J Am Acad Audiol 15:440–455 (2004)

Performance of Directional Microphones for Hearing Aids: Real-World versus Simulation

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Abstract

The purpose of this study was to assess the accuracy of clinical and laboratory measures of directional microphone benefit. Three methods of simulating a noisy restaurant listening situation ([1] a multimicrophone/multiloudspeaker simulation, the R-SPACE™, [2] a single noise source behind the listener, and [3] a single noise source above the listener) were evaluated and compared to the "live" condition. Performance with three directional microphone systems differing in polar pattern (omnidirectional, supercardioid, and hypercardioid array) and directivity indices (0.34, 4.20, and 7.71) was assessed using a modified version of the Hearing in Noise Test (HINT). The evaluation revealed that the three microphones could be ordered with regard to the benefit obtained using any of the simulation techniques. However, the absolute performance obtained with each microphone type differed among simulations. Only the R-SPACE simulation yielded accurate estimates of the absolute performance of all three microphones in the live condition. Performance in the R-SPACE condition was not significantly different from performance in the "live restaurant" condition. Neither of the single noise source simulations provided accurate predictions of real-world (live) performance for all three microphones.

Key Words: Articulation Index-Directivity Index, benefit, directional microphones, Directivity Index, hearing aids

Abbreviations: AI-DI = Articulation Index-Directivity Index; DI = Directivity Index; HINT = Hearing in Noise Test; ITE = in the ear; RTS = Reception Threshold for Sentences; SNR = Signal-to-Noise Ratio

Sumario

El propósito de este estudio fue evaluar la exactitud de las medidas clínicas y de laboratorio sobre el beneficio del micrófono direccional. Se evaluaron tres métodos de simular situaciones auditivas de restaurante ruidoso ([1] una simulación multi-micrófono/multi-altoparlante, el R-SPACE™, [2] una fuente única de ruido detrás del sujeto, y [3] una fuente única de ruido por encima del sujeto) y se compararon con una condición "en vivo". Se evaluó el desempeño de tres sistemas de micrófonos direccionales con diferentes patrones polares (omni-direccional, supercardioide, e hipercardioide) y los índices de direccionalidad (0.34, 4.20, y 7.71), usando una versión modificada de la Prueba de Audición en Ruido (HINT). La evaluación reveló que los tres micrófonos podían ordenarse en relación con el beneficio obtenido utilizando cualquiera de las técnicas de simulación. Sin embargo, el rendimiento absoluto obtenido con cada tipo de micrófono difirió de acuerdo a las simulaciones. Sólo la simulación R-SPACE rindió estimaciones exactas del desempeño

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This dissertation work was supported primarily by a student grant from Etymotic Research. Financial incentives for the listeners were provided by the Gallaudet Research Institute.

absoluto de los tres micrófonos en condiciones en vivo. El desempeño en la condición de R-SPACE no fue significativamente diferente del desempeño en la condición de "restaurante en vivo". Ninguna de las simulaciones de una fuente única de ruido aportó predicciones exactas del desempeño en la vida real (en vivo) de los tres micrófonos.

Palabras Clave: Indice de Articulación-Indice de Direccionalidad, beneficio, micrófonos direccionales, Indice de Direccionalidad, auxiliar auditivo

Abreviaturas: AI-DI = Indice de Articulación-Indice de Direccionalidad; DI = Indice de Direccionalidad; HINT = Prueba de Audición en Ruido; ITE = intra-auricular; RTS = Umbral de Recepción para Frases; SNR = Tasa Señal/Ruido

Improvements in the design and in the performance of directional microphones for hearing aids have led to improved ability to recognize speech in some noisy environments and in increased user satisfaction (Kochkin, 1996, 2000). There are many types of directional microphone systems available for use in hearing aids. As such, research and clinical facilities require an efficient and practical method to (1) document performance for patients, their families, and third party providers, and (2) predict how well directional microphone hearing aids (DMHAs) will perform in real acoustic environments such as restaurants, living rooms, classrooms, churches, and in other settings, characterized by the presence of both noise and reverberation. Typically, performance with DMHAs is compared with performance of the same hearing aid with an omnidirectional microphone. The method of evaluation usually involves simulation of a noisy environment in a sound-treated room. The speech signal is introduced through a loudspeaker placed in front of the hearing aid user, and noise is introduced into the room from one or more loudspeakers. Performance with the hearing aid is measured using an omnidirectional microphone and with the directional microphone under study. A comparison of the two measures gives an indication of the benefit to be obtained with the directional microphone.

