architect's Specifications

The unit shall be a super-cardioid/lobar (long gun) interference tube microphone with infrasonic cut-off filter, switchable pre-attenuation, switchable roll-off filter and switchable treble emphasis. The frequency response shall be 50–20,000 Hz, and sensitivity (free field, no load) of 50 (15) mV/Pa at 1 kHz. The nominal impedance shall be 150 Ω, and min. terminating impedance of 1000 Ω. The dimensions shall be .98 x 16.14 inches, and weight 6.34 oz. The unit shall be a Sennheiser MKH 70.

Values in parentheses with attenuator switched on (–10 dB).


Free-field frontal frequency responses
of all hearing aid microphones measured
anechoically at 1 meter, chirp stimulus.
dB re 1V/ubar
-30
-40
-50
-60
-70
-80
-90
-100
100
1000
10000
100000
Frequency (Hz)
Left Array
Right Array
left Omni ITE
Right Omni ITE
Left D-Mic
Right D-Mic

ETYMŌTIC RESEARCH

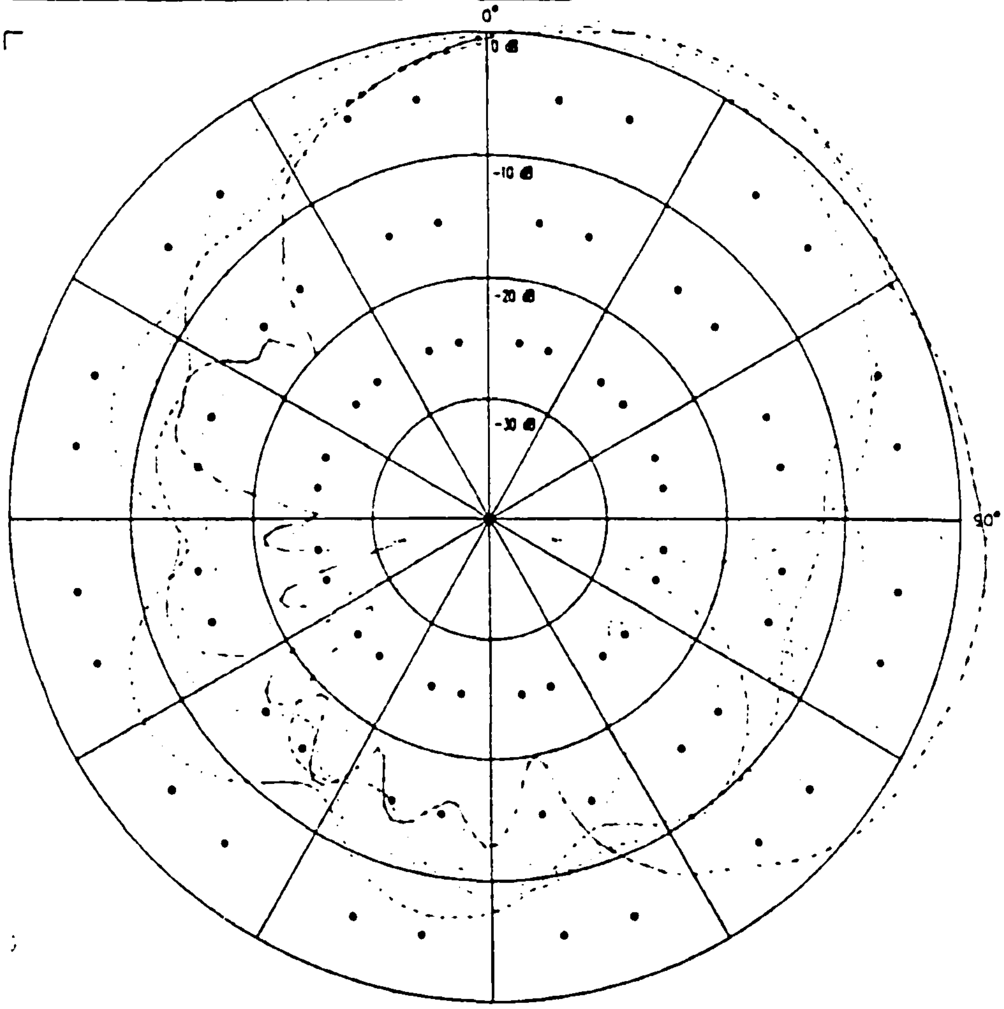


CURVE #5 POLAR CHARACTERISTICS (10 dB/Major Division)

FREQUENCY:	500.	[illegible]30.	800.	[illegible]000.	[illegible].	[illegible]600	Hz.	DATE: [illegible] Aug 2000
SENSITIVITY: (dB re 1V/Pa)	29.[illegible]	23.[illegible]	22.[illegible]	23.[illegible]	2[illegible].9.	2[illegible]	dB	TIME: [illegible]: 30: 43
S/N RATIO:	[illegible]	8.	8.	8.	[illegible]9.	[illegible]	dB	INITIALS: [illegible]G
DIRECTIVITY INDEX:	4.4.	[illegible].5.	4.2.	3.6.	4.2.	2.6	dB	DIRECTION: CCW

ETYMŌTIC RESEARCH, 61 Martin Lane, Elk Grove Village, Illinois 60007 (847) 228-0006
Etymotic (et-im-OH-tik) is a "new ancient Greek word" which means true to the ear. Fax (847) 228-6836

ETYMŌTIC RESEARCH



CURVE #6 POLAR CHARACTERISTICS (10 dB/Major Division)

FREQUENCY: 2000, 2 00, 3150, 000, 5000, 6300 Hz. DATE: 17 Aug 2000

SENSITIVITY: (dB re 1V/Pa) - . .3, .2 . 3 . 9. . .5, -23. dB TIME: 13: 03: 56

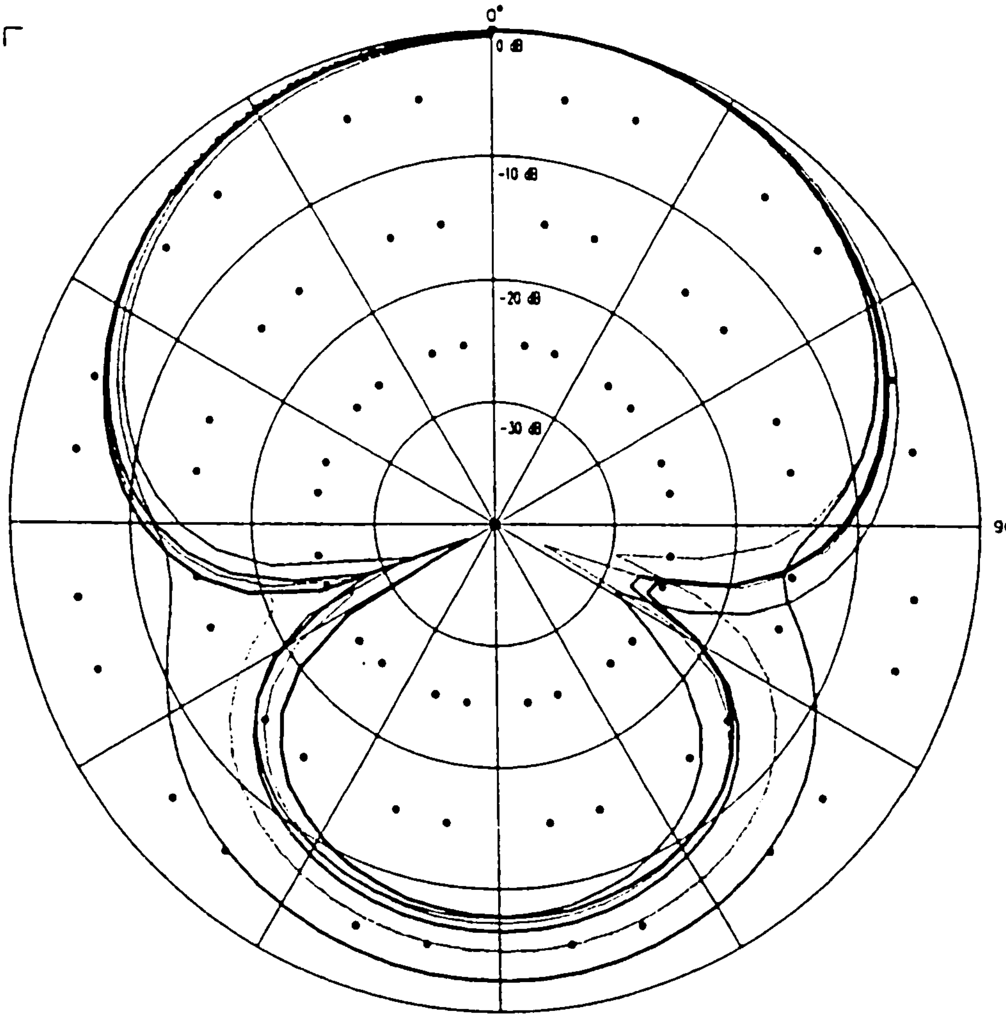
S/N RATIO: 5 . 5 . 58. 5 . . 8 dB INITIALS: DMG

DIRECTIVITY INDEX: 6.3, 6 . . 5.8, 5 . . 3. 2. ; dB DIRECTION: CCW

ETYMŌTIC RESEARCH, 61 Martin Lane, Elk Grove Village, Illinois 60007 (847) 228-0006

Etymotic (et-im-OH-tik) is a "new ancient Greek word" which means true to the ear. Fax (847) 228-6836

ETYMŌTIC RESEARCH

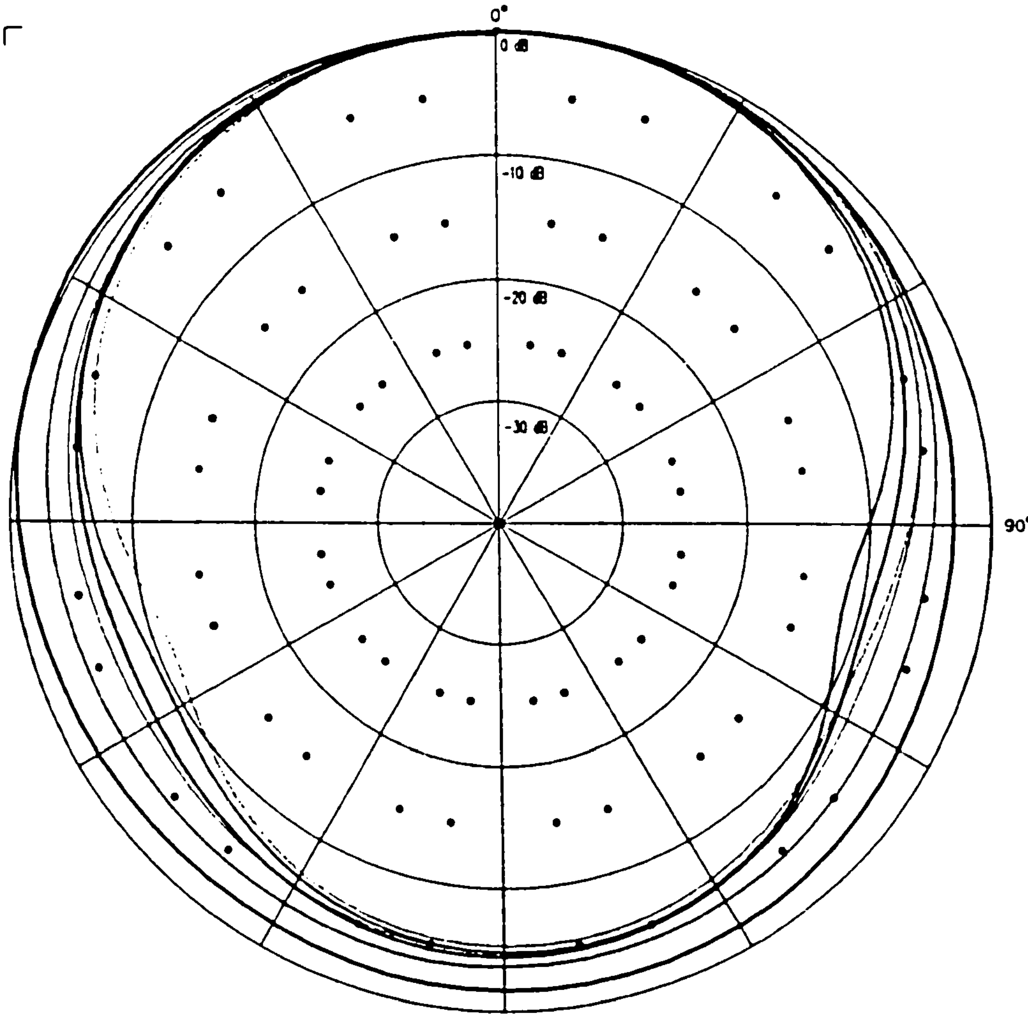


CURVE #9 POLAR CHARACTERISTICS (10 dB/Major Division)