Studies have shown that the benefit provided by a DMHA is related to its polar pattern, as well as variations in the test environment. Factors such as the intensity level of the target stimuli, signal-to-noise ratio (SNR), room reverberation, location of the listener, and location of the noise source(s) will all affect directional microphone benefit (e.g., Nielsen, 1973; Nielsen and Ludvigsen, 1978; Studebaker et al, 1980; Madison and Hawkins, 1983; Hawkins and Yacullo, 1984; Ricketts and Dhar, 1999; Ricketts and Mueller, 1999). A review of previous research reveals several possible deficiencies in the methods used for evaluating directional microphone efficacy. In many studies, simulation of a noisy listening environment has been accomplished by placing a single noise source directly behind the listener, that is, at 180° azimuth (e.g., Lentz, 1972; Mueller and Johnson, 1979; Madison et al, 1983; Hawkins et al, 1984; Valente, et al, 1995; Lurquin and Rafhay, 1996). While there might, indeed, be occasions where a listener would encounter a single noise source directly behind, this test condition is not typical of the listening conditions that bother most people. In addition, an evaluation method that utilizes a signal in front of the listener and noise directly behind the listener will show maximum benefit for microphones with maximum attenuation (null) at 180° (i.e., a cardioid pattern of directivity) as compared to modern day supercardioid and hypercardioid microphones whose polar patterns are characterized by rear lobes. In some early studies, multiple noise sources were used (e.g., Nielsen, 1973; Compton, 1974; Preves, 1975; Rumoshosky, 1976; Lentz, 1977), but in most of these cases the noises were correlated (waveforms from each loudspeaker were similar). Correlated noise is not typical of most listening situations.

The use of multiple noise sources would seem to be necessary because modern directional hearing aids contain microphones having varying polar patterns and degrees of directivity. In real-world environments such as a restaurant or a cocktail party, noises may arise from all directions. Therefore, in order to assess the improvement in SNR achieved by directional hearing aids, it would be advantageous to have noise arising from multiple directions in the evaluation environment.

The effect of room reverberation is another factor that is typically not assessed in evaluation procedures. Most studies have been carried out in anechoic chambers or in sound-attenuating booths; thus, the reverberation conditions are unrealistically low.

The findings of recent studies emphasize the importance of including such factors as part of the evaluation procedure. For example, Ricketts (2000) studied the effect of the configuration of multiple noise source(s) in two reverberant environments. The Hearing in Noise Test (HINT) (Nilsson et al, 1994a) was used to determine the absolute binaural reception threshold for sentences (RTS) for three pairs of different directional hearing aids, as well as the directional benefit (difference between the RTS for omnidirectional and directional conditions). Listeners with sensorineural hearing loss were tested in two listening environments: (1) a "living room" with a reverberation time of 0.6 seconds and (2) a "classroom" with a reverberation time of 1.1 seconds. Four noise source configurations were studied, including a signal located in front and noise at (a) 180° (typical of earlier evaluation methods); (b) 90°, 135°, 180°, 225°, and 270°, (typical of listening in the front of a class or in the theater); (c) 30°, 105°, 180°, 225°, and 330° (typical of an environment with more diffuse noise); and (d) 30°, 105°, 180°, 225°, and 330° but with the 30° and 330° loudspeakers turned perpendicular to the listener (typical of a situation in which the noise sources in the front are farther away).

Both reverberation and noise configuration were found to affect the directional benefit across hearing aids. In the living room environment, directional benefit ranged between 3.6 to 7.9 dB, depending on the noise source configuration. This directional benefit decreased to a range of 2 to 5.1 dB in the classroom setting. Directional benefit was significantly higher for the 0°/180° loudspeaker configuration in comparison with all others. Significantly less directional benefit was provided to listeners in the diffuse restaurant configuration (condition c) than the classroom or restaurant configuration where the background noise at 30° and 330° was reduced by 5 dB (condition d). These results reveal that the 0°/180° test configuration commonly used in clinical evaluation may overestimate the benefit that will be obtained in more realistic environments having multiple noise sources. Second, an inverse relationship was noted between directional benefit/performance and reverberation time across all hearing aid brands, that is, as reverberation time increased, directional benefit/performance decreased. Although smaller in magnitude, this trend is in agreement with previous investigations (Studebaker et al, 1980; Madison and Hawkins, 1983; Hawkins and Yacullo, 1984).

While Ricketts attempted to simulate real-world effects in the clinic, Killion and colleagues (1998) took a very different approach—they recorded evaluation materials in real-world environments. Test recordings were made in several different environments while subjects wore prototypes of binaural in-the-ear (ITE) hearing aids equipped with both omnidirectional and supercardioid microphones. Several pairs of ITE hearing aids were equipped with D-Mic™ cartridges whose outputs were available through subminiature Microtronic four-pin connectors. One pin was connected to the omnidirectional microphone output and another pin 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 sequence of sentence blocks modeled after the SIN (Speech in Noise) Test (Fikret-Pasa, 1993; Killion and Villchur, 1993) was recorded in various noisy real-world environments: a crowded street party (90–95 dBA), two restaurants (70–80 dBA and 60–65 dBA), a museum party (80–85 dBA), and a classroom party simulation (80–85 dBA). 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 attempted to address the need for a test environment that approximates common real-world reverberation and noise conditions. A comparison of the results measured with the outdoor (street party) and indoor recordings showed that individuals with hearing loss obtained greater benefit (9 dB improvement) with the directional microphones in the outdoor situation. 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 room reverberation and talker-listener distance made listening more difficult.

As demonstrated by these studies, it may be difficult to predict performance of a particular DMHA for a specific listening environment because of the complex interaction between the characteristics of the microphone and the characteristics of the environment in which the hearing aid is used. Recently a system was developed for the purpose of accurately recording and then reproducing/simulating real-world environments for hearing aid evaluations (Revit et al, 2002a, 2002b). The system, called the R-SPACE, consists of a circular, horizontal array of eight interference-tube (shotgun) microphones that can be placed in a circular configuration in the environment to be recorded for later simulation. Once the noisy/reverberant recordings have been made, the recorded environment can be recreated by playing back the recordings through an array of eight loudspeakers placed in a configuration that mimics the configuration and placement that had been used for the microphones at the time of recording. If this technique were successful in reproducing specific listening environments, it would make it possible to obtain accurate assessments of hearing aids with directional microphones.

The purpose of the present study was to assess whether real-world/"live" performance with DMHAs could be accurately assessed in a clinical/laboratory environment. The absolute accuracy of the measured benefit as well as the ability to correctly order the relative benefit of three hearing aid microphones were considered.

In order to be able to assess real-world/live performance, recordings were made in a noisy restaurant through three sets of binaural microphones placed on a Knowles Electronic Manikin for Acoustic Research (KEMAR). This condition, which we call "live," served as the reference condition, or the gold standard to which the simulations would be compared. The R-SPACE simulation technique was compared to more traditional methods for simulating a noisy environment: (1) use of a single loudspeaker for generating noise from the rear of the listener, a traditional technique, and (2) use of a single loudspeaker for generating noise from overhead. These two single-loudspeaker competition paradigms have previously been used for evaluation purposes. The latter was proposed as an efficient method of creating a simulated diffuse noise field (Mueller and Sweetow, 1978). For all of the test conditions, noise was recorded in the busy neighborhood restaurant that was to be simulated.

METHOD

Directional Microphones under Test

Three pairs of hearing aid microphones differing in directionality were used in this study: (1) an ITE omnidirectional microphone; (2) an ITE supercardioid microphone (D-Mic); and (3) a five-element endfire array microphone with hypercardioid characteristics (Soede et al, 1993a).

These three microphones were selected because they represented a wide range of directivity and had different polar patterns. The in situ (all microphones in place on KEMAR) Directivity Index (DI) and Articulation Index-weighted Directivity Index (AI-DI) values of these microphones as measured under anechoic conditions appear in Table 1. The AI-DI is a method used to predict the effect of the directivity on speech recognition performance. Measurements were made at a distance of 24 inches from the

Table 1. DI and AI-DI Values (AI-DI values in parentheses) for Hearing Aid Microphones as Measured on KEMAR under Anechoic Conditions, 24 Inches from the Loudspeaker

	Right Ear	Left Ear	Average
Omnidirectional	1.10 (1.30)	-0.42 (0.36)	0.34 (0.83)
Supercardioid	4.35 (4.70)	4.05 (4.32)	4.20 (4.51)
Hypercardioid	8.03 (8.03)	7.39 (7.33)	7.71 (7.68)

loudspeaker, the listening distance used in the study.