MICROPHONE: RT. ITE DIR. FF (Right D-mic free-field) INPUT: 90 dB SPL
FREQUENCY: 1600, 1250, 1000, 800, 630, 500 Hz. DATE: 20 Jul 2001
SENSITIVITY: (dB re 1V/μBAR) -28.8, -29.4, -29.1, -29.9, -29.9, -31.2 dB TIME: 10: 54: 08
S/N RATIO: 41, 40, 41, 40, 39, 38 dB INITIALS: DEG
DIRECTIVITY INDEX: 5.3, 6.0, 6.1, 6.1, 6.3, 6.2 dB DIRECTION: CCW

ETYMŌTIC RESEARCH, 61 Martin Lane, Elk Grove Village, Illinois 60007 (847) 228-0006
Etymotic (et-im-OH-tik) is a "new ancient Greek word" which means *true to the ear* Fax (847) 228-6836

ETYMŌTIC RESEARCH

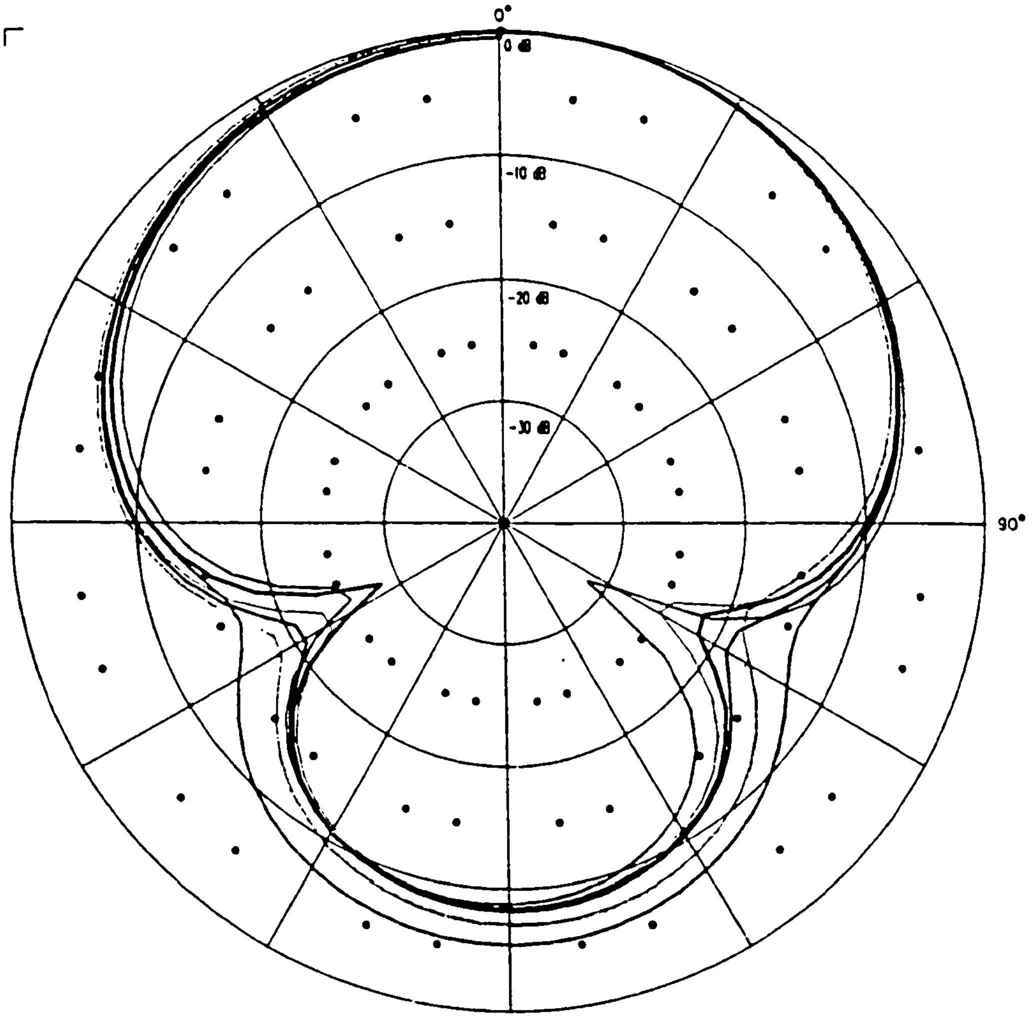


CURVE #10 POLAR CHARACTERISTICS (10 dB/Major Division)

MICROPHONE: RT. ITE DIR. FF (Right D-mic free field) INPUT: 90 dB SPL
FREQUENCY: 2000, 2500, 3150, 4000, 5000, 6300 Hz. DATE: 20 Jul 2001
SENSITIVITY: (dB re 1V/Pa): -28.0, -28.0, -30.9, -32.9, -36.2, -38.6 dB TIME: 11: 04: 59
S/N RATIO: 42, 42, 39, 37, 33, 31 dB INITIALS: DEG
DIRECTIVITY INDEX: 4.9, 4.8, 4.5, 4.0, 2.9, 1.3 dB DIRECTION: CCW

ETYMŌTIC RESEARCH, 61 Martin Lane, Elk Grove Village, Illinois 60007 (847) 228-0006
Etymotic (et-im-OH-tik) is a "new ancient Greek word" which means *true to the ear*. Fax (847) 228-6836

ETYMŌTIC RESEARCH

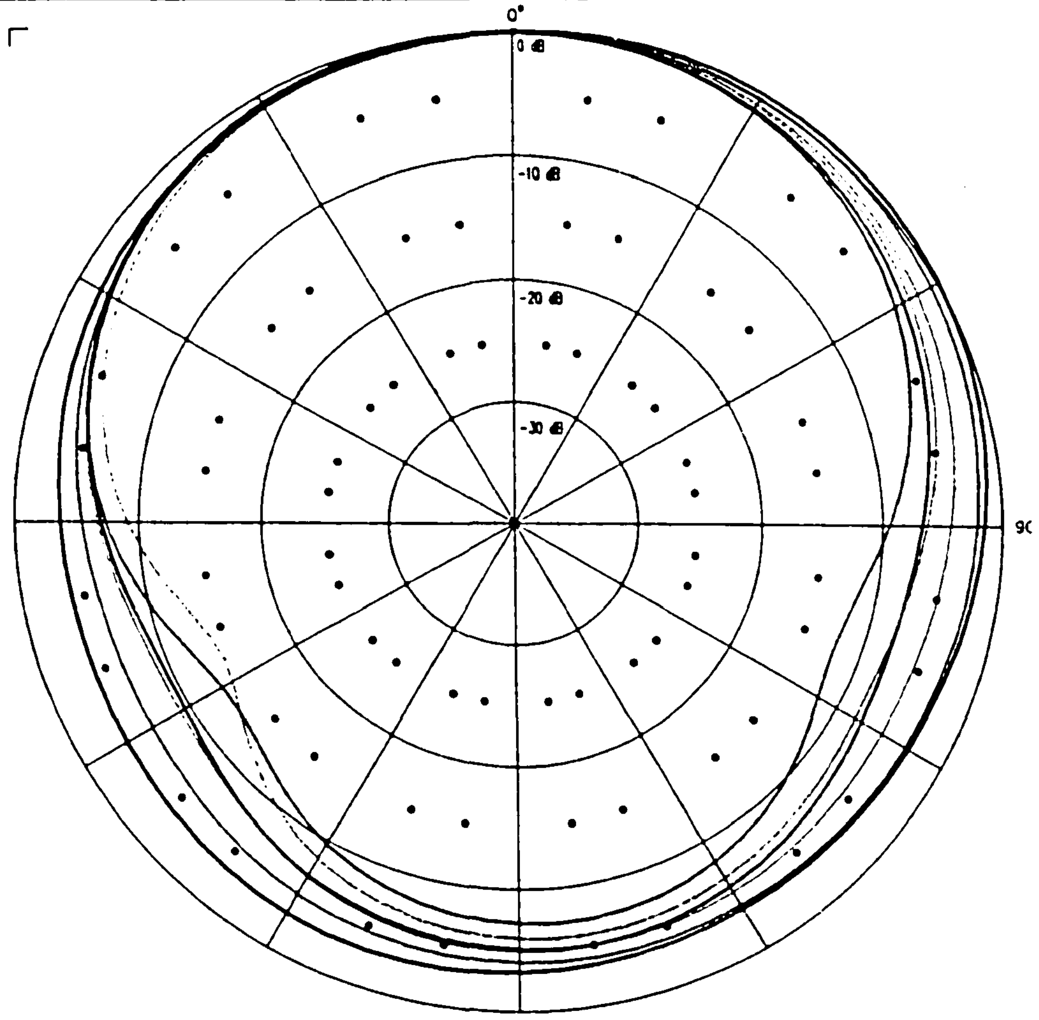


CURVE #11 POLAR CHARACTERISTICS (10 dB/Major Division)

MICROPHONE: LT. ITE DIR FF left D-mic free-field INPUT: 90 dB SPL
FREQUENCY: 1600, 1250, 1000, 800, 630, 500 Hz. DATE: 20 Jul 2001
SENSITIVITY: (dB re 1V/μBAR) -28.8, -29.7, -29.4, -30.3, -30.5, -31.9 dB TIME: 11: 38: 21
S/N RATIO: 41, 40, 40, 39, 39, 38 dB INITIALS: DEG
DIRECTIVITY INDEX: 5.7, 5.8, 5.8, 6.0, 6.3, 6.2 dB DIRECTION: CCW

ETYMŌTIC RESEARCH, 61 Martin Lane, Elk Grove Village, Illinois 60007 (847) 228-0006
Etymotic (et-im-OH-tik) is a "new ancient Greek word" which means *true to the ear.* Fax (847) 228-6836

ETYMŌTIC RESEARCH



CURVE #13 POLAR CHARACTERISTICS (10 dB/Major Division)