Instrumentation for Recording

The same KEMAR and microphone setup was used to make recordings in all four environments (live, R-SPACE, 0°/180°, and 0°/90°), as well as to measure the electroacoustic characteristics of the microphones, including directivity.

Three sets of binaural hearing aid microphones were simultaneously mounted to KEMAR. ITE hearing aid cases contained both omnidirectional and supercardioid microphones that functioned simultaneously. The hypercardioid (array) microphone was taped to the side of KEMAR's head approximately an inch above each ear and facing slightly downward. The output of each of the hearing aid microphones was pre-amplified and then amplified before recording. The outputs from all of the microphones were recorded without additional hearing aid circuitry, such as signal processing and frequency shaping.

The same KEMAR and microphone setup was used to make recordings in all four environments (live, R-SPACE, 0°/180°, and 0°/90°), as well as to measure the electroacoustic characteristics of the microphones, including directivity.

An ER-11 half-inch microphone was suspended six inches above the apex of KEMAR's head and was used for calibration, as well as for recording. The output of this microphone was also amplified before being sent to the multitrack recording system.

To make the noise recordings for playback in the R-SPACE, a circular, horizontal array of eight interference-tube (shotgun) microphones was placed in equally distributed, 45-degree angular increments around KEMAR (Figure 1).

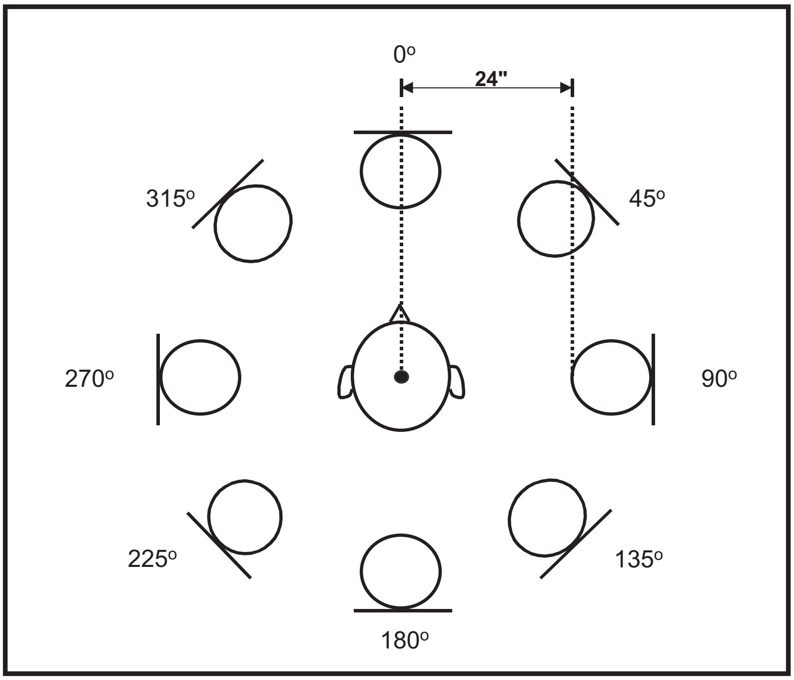

Figure 1. Multimicrophone array (surrounding KEMAR) used to record restaurant background noise for R-SPACE. Illustration adapted from Revit et al (2002b).

The acoustic center of each microphone was positioned at a distance of 24 inches from the center of KEMAR's head, facing outward. A multitrack DTRS, consisting of two Tascam DA-38 eight-track recorders was used to record the three pairs of binaural hearing aid microphone tracks and the ER-11 omnidirectional microphone signals. A third recorder (DA-98) was used to record the signals for the R-SPACE simulation. Each DTRS unit uses a helical scan head system to record up to eight tracks of 16-bit, linear pulse-code modulated digitized audio on a Hi8-size cassette tape. A sampling rate of 48 kHz was employed. The three DTRS units were driven in synchrony by the clock of the DA-98. Thus, 24 synchronized tracks were available.

Restaurant Noise Recordings

Noise recordings were made in a busy neighborhood restaurant. The noise stimuli were recorded during a breakfast party attended by 42 people. A restaurant environment was chosen because it is an environment that causes considerable speech recognition problems for hearing aid users due to the presence of uncorrelated noise sources at all azimuths. Ambient noise levels measured in the restaurant on several different occasions revealed the average noise level to be 75 dB SPL and the average signal-to-noise ratio (C-scale) to be approximately +5 to +10 dB. Subjectively, the nature of the noise was judged to be rather diffuse, that is, it was difficult to single out any one particular person's speech over another's.

Three sets of noise recordings were made simultaneously: These recordings included (1) recordings through the three pairs of hearing aid microphones mounted on KEMAR (for use in the live condition); (2) an eight-track multimicrophone array recording of the restaurant noise (for use in the R-SPACE condition); and (3) one ER-11 overhead omnidirectional microphone track (for calibration and for preparation of test materials for the 0°/90° and 0°/180° degree conditions).

The KEMAR was positioned in the middle of the main dining room of the restaurant in a location normally used for a small dining table, situated among many nearby tables occupied by other diners. The manikin was oriented at an angle to the walls of the restaurant and a foam "hairpiece" was affixed to the top of KEMAR's head to reduce the reflections from the head to the reference microphone.

A Tannoy Arena loudspeaker, equalized for flat response +/- 3 dB for the 1/3-octave bands centered at 160 Hz to 16 kHz, was placed at a distance 24 inches in front of KEMAR (24 inches from the pick-up point of each head-worn microphone). A pink noise calibration signal was delivered through the loudspeaker. The calibration signal was 84 dB SPL at the chosen field reference point (FRP),

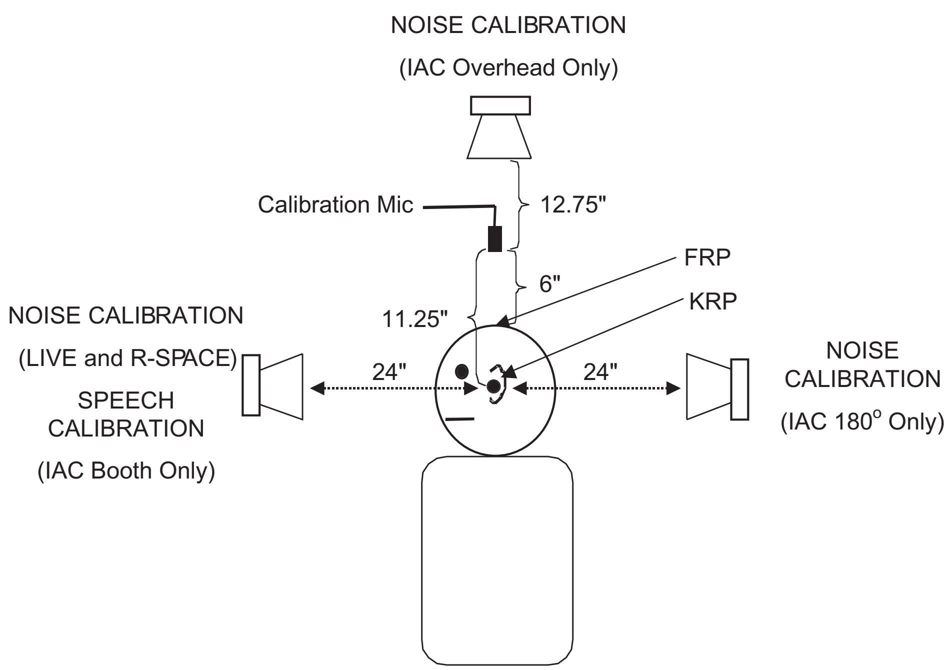

Figure 2. Calibration setup for field reference point (FRP).