MICROPHONE: LT. ITE DIR FF left D-mic free-field INPUT: 90 dB SPL

FREQUENCY: 2000, 2500, 3150, 4000, 5000, 6300 Hz. DATE: 20 Jul 2001

SENSITIVITY: -27.8, -28.3, -31.2, -32.9, -35.8, -37.0 dB TIME: 12:51:00

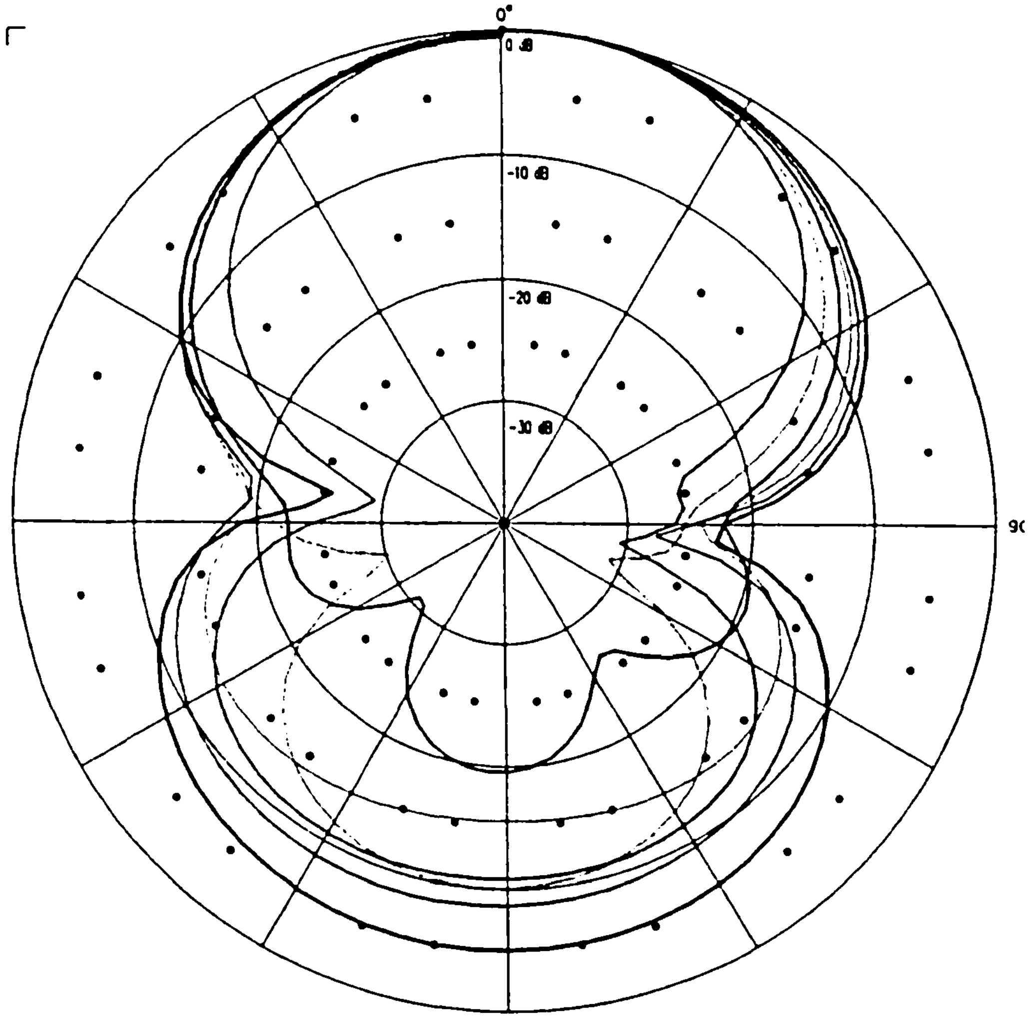
S/N RATIO: 42, 42, 39, 37, 34, 33 dB INITIALS: DEG

DIRECTIVITY INDEX: 5.2, 4.9, 4.5, 3.8, 2.7, 2.0 dB DIRECTION: CCW

ETYMŌTIC RESEARCH, 61 Martin Lane, Elk Grove Village, Illinois 60007 (847) 228-0006

Etymotic (et-im-OH-tik) is a "new ancient Greek word" which means true to the ear. Fax (847) 228-6836

ETYMŌTIC RESEARCH

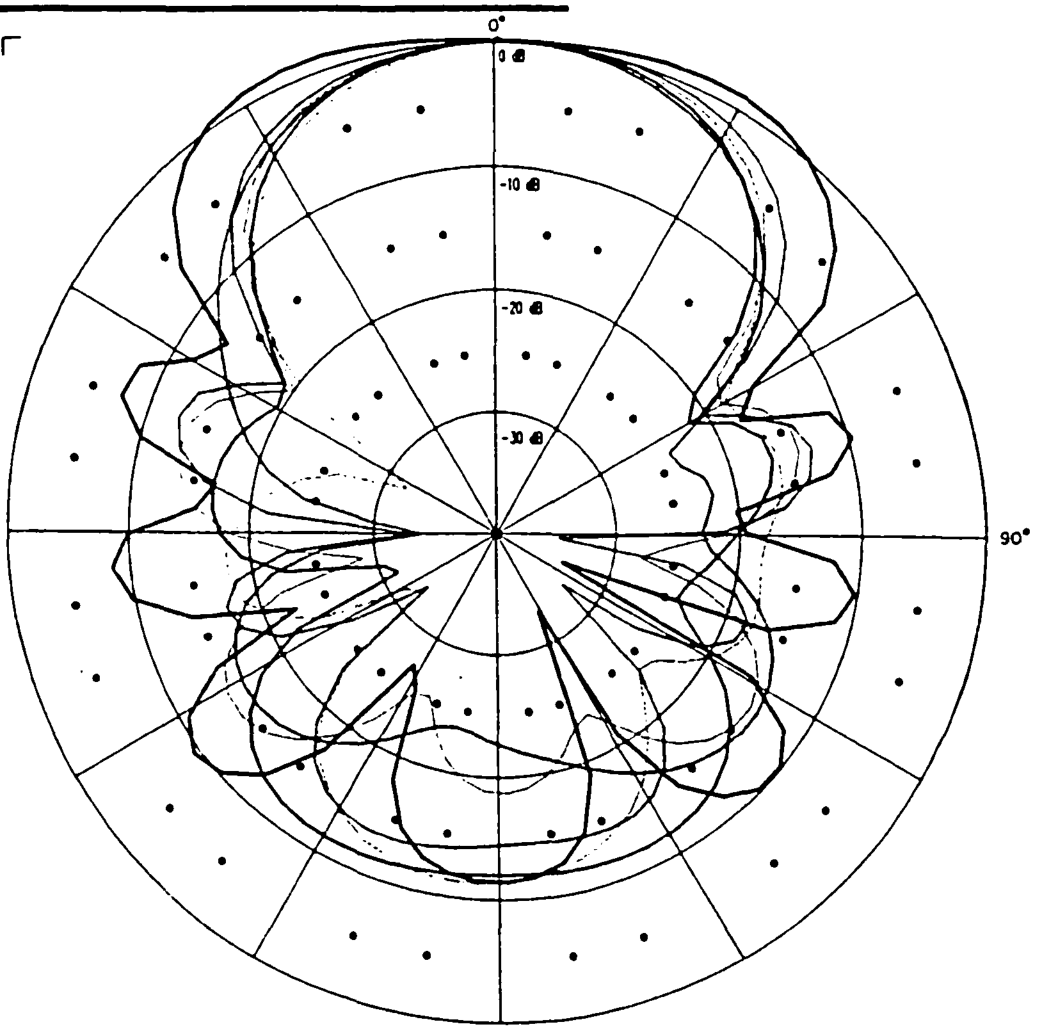


CURVE #16 POLAR CHARACTERISTICS (10 dB/Major Division)

MICROPHONE:	AT. ARRAY FF						INPUT:	90 dB SPL
FREQUENCY:	1600,	1250,	1000,	800,	630,	500 Hz.	DATE:	20 Jul 2001
SENSITIVITY: (dB re 1V/ubar)	-34.2,	-36.4,	-36.4,	-37.7,	-37.9,	-39.9 dB	TIME:	13: 57: 17
S/N RATIO:	36,	34,	34,	33,	32,	44 dB	INITIALS:	DEG
DIRECTIVITY INDEX:	9.8,	9.2,	9.3,	8.4,	7.8,	6.9 dB	DIRECTION:	CCW

ETYMŌTIC RESEARCH, 61 Martin Lane, Elk Grove Village, Illinois 60007 (847) 228-0006
Etymotic (et-im-OH-tik) is a "new ancient Greek word" which means true to the ear Fax (847) 228-6836

ETYMŌTIC RESEARCH



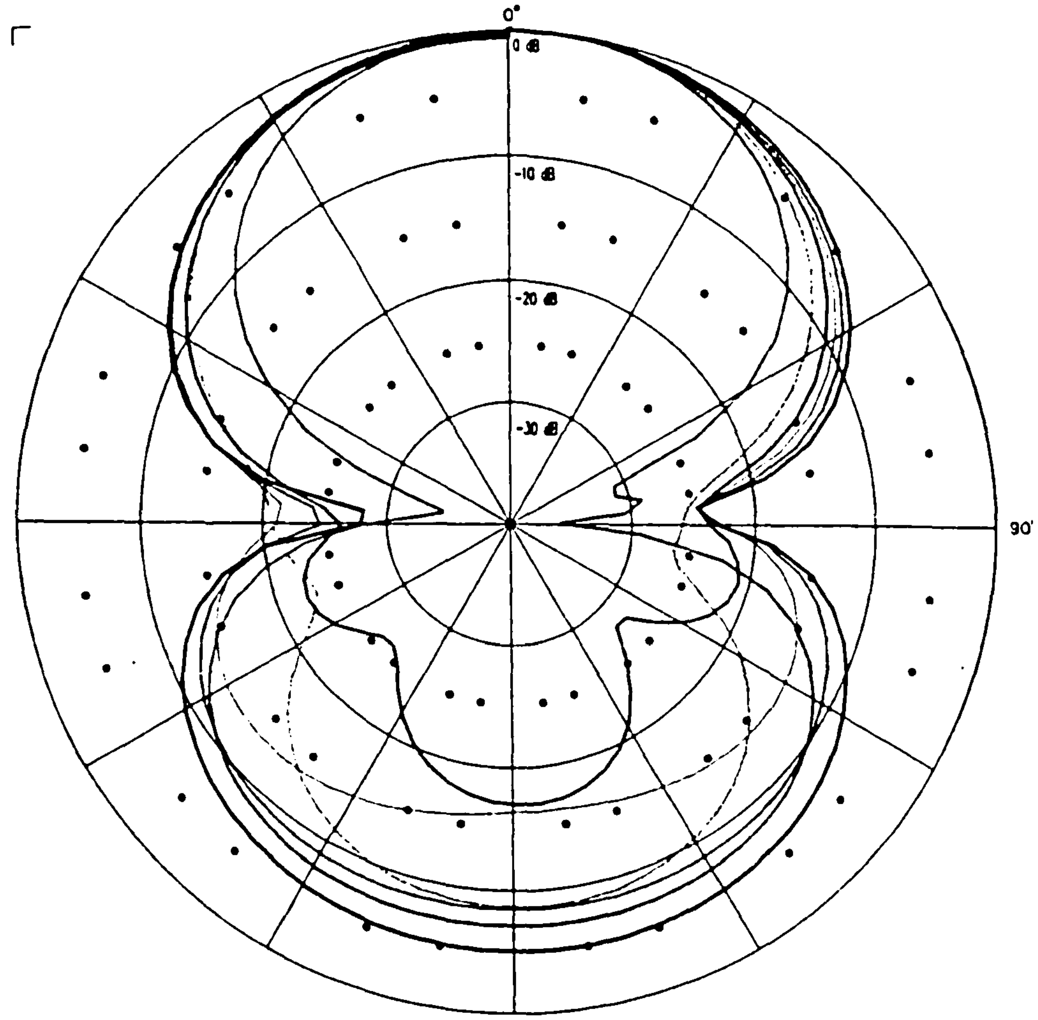
CURVE #17 POLAR CHARACTERISTICS (10 dB/Major Division)

MICROPHONE:	RT. ARRAY FF						INPUT:	90 dB SPL
FREQUENCY:	2000,	2500,	3150,	4000,	5000,	6300 Hz.	DATE:	20 Jul 2001
SENSITIVITY: (dB re 1V/μBAR)	-33.8,	-31.0,	-28.5,	-28.7,	-31.2,	-33.3 dB	TIME:	14: 11: 49
S/N RATIO:	36,	39,	42,	41,	39,	37 dB	INITIALS:	DEG
DIRECTIVITY INDEX:	10.7,	11.?,	10.7,	10.4,	9.9,	6.4 dB	DIRECTION:	CCW

ETYMŌTIC RESEARCH, 61 Martin Lane, Elk Grove Village, Illinois 60007 (847) 228-0006

Etymotic (et-im-OH-tik) is a "new ancient Greek word" which means *true to the ear* Fax (847) 228-6836