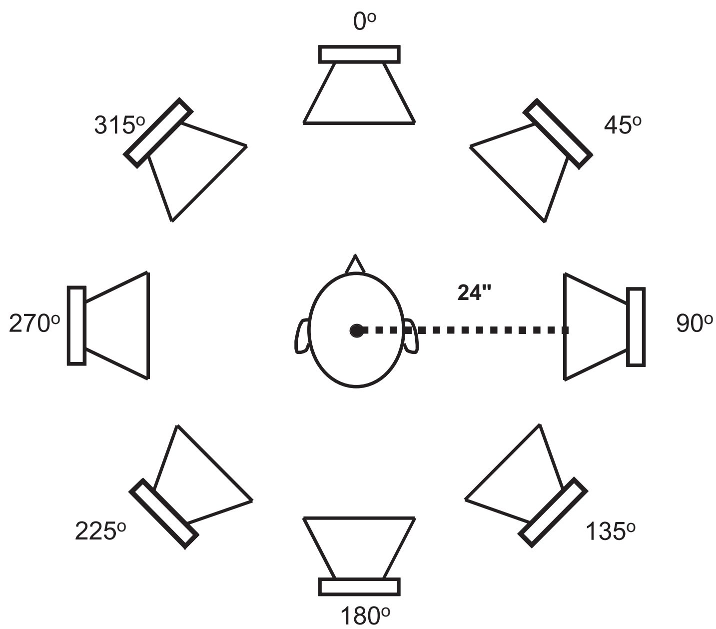

Figure 3. R-SPACE playback/recording system (all loudspeakers 24 inches from the KEMAR reference point [KRP]). Illustration adapted from Revit et al (2002b).

six inches above the apex of KEMAR. Figure 2 illustrates the arrangement of the equipment for calibration and recording in the restaurant, as well as the recording sessions that took place in the simulator and in the IAC booth. The calibration signal was recorded simultaneously on separate tracks of the DTRS through each hearing aid microphone, as well as through the ER-11 calibration/recording microphone.

Recording of Noise for R-SPACE Condition

The recordings obtained in the restaurant from the array of shotgun microphones were used to produce the R-SPACE condition. The R-SPACE recording/playback system was placed in a large room (dimensions = 19.4' L x 17' W x 7' H). Each of the eight tracks recorded in the restaurant through the multimicrophone array was played back through each of eight Tannoy Arena loudspeakers placed in a circular pattern around KEMAR with the emanating surface of each loudspeaker 24 inches from the KEMAR reference point (KRP) (Figure 3). The same calibration loudspeaker, KEMAR manikin, head-worn microphones, and ER-11 reference microphone that were used in the previous recordings were employed in this condition. As in the real restaurant, the orientation of the recording/playback system and KEMAR were such that neither was directly facing any walls in the simulation room.

Recording of the Noise for 0°/90° and 0°/180° Conditions

The recording of the output of the ER-11 omnidirectional reference microphone made in the restaurant was used for the simulations in the 0°/90° and 0°/180° conditions. These recordings were made in an IAC booth (dimensions = 12' L x 9.4' W x 7.5' H). The noise recording was played back through a single Tannoy Arena loudspeaker placed either 24 inches above KEMAR (90°) or 24 inches behind (180°) KEMAR. Again, the same equipment and procedures used in the restaurant were used to record these simulations.

Analysis of the Spectrum of the Restaurant Noise

The long-term speech spectrum of the restaurant noise used in the study was verified to be similar to that of the speech-shaped HINT noise (Figure 4).