ETYMŌTIC RESEARCH

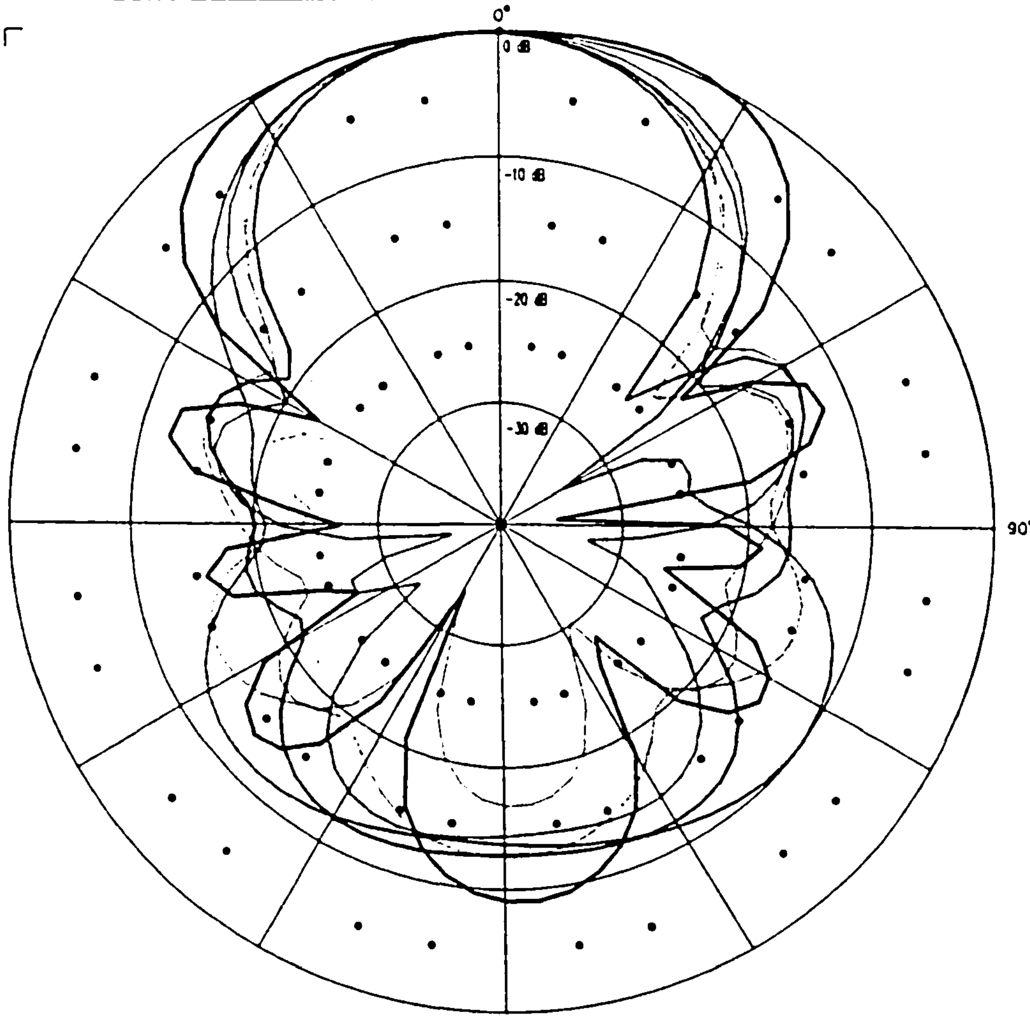


CURVE #18 POLAR CHARACTERISTICS (10 dB/Major Division)

MICROPHONE:	LT. ARRAY FF	INPUT:	90 dB SPL
FREQUENCY:	1600, 1250, 1000, 800, 630, 500 Hz.	DATE:	20 Jul 2001
SENSITIVITY: (dB re 1V/Pa)	-31.3, -34.2, -35.0, -37.0, -38.0, -40.2 dB	TIME:	14: 36: 36
S/N RATIO:	39, 36, 35, 33, 32, 44 dB	INITIALS:	DEG
DIRECTIVITY INDEX:	9.4, 9.1, 9.4, 8.5, 7.9, 7.0 dB	DIRECTION:	CCW

ETYMŌTIC RESEARCH, 61 Martin Lane, Elk Grove Village, Illinois 60007 (847) 228-0006
Etymotic (et-im-OH-tik) is a "new ancient Greek word" which means *true to the ear* Fax (847) 228-6836

ETYMŌTIC RESEARCH

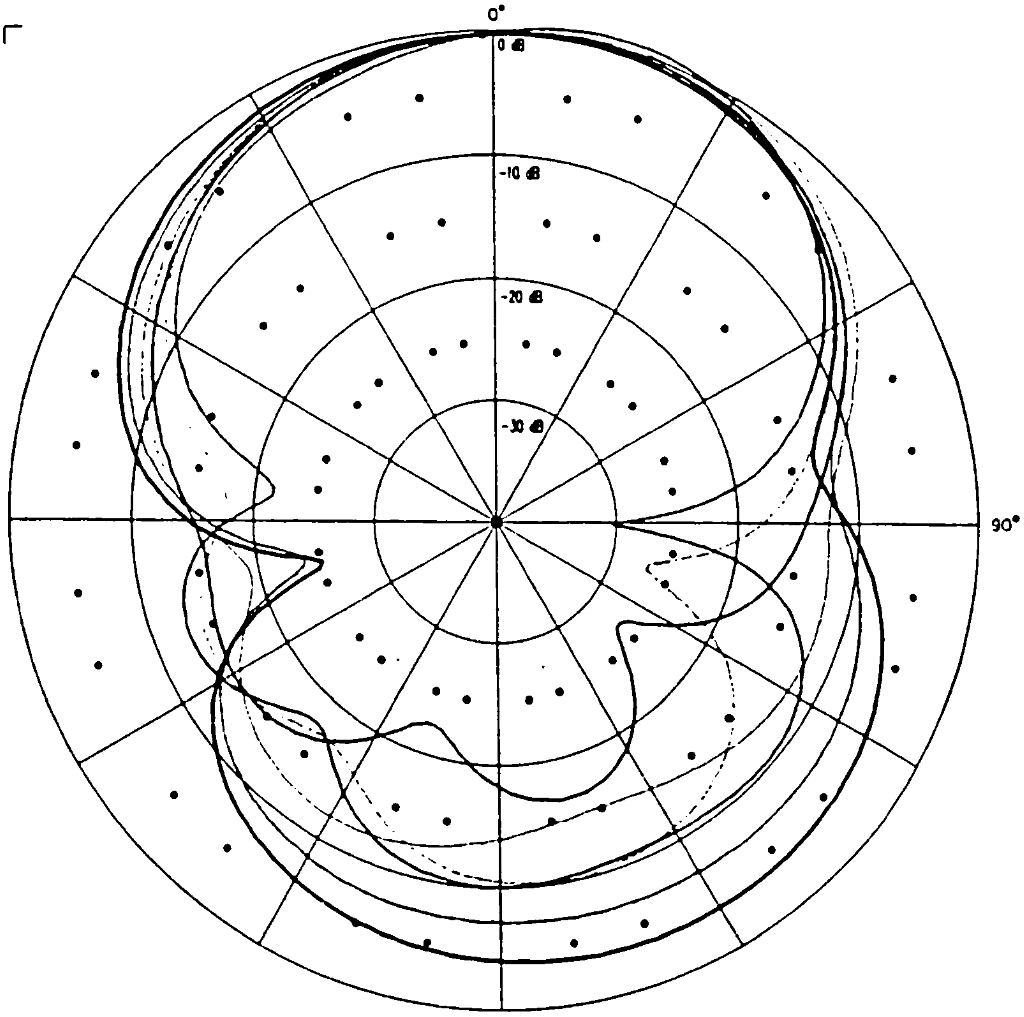


CURVE #21 POLAR CHARACTERISTICS (10 dB/Major Division)

MICROPHONE:	LT ARRAY FF						INPUT:	90 dB SPL
FREQUENCY:	2000,	2500,	3150,	4000,	5000,	6300 Hz.	DATE:	20 Jul 2001
SENSITIVITY: (dB re 1V/Pa)	-39.9,	-36.1,	-32.2,	-30.1,	-32.5,	-32.7 dB	TIME:	14: 51: 16
S/N RATIO:	43,	34,	38,	40,	38,	37 dB	INITIALS:	DEG
DIRECTIVITY INDEX:	10.0,	11.5,	11.4,	11.5,	10.9,	8.4 dB	DIRECTION:	CCW

ETYMŌTIC RESEARCH, 61 Martin Lane, Elk Grove Village, Illinois 60007 (847) 228-0006
Etymotic (et-im-OH-tik) is a "new ancient Greek word" which means *true to the ear* Fax (847) 228-6836

ETYMŌTIC RESEARCH

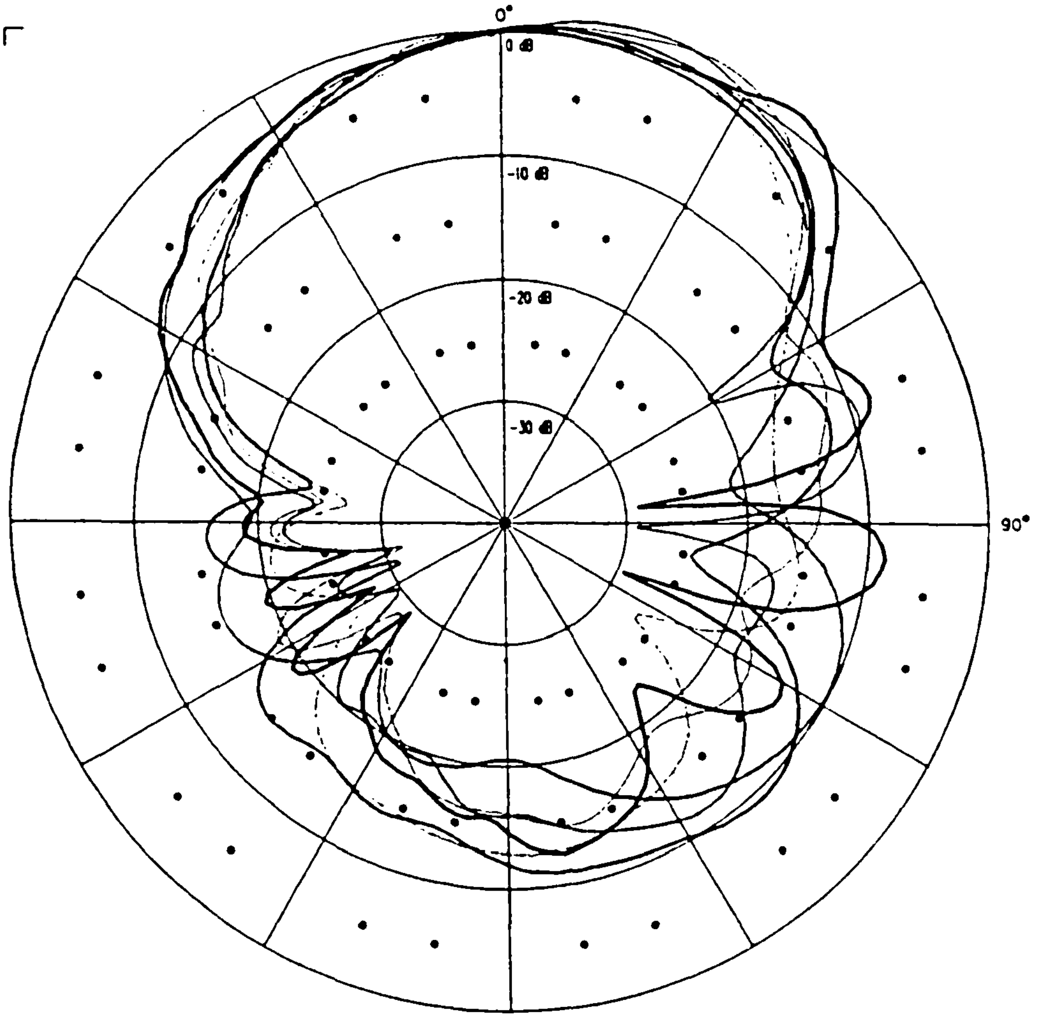


CURVE #5 POLAR CHARACTERISTICS (10 dB/Major Division)