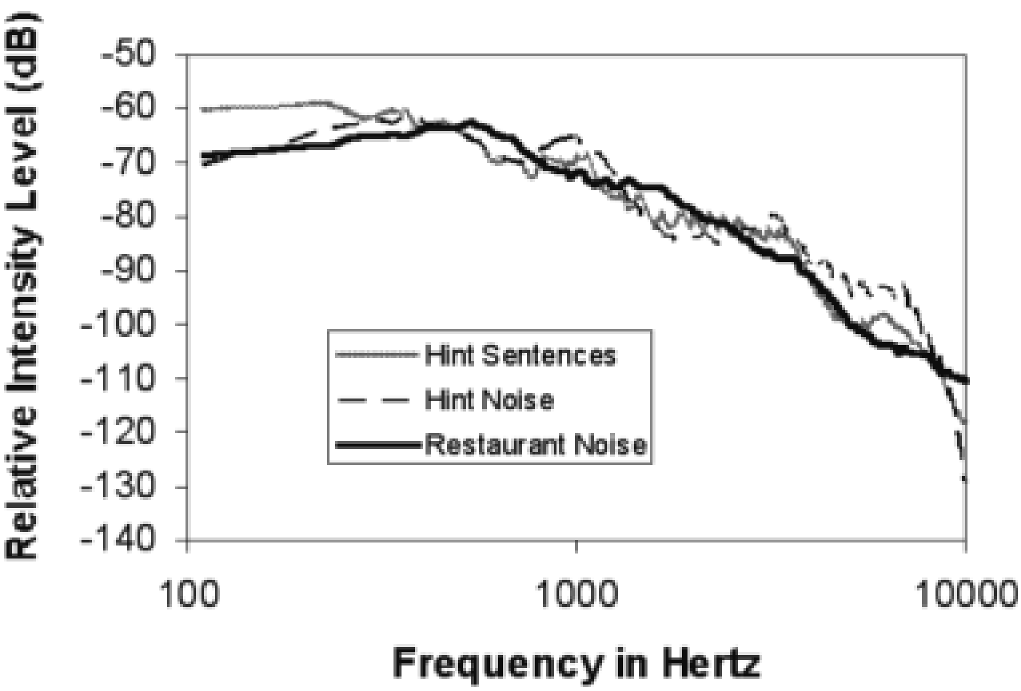

Figure 4. Frequency spectra of HINT sentences, HINT noise, and restaurant noise as measured by Sound Forge 4.5.

Production of Speech Recordings

The HINT sentences were chosen as the speech material to be used for the evaluation of the directional microphones for several reasons. These materials were specifically designed for measuring the reception threshold for sentences (RTS) in noise using an adaptive test procedure. The HINT material and procedure have been used in previous studies of directional microphone hearing aids. A sufficient number of lists were available for the design requirements of this study. This material provides a simple method for determining directional performance and benefit, without regard for variation of real-world SNRs. Normative data are available for listeners with normal hearing as well as sensorineural hearing loss (Nilsson et al, 1992; Nilsson et al, 1994b).

Initial plans called for speech materials to be recorded in each of the environments to be evaluated. However, ambient traffic noise from a nearby four-lane truck route and airport made it impossible to record the speech materials in the restaurant without distortion and with an acceptable SNR. Therefore, it was necessary to record all of the speech materials in a sound-treated room. A speaker-to-listener distance of 24 inches was chosen for evaluation. Although rather close, this distance was chosen to represent the distance at which a hearing aid user might position him- or herself in inorder to maximize the signal-to-noise ratio in a very difficult listening situation. Estimates of the critical distance were obtained in each recording environment (restaurant, R-SPACE room, and IAC room) by measuring the level of pink noise at several distances from the same Tannoy Arena loudspeaker used for delivering the pink noise calibration signal and the HINT sentences. Measurements were made under acoustic conditions similar to those present during recording sessions. In all rooms, a distance of 24 inches was well within the critical distance. Because the distance between KEMAR and the loudspeaker would have been within the critical distance in any of the test environments, recording the speech in the sound-treated room at this distance would yield a recorded test material similar to what would have been recorded in each of the environments. In addition, the direct-to-reverberant ratio at the 24-inch distance was greater 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 would not contaminate measurements of threshold when using the KEMAR-recorded speech and noise presented to the subjects.

Recordings of the HINT sentences were played from a single Tannoy Arena loudspeaker placed 24 inches in front of KEMAR (0° azimuth) in an Iac booth (dimensions = 12' L x 9.4' W x 7.5' H) and recorded through the three pairs of microphones mounted on KEMAR and through the ER-11 microphone placed above KEMAR.

Final Token Preparation

The goal of the 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. To accomplish this, the following procedures were followed:

1. The ER-11 reference IAC booth recording of the HINT sentences was re-recorded so that all sentences were concatenated, deleting any waveform 40 dB or more below the instantaneous peak level. The rms level of the sentences was then measured.
2. The ER-11 reference microphone recording of the restaurant noise was divided into 14 approximately 2.5-minute-long segments, and Sound Forge 4.5 (2000) was used to determine the rms level for each noise sample. Then, the rms level of each of the 14 ER-11 noise samples (in each environment) was adjusted to achieve equal rms of all noise samples.
3. Sentence and noise samples were recorded onto new tapes as test tokens to produce a four-track recording in which binaural hearing aid recordings of the HINT sentences presented at 0° azimuth in the IAC booth were aligned with binaural recordings of restaurant noise as recorded through the same hearing aid microphones under four test conditions. Each 20-sentence HINT list was matched to one of 12 possible noise segments using a 12 x 12 Hyper Greco Latin Square Design where each sentence and noise segment forms a unique pair. The four test conditions were :
   a. Live: Noise from real restaurant
   b. R-SPACE: Noise from simulator
   c. 0°/180°: Noise from single speaker at 180° azimuth, IAC booth
   d. 0°/90°: Noise from single speaker at 90° azimuth (overhead), IAC booth

This process resulted in a DTR four-track recording of HINT sentences (tracks 1 and 2; left/right) and noise segments (tracks 3 and 4; left/right). 1000 Hz calibration tones equaling 78 dB SPL were applied such that -3 dB VU equaled 75 dB SPL as the average level of the recorded HINT sentences and restaurant noise in each ear, at the FRP.