MICROPHONE: RT.ARRAY TEST POSITION INPUT: 90 dB SPL
FREQUENCY: 1600, 1250, 1000, 800, 630, 500 Hz. DATE: 26 Jul 2001
SENSITIVITY: -38.2, -39.4, -39.9, -40.7, -41.3, -42.6 dB TIME: 11: 50: 44
S/N RATIO: 34, 50, 50, 49, 49, 47 dB INITIALS: DEG
DIRECTIVITY INDEX: 7.9, 7.9, 7.7, 6.9, 6.4, 5.6 dB DIRECTION: CCW

ETYMŌTIC RESEARCH, 61 Martin Lane, Elk Grove Village, Illinois 60007 (847) 228-0006
Etymotic (et-im-OH-tik) is a "new ancient Greek word" which means *true to the ear.* Fax (847) 228-6836

ETYMŌTIC RESEARCH

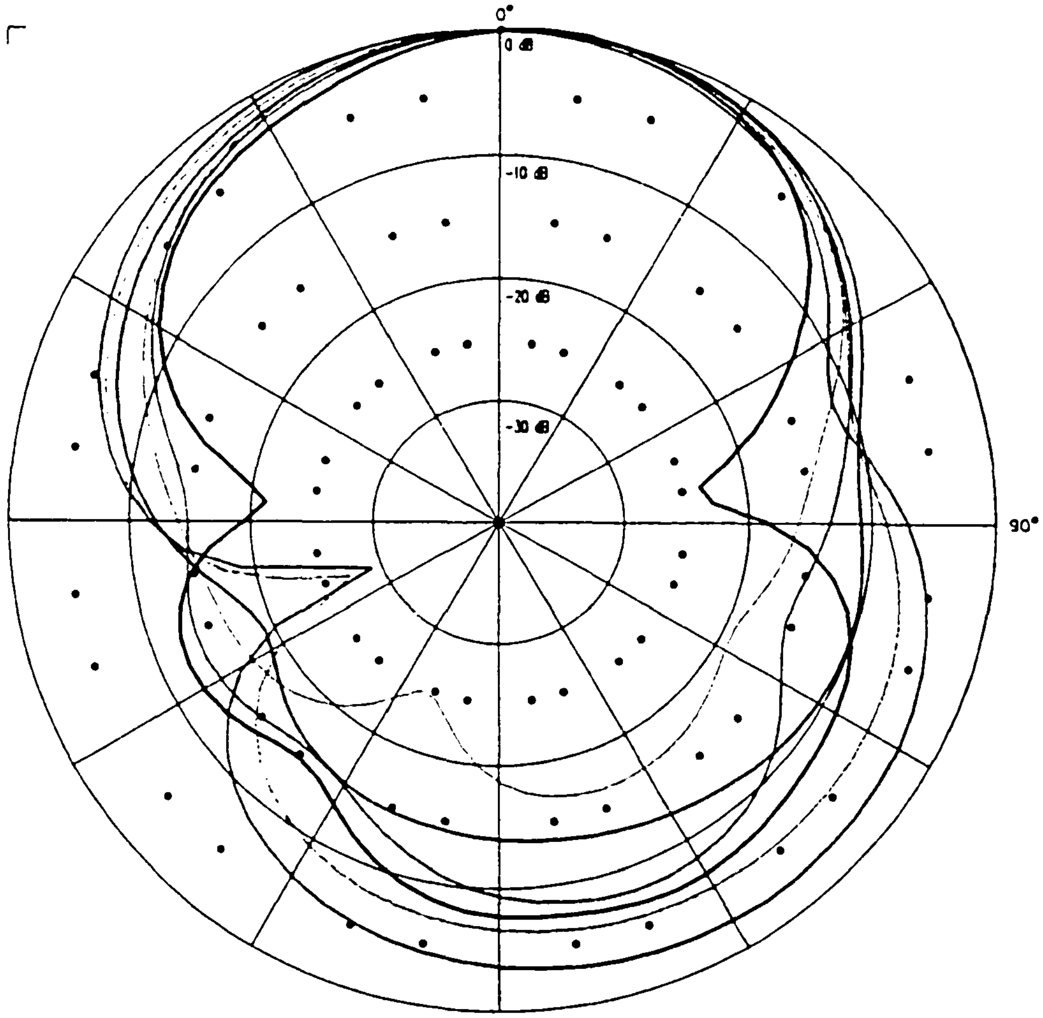


CURVE #7 POLAR CHARACTERISTICS (10 dB/Major Division)

MICROPHONE: RATPHI.TXT INPUT: 90 dB SPL
FREQUENCY: 2000, 2500, 3150, 4000, 5000, 6300 Hz. DATE: 26 Jul 2001
SENSITIVITY: -36.7, -34.5, -34.3, -34.2, -37.0, -38.0 dB TIME: 13: 49: 05
S/N RATIO: 36, 38, 38, 38, 35, 34 dB INITIALS: DEG
DIRECTIVITY INDEX: 8.5, 9.9, 8.6, 8.5, 8.5, 7.3 dB DIRECTION: CCW

ETYMŌTIC RESEARCH, 61 Martin Lane, Elk Grove Village, Illinois 60007 (847) 228-0006
Etymotic (et-im-OH-tik) is a "new ancient Greek word" which means *true to the ear.* Fax (847) 228-6836

ETYMŌTIC RESEARCH

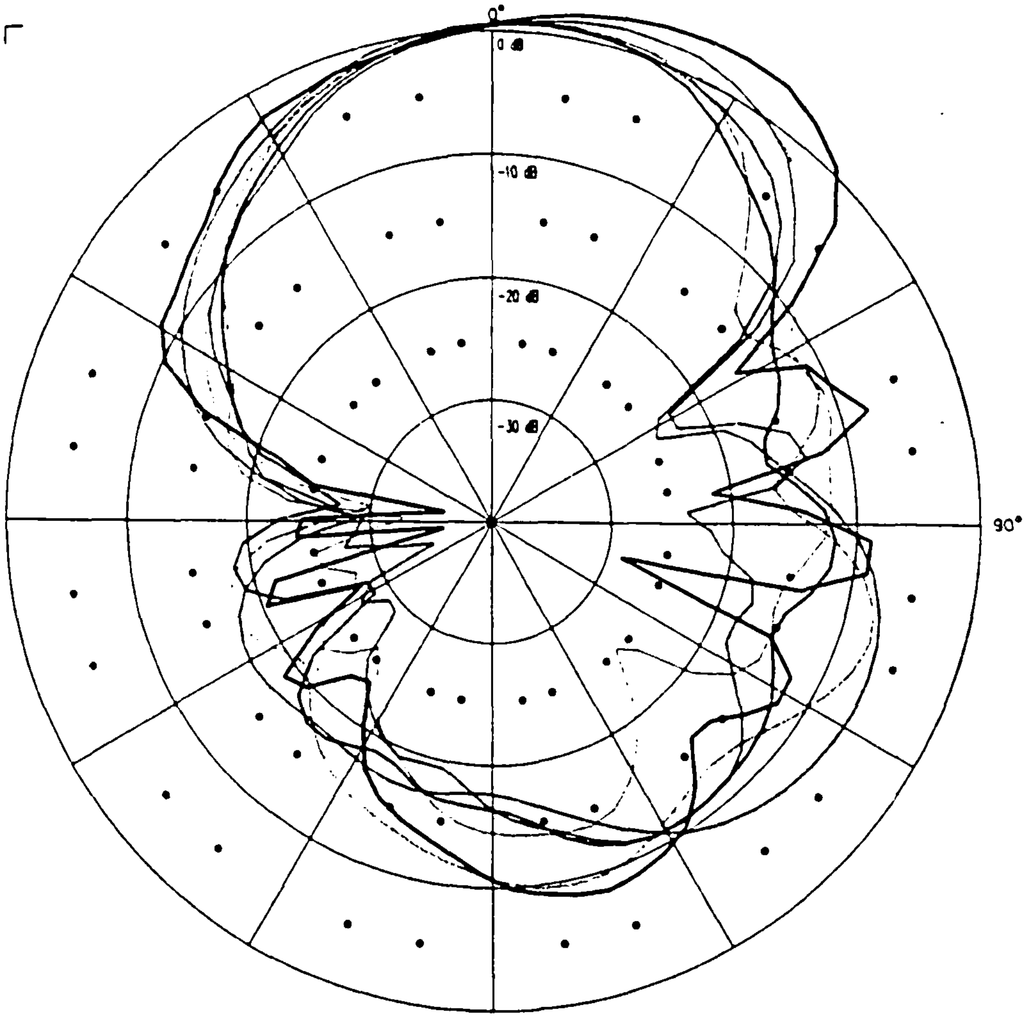


CURVE #23 POLAR CHARACTERISTICS (10 dB/Major Division)

MICROPHONE: Left Array KEMAR INPUT: 90 dB SPL
FREQUENCY: 500, 630, 800, 1000, 1250, 1600 Hz. DATE: 20 Jul 2001
SENSITIVITY: -49.1, -46.7, -46.3, -44.2, -43.8, -40.7 dB TIME: 15: 45: 03
S/N RATIO: 37, 40, 40, 42, 42, 46 dB INITIALS: DEG
DIRECTIVITY INDEX: 4.9, 6.0, 6.9, 8.0, 7.6, 8.1 dB DIRECTION: CW

ETYMŌTIC RESEARCH, 61 Martin Lane, Elk Grove Village, Illinois 60007 (847) 228-0006
Etymotic (et-im-OH-tik) is a "new ancient Greek word" which means *true to the ear* Fax (847) 228-6836

ETYMŌTIC RESEARCH



CURVE #24 POLAR CHARACTERISTICS (10 dB/Major Division)

MICROPHONE:	LEFT ARRAY KEMAR	INPUT:	90 dB SPL
FREQUENCY:	2000, 2500, 3150, 4000, 5000, 6300 Hz	DATE:	20 Jul 2001
SENSITIVITY: (dB re 1V/μbar)	-39.2, -35.8, -32.0, -30.4, -32.6, -33.3 dB	TIME:	15: 54: 24
S/N RATIO:	47, 34, 38, 40, 38, 37 dB	INITIALS:	DEG
DIRECTIVITY INDEX:	8.7, 9.5, 10.3, 9.4, 9.5, 7.0 dB	DIRECTION:	CCW

ETYMŌTIC RESEARCH, 61 Martin Lane, Elk Grove Village, Illinois 60007 (847) 228-0006
Fax (847) 228-6836

Etymotic (et-im-OH-tik) is a "new ancient Greek word" which means true to the ear

Appendix D. Summary of Calibration & Recording Process

LISTENING CONDITION	PROCESS
LIVE (Actual Restaurant)	I. Record binaural (KEMAR) noise recordings for ITE Omni, ITE D-Mic, and array microphones A. Calibration Source: Pink noise measured at 84 dB SPL[1] at FRP with loudspeaker 24" at 00 azimuth from KEMAR. B. Restaurant noise: 75 dB SPL at FRP (9 dB below pink noise calibration signal as measured post hoc via SoundForge 4.5) C. 84 dB FRP pink noise recorded through each microphone on its track at –16 dB FS (peaks) to allow 100 dB SPL to be recorded without overload. 1. Note: Target sentences were recorded later with the same microphones on KEMAR in an IAC booth. 2. Thus, if in target talker recordings the pink calibration noise is set to produce the same playback level as that recorded in the restaurant, then the SNR at the FRP should be determined solely by directional noise reduction, not level errors. D. Recordings of the live restaurant noise were made with all recording gains fixed. II. Record HINT sentences starting with 84 dB SPL pink noise calibration segment A. KEMAR in IAC booth, loudspeaker at 24" at 0^0 azimuth. B. Adjust gains for reasonable non-clipped levels in each microphone channel. C. Record HINT sentences from 8-track digital tape made earlier from HINT CD. 1. HINT calibration noise adjusted to produce 73 dB SPL at FRP 2. Thus, assume that typical HINT sentences are 11 dB below the 84 dB pink noise calibration signal. III. Make Master Binaural Test Token Tapes: 3 Pair (6 tracks): ITE-omni, ITE-D-MIC, Array Calibrate with goal of equating restaurant (LIVE) noise calibration to HINT sentences. A. Feed restaurant noise playback to mixer (six tracks) B. Feed HINT Playback to mixer (six more tracks) (Mixer

[1] All SPLs measured with C-weighting except where indicated otherwise.

	has four tracks left for monitoring recording). C. Feed mixer output to DTRS (4 tracks) D. Adjust mixer so 0 VU = -20.7 dB re digital signal clipping on record. Note: Mic noise is ≥ 26 dB SPL, max signal 81 dB SPL (55 dB dynamic range). Thus, -20 dB record - 55 dB = -80 re clipping. 16 bit digital = -96dB re: clipping. E. Record 1 kHz calibration tone at 0 VU = - 20.7 dB as in step 1 1. For noise for live restaurant condition, set record gain so 84 dB SPL pink noise = 6 dB above 0 VU (derived by the 9 dB average difference between 84 dB SPL pink noise and 75 dB SPL restaurant noise, plus a -3 dB safety factor to prevent audiometer VU meter pegging (see note in step 3 below). Set voltmeter to 0 dB = +6 VU (for purpose of adjusting each noise segments to equate rms as shown in Table 10). Thus, 1 kHz calibration tone = 78 dB SPL so that - 3 dB on VU meter = 75 dB SPL @ FRP (the level of the background noise recorded in the restaurant). 2. For HINT sentences, set record level so that 84 dB SPL pink noise = 8 dB above 0 VU (derived from the 84-73 = 11 dB SPL difference, plus 3 dB safety, or 8 dB). Thus 1 kHz calibration tone = 78 dB SPL so that - 3 dB VU = 75 dB SPL as the average level of recorded HINT sentences at the FRP. 3. Thus, for both the live restaurant noise and the HINT sentences recorded in the IAC booth, 0 VU = approximately 78 dB SPL at the FRP. (Note: the original noise level in Lou's was typically 75 dB, with bursts to 81 dB. Setting 0 VU at 78 dB means the highest bursts should just hit + 3 dB on VU meter.) F. Set record gain for each microphone as above for restaurant noise and HINT sentence recordings. G. Organization for test token recordings: 1. Subject recordings: Two, 8-track players and one 8-track recorder required. 2. For each hearing aid microphone pair, mix restaurant noise tracks with HINT IAC recordings: i. 1 DTR playback of binaurally recorded noise samples from restaurant. ii. 1 DTR playback of binaurally recorded HINT sentences recorded in IAC booth at 0^0

	azimuth. iii. 1 DTR 4-track recording of HINT sentences (tracks 1 & 2 (left/right) and noise samples (tracks 3 & 4 (left/right)). Each 20-sentence list will be matched to one of 12 possible noise segments using a Greco Latin Square design.

R-SPACE (Simulator)	<u>I. R-SPACE recordings</u> A. Setup 8 loudspeakers at points of compass and 24" from KRP B. Gain of each loudspeaker adjusted individually to produce 84 dB SPL at the FRP from previously recorded pink noise measured at 84 dB SPL in the restaurant on corresponding shotgun array mic track. C. Exception: Front-center microphone array track from restaurant is reproduced as a phantom created by splitting the signal equally between the left-front and right-front speakers such that the pink noise from each speaker adds in phase to equal 84 dB SPL (78 dB SPL + 78 dB SPL = 78 + 6 = 84) D. 84 dB SPL pink noise calibration signal from front and center as before. E. Set binaural mic recording levels, play back (and record) original restaurant noise recordings made with array of microphones. F. Use HINT sentence recordings made in IAC booth as outlined in section II (Live). <u>II. Make Master Binaural Test Token Tapes: 3 Pair (6 tracks): ITE-omni, ITE-D-Mic, Array</u> A. Set 0 VU for –20>7 dB FS B. Raise record gain 14 dB C. Set 70 dB SPL pink noise for 0 dB VU, all microphones. D. Set Equalization for R-Space (to reduce low frequency "bump" (1) Set tone generator for 0 VU at 1K (2) Add 18dB/octave filter with F_c at 75 Hz. (3) Add low frequency shelving cut adjusted so that 100Hz = -5 dB for all 3 pairs of hearing aid mics. (4) Verify that pink noise calibration signal is unaffected by these changes. E. Decrease record gain 14 dB F. Increase record gain by 6 dB so 84 dB SPL cal pink noise = 6 dB above 0 VU. This was derived by the 9 dB average difference between 84 cal and 75 noise, plus a -3 dB safety factor to prevent audiometer VU meter slamming. Thus, 1 kHz cal = 78 dB SPL so that - 3 dB on VU meter = 75 dB SPL @ FRP (the level of the background noise recorded in the restaurant). G. Set voltmeter = 0 dB = +6 VU (for purpose of adjusting each noise segments to equate rms .

IAC BOOTH 180^0	I. Noise Recordings: a. Using exactly the same setup as for the recording of the HINT sentences b. (described in section II, Live), rotate KEMAR 180^0 (24" from loudspeaker). c. 84 dB SPL pink noise calibration signal at FRP. d. Play back reference noise recording made in real restaurant and record through all binaural HA mics and reference mic without any further changes in recording system gain. e. Use previously recorded HINT sentences (see Section II, Live) II. Make Master Binaural Test Token Tapes: 3 Pair (6 tracks): ITE-omni, ITE-D-Mic, Array a. Set 0 VU for –20.7 dB FS b. Raise record gain 14 dB c. Set 70 dB SPL pint noise for 0 dB VU, all microphones d. Lower record gain 14 dB and raise it 6 dB, giving a sum total of lowering it by 8 dB. e. .Adjust voltmeter to 0 dB for purposes of adjusting each noise segments to equate rms.
IAC BOOTH OVERHEAD	I. Noise Recordings: f. Following the noise recording made at 180^0 in IAC booth, the loudspeaker was then moved overhead 24" from the KRP. g. No further adjustment was made to the loudspeaker drive or recording system gain. h. The pink noise calibration signal that had produced 84 dB SPL at the FRP from the front and from behind now produced 92 dB SPL at the FRP due to closer loudspeaker-to-FRP distance i. Play back reference noise recording made in real restaurant and record through all binaural HA mics and reference mic without any further changes in recording system gain. j. Use previously recorded HINT sentences (see Section II, Live) II. Make Master Binaural Test Token Tapes: 3 Pair (6 tracks): ITE-omni, ITE-D-Mic, Array f. Set 0 VU for –20.7 dB FS g. Raise record gain 14 dB h. Set 70 dB SPL pink noise for 0 dB VU, all microphones i. Lower record gain 14 dB and raise it 6 dB, giving a sum total of lowering it by 8 dB.. j. Adjust voltmeter to 0 dB for purposes of adjusting each noise segments to equate rms.

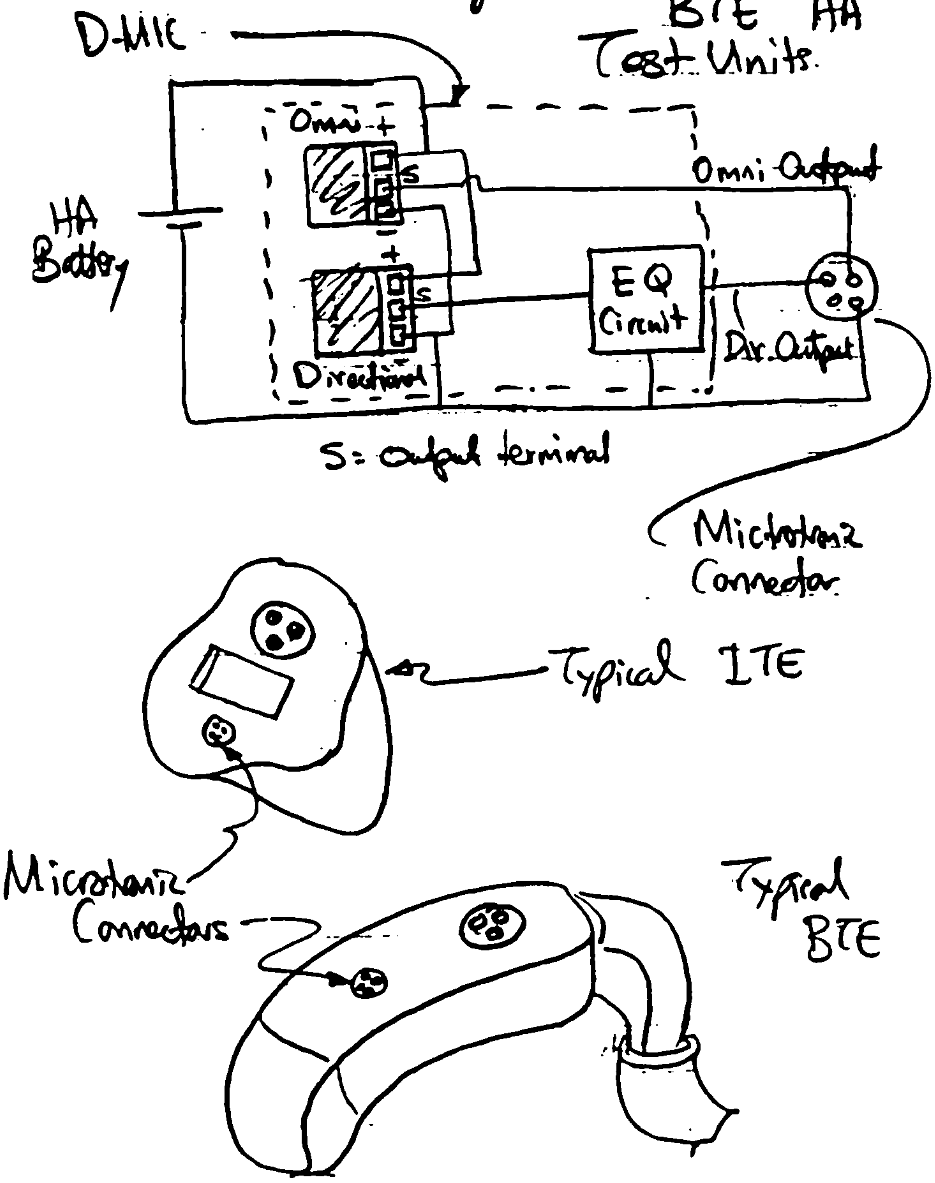
SUMMARY	**Relationship Between dBSPL and VU Reading** HINT Sentences / Restaurant Noise 84 dB SPL = +8 VU / 84 dB SPL = +6 VU 78 dB SPL = 0 VU / 78 dB SPL = 0 VU 75 dB SPL = -3 VU / 75 dB SPL = -3 VU

Relationship Between dBSPL and VU Reading	
HINT Sentences	Restaurant Noise
84 dB SPL = +8 VU	84 dB SPL = +6 VU
78 dB SPL = 0 VU	78 dB SPL = 0 VU
75 dB SPL = -3 VU	75 dB SPL = -3 VU

APPENDIX E

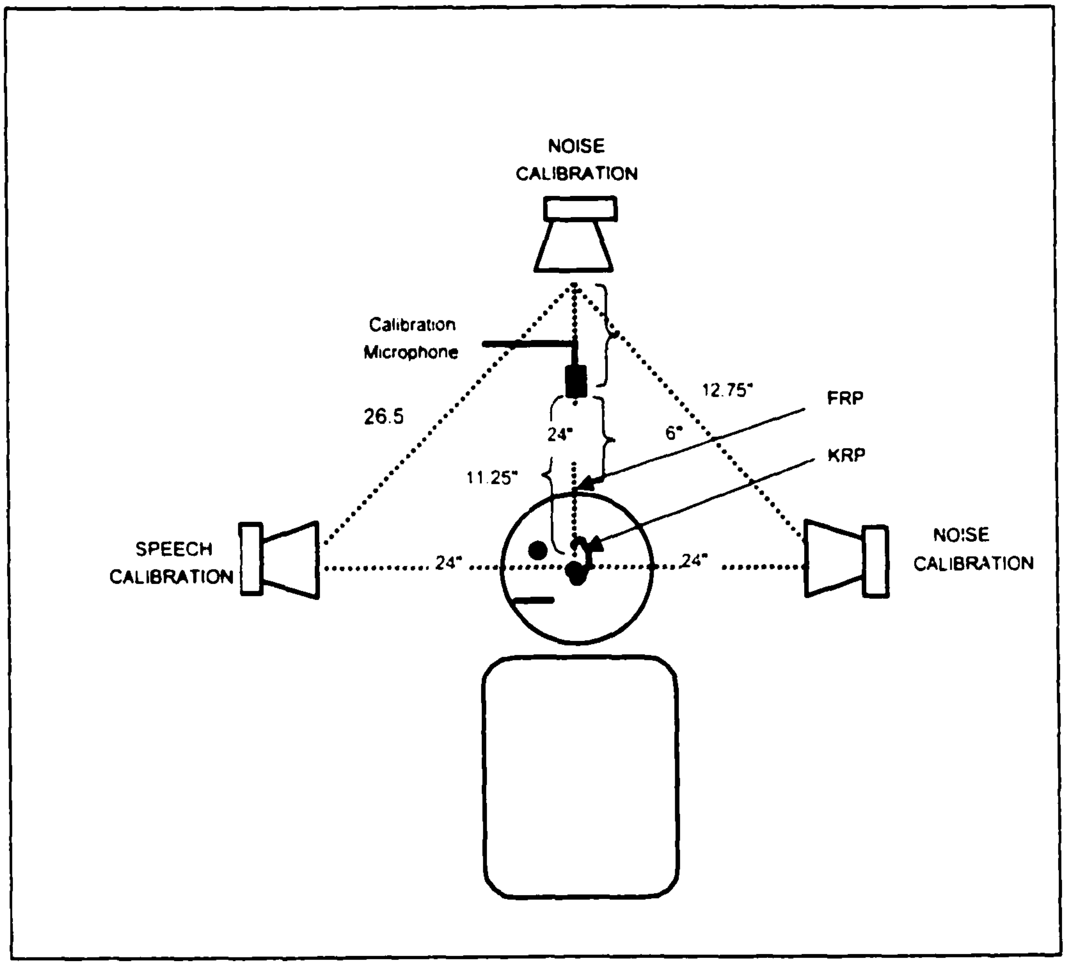
P.B. Schulein
11-14-2000

Wiring Diagrams for ITE or BTE HA Test Units



APPENDIX F

Application of the Pythagorean Theorem and the Inverse Square Law to determine the SPL at the KRP as compared to the FRP calibration point for the live condition.



1. Apply Pythagorean Theorem to determine length of hypotenuse between sentence loudspeaker and FRP.
   - $24^2 + 11.25^2 = X^2$
   - $576 + 126.56 = 702.6$
   - X^2 = SQRT of 702.6 or 26.5"

2. Apply Inverse Square Law to determine loss of signal at FRP versus KRP for Speech
   - dB loss = $20 * \log_{10}(26.5/24)$
   - = $20 * \log_{10}(1.104)$
   - = +0.9 dB

Therefore, in the direct field, it is expected that the signal at the KRP will be approximately 1 dB louder than the signal at the KRP.

3. Apply Pythagorean Theorem to determine length of hypotenuse between one of the array microphone pickups and the FRP.
   a. $120^2 + 11.25^2 = X^2$ (Microphones were at least 10 feet (120") from the KRP)
   b. 14400 + 126.6 = 14526
   c. SQRT of 14526 = 120.5"

4. Apply Inverse Square Law to determine loss of signal at FRP versus KRP for restaurant noise.
   - dB loss = $20 * \log_{10} (120/120.5)$
   - $= 20 * \log_{10}(0.995)$
   - = -0.04 dB

Therefore, in the direct field, it is expected that the noise signal at the KRP remains approximately at 84 dB SPL.

APPENDIX G

INVESTIGATION OF ARRAY MICROPHONE CLIPPING

Investigation 1: Cool Edit™ Analysis

Rationale for Investigative Procedure

One day prior to recording the live breakfast party at Lou Malnati's restaurant, a pink noise calibration signal was recorded (from an equalized loudspeaker two feet in front of KEMAR's head) at 84-dB-SPL through each hearing aid microphone fixed to KEMAR as well as through the overhead ER11 reference microphone. Immediately thereafter, the IEEE sentences were recorded in the same manner, at about 80 dB SPL, without changing recording levels. Although the IEEE sentence recordings were never used for the current investigation, they were valuable because they later alerted us to a potential calibration problem.

Subsequent to making all of the recordings in the restaurant, a listening check of all recordings made revealed the following: Although both the calibration signal and IEEE sentence materials were found to be clean (unclipped) on the ER11 track, many of the IEEE sentences recorded on the array tracks had audible clipping, especially on the left array track. Because the overall SPL of IEEE sentences was set to be about 80 dB SPL, it was suspected that clipping might be present in the 84 dB SPL pink noise calibration signal recorded through the array microphones. On December 20, 2001, visual inspection of the signal on an oscilloscope revealed no apparent clipped peaks and a normal hearing listener did not detect any audible clipping. However, since significant clipping could cause an error in playback calibration of the restaurant party recorded through the array microphones at the restaurant, a post hoc experiment was performed later to determine the extent of this clipping. Larry Revit carried out this experiment at the request of Mead Killion.

Since the IEEE sentences and pink noise calibration signal recorded on Saturday were done with the same apparatus and settings as the calibration recordings on Sunday, it was reasoned that the spectrum level of the sentences could be compared directly to the pink noise recording made on Sunday. Both the calibration signal and the IEEE sentence materials were clean (unclipped) on the ER11 track. Because the overall SPL of IEEE sentences was set to about 80 dB SPL, clipping of the 84 dB SPL calibration signal recorded through the array microphones, in the live condition only, was suspected. The party noise is known to be about 75 dB SPL overall, and was confirmed to be clear of distortion via listening checks. If a calibration error did exist, its effect would be seen in a decreased ratio (number of decibels) between the (clipped) calibration signal and the (clean) party noise due to calibration signal ceiling effects. A listening check of the IEEE sentence recordings determined some sentences to be free of clipping. Thus, they represented a known, unclipped signal recording by both the ER11 and the array microphones (again, from exactly the same apparatus and with the same settings as the pink noise calibration signal). Whatever observations could be made about the preliminary recordings would apply to the main recordings. The IEEE sentence recordings where used to provide a means equalizing and comparing the frequency responses of the array microphones to those of the ER11 reference microphone.

Assuming that a difference in frequency response, and not clipping, was the only difference between the array microphone and the ER11, then, if the ER11 were equalized to match the frequency response of the array microphones, the decibel difference (in average rms level) between the calibration signal and IEEE sentences would be identical for the array microphones and the ER11. If the difference between the calibration signal and the IEEE sentences was greater for the ER11 than for the array microphones, then the relative reduction in the difference between the two signals for the array microphones could be attributed to clipping (saturation or ceiling) effects.

Investigative Process

To investigate the above hypotheses, two IEEE sentences (which sounded clean on both array tracks) were selected. The ER11 version of one of those sentences was then equalized to match the spectrum of the same sentence as recorded through one of the array microphones. This process was repeated with the second sentence to check for the robustness of the matching process. Once it was determined that the array and ER11 versions of the two sentences were closely matched for spectrum and rms level,the same equalization was applied to the pink noise calibration signal on the ER11 track. The spectra and rms levels of the array pink noise were then compared to those of the ER11 microphone. The procedure was repeated for the other array microphone, using the same two sentences, but performing the equalization to the second sentence and checking it against the first (opposite from the experiment for the other array microphone).

Results

In both cases (left and right array microphones) the saturation of the calibration signal was determined to be, on the average, 0.4 dB (less than 0.8 dB in the left, and less than 0.1 dB for the right). This resulted in a rounded up correction factor of 0.5 dB that was to be applied, post hoc, to the live array scores only.

Investigation 2: Relationship Between Clipped Sentences and Subjects' Responses

The author of this dissertation carried out this investigation. A listening check of the array tracks revealed slight clipping of the beginning of sentences 14 and 20 (List 21/22). A detailed explanation of the clipping is illustration in the Table G1below:

Table G1. Description of Clipping in two array track HINT sentences.

SENTENCE	DESCRIPTION OF CLIPPING
"The big boy kicked the ball."	The first word sounded "clipped", even guttural in nature. Fortunately this is irrelevant since articles are not counted in the scoring process and the remainder of the sentence sounded clean.
"The machine was noisy."	The word "the" sounded distorted and almost like a clipped /ei/ with the word "machine" following in a very abridged time period. While the article is not counted in the scoring process, the word "machine" is.

Clipping was found to occur for only the conditions and subjects show in Table G2 below. The table lists the subjects and conditions and also includes an analysis of the effect (or lack thereof) on the affected subjects' scores. It should be noted that the presentation levels listed below are for the sentences only; the noise level was always 66 dB HL (81 dB SPL peak; 78 dB SPL average level).

Table G2: Analysis of effect of clipping on subjects' scores.

Subject	Condition	Analysis of Responses to Sentence 14	Analysis of Responses to Sentence 20
5	180	52 dB HL: No response. Other sentences were missed at this level, but not all. 50 dB HL was the softest presentation level (no correct responses). It is assumed that the sentence was not sufficiently audible to even make a guess.	52 dB HL: No response. Insufficient audibility for guessing.
6	Restaurant	54 dB HL: Response: "The ? was noisy." (Did not understand the word "machine.") This occurred at the softest level presented, where other sentences were sometimes counted correct and sometimes incorrect. The subject did hear the word "the" correctly. Since "boy" was not clipped, it is assumed that clipping was not a factor.	54 dB HL: Correct response.
9	Overhead	44 dB HL: Response: "She was noisy." This was the softest level presented to this subject for this condition; NO sentences were correct at this level. It is assumed the subject did not hear the word "the/a" and that "ma" of machine was masked, as was the "n" of machine. No conclusion reached with regard to clipping.	44 dB HL: Response: "The bulky boy kicked the ball." Assumption: No clipping effect.
10	R-Space	56 dB HL: Correct Response	52: Inaudible.

SUMMARY COMMENTS
Sentence 14: For subjects 5 and 10, either the sentence was too soft for guessing or it was scored as correct. For subject 6, it is likely that clipping was not a factor. For subject 9, ALL other sentences were incorrect at that level, indicated lack of sufficiently audibility. Missing the token due to clipping cannot be ruled out. However, this is balanced by the fact that subject 10 got the same sentence correct. **Sentence 20**: No problem here. Either the subject got the sentence correct (subject 6) or missed it due to inaudibility or for other reasons.
CONCLUSION
Clipping did not adversely affect the results of this study.

REFERENCES

Agnew, J. (1999). Challenges and some solutions for understanding speech in noise. High performance hearing aid solutions. *The Hearing Review*, 3, 4-13.

Agnew, J., & Block, M. (1997). HINT thresholds for a dual-microphone BTE. *Hearing Review,* 4(9), 26,29-30.

ANSI (1973). American national standard psychoacoustical terminology. New York, New York: American National Standards Institute. 1 p.

ANSI (1976). U.S.A. Standard acoustical terminology. New York, New York: American National Standards Institute.

ANSI S3.5-1969 (R 1986). American National Standard Methods for the Calculation of the Articulation Index. New York, New York: American National Standards Institute.

ANSI S3.5-1997 (Revision of ANSI S3.5-1969 (R 1986). American National Standard Methods for Calculation of the Speech Intelligibility Index. New York, New York: American National Standards Institute.

ANSI S3.35-1985 (ASA 59-1985). Methods of Measurement of Performance Characteristics of Hearing Aids under Simulated IN-SITU Working Conditions. New York, New York: American National Standards Institute.

Ballou, G.M. (1991). *Handbook for Sound Engineers: The New Audio Cyclopedia*. (Second Ed.). SAMS: a Division of Macmillan Computer Publishing.

Beck, L. (1983). Assessment of directional hearing aid characteristics. *Audiological Acoustics*, 22, 178-191.

Beranek, L.L. (1954). *Acoustics.* New York: McGraw Hill.

Burkhard, M.D., & Sachs, R. M. (1975). Anthropometric manikin for acoustic research. *Journal of the Acoustical Society of America*, 58, 214-222.

Compton, C.L. (1974). The effect of conventional and directional microphone hearing aids on speech discrimination scores. Vanderbilt University; unpublished master's thesis.

Dirks, D.D., Morgan, D.E., & Dubno, J.R. (1982). A procedure for quantifying the effects of noise on speech recognition. *Journal of Speech and Hearing Disorders*, 47, 114-123.

Dunn, O.J. (1961). Multiple comparisons among means. Journal of the American Statistical Association, 56, 52-64.

Fikret-Pasa, S. (1993). The effects of compression ratio on speech intelligibility and quality. Northwestern University Ph.D. Dissertation, University Microfilms, Ann Arbor, MI.

Fletcher, H. (1953). Speech and Hearing in Communication (van Nostrand, New York).

French, N.R. & Steinberg, J.C. (1947). Factors governing the intelligibility of speech sounds. *Journal of the Acoustical Society of America,* 19, 90-119.

Fortune, T. (1997). Real ear polar patterns and aided directional sensitivity. *Journal of the American Academy of Audiology*, 8(2), 119-131.

Hawkins, D.B., Walden, B.E., Montgomery, A., Prosek, R.A. (1987). Description and validation of an LDL procedure designed to select SSPL90. *Ear and Hearing*, 8, 162-169.

Hawkins, D.B., & Yacullo, W.S. (1984). Signal-to-noise ratio advantage of binaural hearing aids and directional microphones under different levels of reverberation. *Journal of Speech and Hearing Disorders*, 49, 278-286.

Killion, M., Schulein, R., Christensen, L., Fabry, D., Revit, L., Niquette, P., & Chung, K. (1998). Real-world performance of an ITE directional microphone. *The Hearing Journal*, 51(4), 1-6.

Killion, M., & Villchur, E. (1993). Kessler was right--partly: But SIN test shows some aids improve hearing in noise. *The Hearing Journal*, 46(9), 31-35.

Killion, M.C. (1997a). The SIN report. Circuits haven't solved the hearing-in-noise problem. *The Hearing Journal*, 50(10), 28-30, 32.

Killion, M.C. (1997b). Hearing aids: past, present, future: moving toward conversation in noise. *British Journal of Audiology*, 31, 141-148.

Kochkin, S. (1993). MarkeTrak III identifies key factors in determining customer satisfaction. *The Hearing Journal*, 46, 39-44.

Kochkin, S. (1994). MarkeTrak III: Why 20 million in U.S. don't use hearing aids for their hearing loss. *The Hearing Journal*, 46(1), 20-27.

Leeuw, A.R., & Dreschler, W.A. (1991). Advantages of directional hearing aid microphones related to room acoustics. *Audiology*, 30, 330-344.

Lentz, W. (1997). A summary of research using directional and omnidirectional hearing aids. J. Audiological Technique. 42-65.

Lentz, W.E. (1972). Speech discrimination in the presence of background noise using a hearing aid with a directionally-sensitive microphone. *Maico Audiologic Library Series*, 10(9), 1-4.

Levitt, H. (1971). Transformed up-down methods in psychoacoustics. *Journal of the Acoustical Society of America*, 49(2), 467-477.

Levitt, H. (2000). Personal communication.

Lurquin, P., & Rafhay, S. (1996). Intelligibility in noise using multi-microphone hearing aids. *Acta Otorhinolaryngol Belg*, 50(2), 103-109.

Madison, T.K., & Hawkins, D.B. (1983). The signal-to-noise ratio advantage of directional microphones. *Hearing Instruments*, 34(2), 18-49.

Moncour, J.P., & Dirks, D. (1967). Binaural and monaural speech intelligibility in reverberation. *Journal of Speech and Hearing Research*, 10, 186-195.

Mueller, H., & Johnson, R. (1979). The effects of various front-to-back ratios on the performance of directional microphone hearing aids. *Journal of the American Auditory Society*, 5, 30-34.

Mueller, H. G., & Killion, M.C. (1990). An easy method for calculating the articulation index. *The Hearing Journal*, 43 (9), 14-17.

Mueller, H., & Sweetow, R. (1978). Clinical rationale for using an overhead speaker in the evaluation of hearing aids. *Archives of Otolaryngology*, 104, 417-418.

Mueller, H.G. (1981). Directional hearing aids: a ten-year report. *Hearing Instruments*, 32, 16-19.

Mueller, H.G., Grimes, A.M., & Erdman, S.A. (1983). Subjective ratings of directional amplification. *Hearing Instruments*, 34(2), 14-16-47-48.

Mueller, H.G. & Killion, M.C. (1990). The count-the-dot method for calculation of the articulation index. *The Hearing Journal*, 43 (9).

Mueller, H.G., & Hawkins, D.B. (1990). Three important considerations in hearing aid selection. In Sandlin, R.E. *Handbook of Hearing Aid Amplification*, volume II. (pp. 31-60). Boston: College-Hill Press.

Nabelek, A.K., & Mason, D. (1981). Effect of noise and reverberation on binaural and monaural word identification by subjects with various audiograms. *Journal of Speech and Hearing Research*, 24, 375-383.

Nielsen, H.B. (1973). A comparison between hearing aids with a directional microphone and hearing aids with conventional microphone. *Scandinavian Audiology*, 2, 45-48.

Nielsen, H.B., & Ludvigsen, C. (1978). Effect of hearing aids with directional microphones in different acoustic environments. *Scandinavian Audiology*, 7, 217-224.

Nilsson, M., Gellnet, D., & Sullivan, J.A. (1992). Norms for the hearing in noise test: The influence of spatial separation, hearing loss and English language experience on speech reception thresholds. *Journal of the Acoustical Society of America*, 92(S2385)

Nilsson, M., Soli, S.D., & Sullivan, J.A. (1994). Development of the Hearing in Noise Test for the measurement of speech reception thresholds in quiet and in noise. *Journal of the Acoustical Society of America*, 2, 1085-1099.

Nilsson, M., Soli, S.D., & Sumida, A. (1994). A definition of normal binaural sentence recognition in quiet and noise. Unpublished House Ear Institute internal document, 1-13.

Pearsons, K.S., Bennett, R. L., & Fidell, S. (May 1977). Speech levels in various noise environments. (Report No. 600/1-77-025). Washington, DC: Office of Health and Ecological Effects, Office of Research and Development, U.S. Environmental Protection Agency.

Peutz, U.M.A. (1971). Articulation loss of consonants as a criterion for speech transmission in a room. *J. Audio Engineering Society*, 19, 915-919.

Plomp, R. (1978). Auditory handicap of hearing impairment and the limited benefit of hearing aids. *Journal of the Acoustical Society of America*, 63(2), 533-549.

Plomp, R., & Mimpen, A. (1979). Speech reception threshold for sentences as a function of age and noise level. *Journal of the Acoustical Society of America*, 66, 1333-1342.]

Preves, D. (1975). Selecting the best directivity pattern for unidirectional noise suppressing hearing aids. Hearing Instruments, October, 18-19;42.

Preves, D. (1976). Directivity of in-the-ear aids with non-directional and directional microphones. *Hearing Aid Journal*, 29(6), 9-12.

Preves, D. (1997). Directional microphone use in ITE instruments. *The Hearing Review*, 4 (7), 21-27.

Preves, D., Sammeth, C.A., & Wynne, M.K. (1999). Field Trial Evaluations of a switch directional/omnidirectional in-the-ear hearing instrument. *Journal of the American Academy of Audiology*, 10, 273-284.

Resnick, S.B., Dubno, J.R., Hoffnung, S., & Levitt, H. (1075). Phoneme erros on a nonsense syllable test. Journal of the Acoustical Society of America, 58, 114.

Revit, L., Schulein, R., Compton, C., & Killion, M. (2000). Multi-Channel Sound-Field Reproduction and Simulation System for Assessing Hearing and Hearing-Aid Performance, Paper presented at the 45th Congress of Hearing Aid Acousticians, October 6, 2000, Cologne, Germany.

Ricketts, T. (2000a). Impact of noise source configuration on directional hearing aid benefit and performance. *Ear & Hearing*, 21(3), 194-205.

Ricketts, T. (2000b). Directivity quantification in hearing aids: fitting and measurement effects. *Ear & Hearing*, 21(1), 45-58.

Ricketts, T., & Dhar, S. (1999a). Aided benefit across directional and omni-directional hearing aid microphones for behind-the-ear hearing aids. *Journal of the American Academy of Audiology* 10(4), 180-189.

Ricketts, T., & Dhar, S. (1999b). Comparison of performance across three directional hearing aids. *Journal of the American Academy of Audiology*, 10, 180-189.

Ricketts, T., & Mueller, H. (1999). Making sense of directional microphone hearing aids. *American Journal of Audiology*, 8, 117-127.

Roberts, M., & Schulein, R. (1997). Measurement and intelligibility optimization of directional microphones for the use in hearing aid devices. Presented at the 103rd meeting of the Audio Engineering Society, New York.

Rumoshosky, J. (1976). Directional microphones in in-the-ear aids. Hearing Aid Journal, June, 11;48-50.

Schulein, R. (2000). Personal communication.

Soede, W., (1990). Improvement of speech intelligibility in noise: Development and evaluation of a new directional hearing instrument based on array technology. Ph.D. Thesis. The Netherlands: Delft University of Technology. 15-21.

Soede, W., Berkhout, A.J., & Bilsen, F.A. (1993a). Development of a directional hearing instrument based on array technology. *Journal of the Acoustical Society of America*, 94(2), 785-798.

Soede, W., Bilsen, F.A., & Berkhout, A.J. (1993b). Assessment of a directional microphone array for hearing impaired listeners. *Journal of the Acoustical Society of America*, 94(2), 799-808.

Soede, W., (2001). Personal communication.

Soli, S.D., & Nilsson, M. (1994). Assessment of communication handicap with the HINT. *Hearing Instruments*, 45(12), 15-16.

Sound Forge 4.5 [Computer software]. (1998). Sonic Foundry Corporate Headquarters, 1617 Sherman Avenue, Madison, WI 53704.

Studebaker, G., Cox, R., & Formby, C. (1980). The effect of environment on the directional performance of head worn hearing aids. In Hochberg, I. (Ed), *Acoustical factors affecting hearing aid performance.* (pp. 81-105). Baltimore: University Park Press.

Syntrillium Cool Edit Pro 1.2 [Computer software]. (2001). Syntrillium Software, P.O. Box 62255, Phoenix, Arizona 85082.

Tillman, T.W., Carhart, R., & Olsen, W.O. (1970). Hearing aid efficiency in a competing speech situation. *Journal of Speech and Hearing Research*, 13, 789-811.

Valente, M., & Fabry, D.A.P.L.G. (1995). Recognition of speech in noise with hearing aids using dual microphones. *Journal of the American Academy of Audiology*, 6, 440-449.

Valente, M., Schuchman, G., Potts, L. G., & Beck, L. (2000). Performance of dual-microphone in-the-ear hearing aids. *Journal of the American Academy of Audiology*, 11, 181-189.

Veit, I. (2000). Production of spatially limited (diffuse) sound field in an anechoic room. Journal of the *of the Audio Engineering Society* 35, 138-143.

Villchur, E. (1962). A method of testing loudspeakers with random noise input. *Journal of Audio Engineering*, 10, 306-309.

Villchur, E. (1964). Technique of masking live versus recorded comparisons, *Audio* (October, 1964), 34.

Voss, T. (1997). Clinical evaluation of multi-microphone hearing instruments. *Hearing Review*, 4(9), 73-74.

Weston, D. E., (1986). Jacobi sensor arrangement for maximum array directivity. *Journal of the Acoustical Society of America* 80, 1170-1181.

Wouters, J., Lltiere, L., & van Wieringen, A. (1999). Speech intelligibility in noisy environments with one- and two-microphone hearing aids. *Audiology*, 38, 91-98.