Prior to data collection, a pilot study with five normal-hearing young adults (ages 21 to 25) was completed to determine whether substituting restaurant noise for the noise provided as part of the HINT test would affect the reliability of the thresholds measured with the adaptive test procedure. The pilot study revealed a 95% critical difference similar to that obtained by Nilsson et al (1994a) for testing with either the original HINT noise or with the restaurant noise. As indicated previously, the restaurant noise was very similar in spectrum to the speech-spectrum shaped noise used in the standard administration of the HINT test.

METHOD

A repeated measures design was used in order to determine the effect of the four test environments for the three directional microphones.

Twelve listeners, ages 22 to 28 years, with bilaterally symmetrical normal-hearing and excellent (92-100%) speech recognition ability (NU-6) served as subjects. The number of subjects included in the study was determined based on a statistical power analysis. Estimates of the error variance obtained in the pilot study showed that for a statistical power of 0.8, a repeated measures design with 12 subjects per condition would result in an expected error probability of 0.034 for not detecting a difference of up to 1 dB between two conditions of greatest interest (e.g., live vs. R-SPACE). Thus, the experimental design was found to be reasonably powerful.

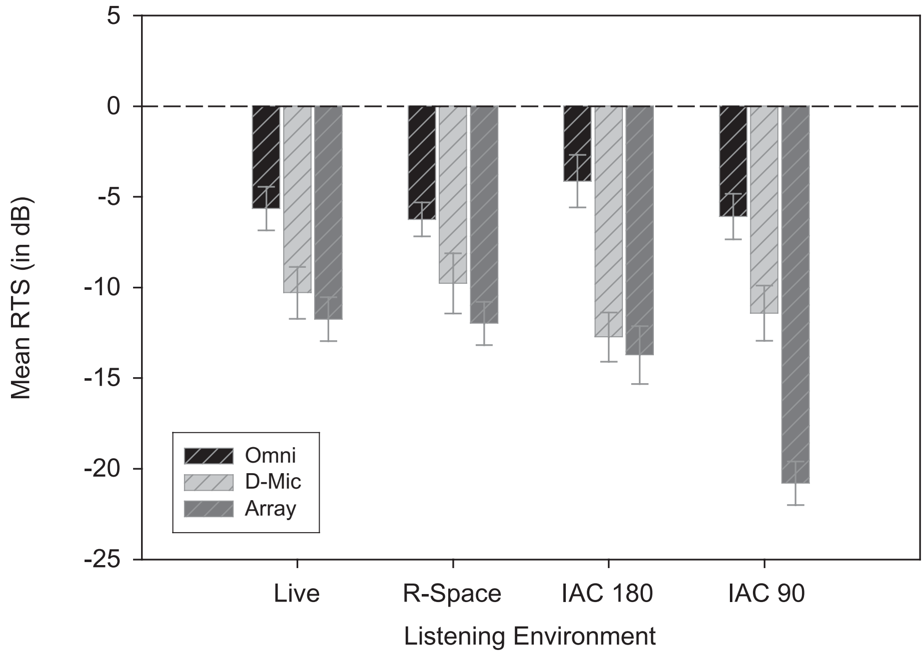

Figure 5. Mean field-referred RTSs (dB) required across three hearing aid microphone conditions and four noise delivery environments. Standard deviations are also shown.

Test Procedure

A Tascam DA-38 Digital Audio Tape Recorder was used to play the test stimuli. The output of the DA-38 was connected to two two-channel audiometers. 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 (GSI) 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.

The settings on the mixer were adjusted to yield a level of 75 dB SPL through the insert earphones (as measured in a 2-cc coupler) when the attenuator dials of both audiometers were set at 63 dB HTL and the calibration tone for each track of the test tape was set to “0” on the VU meter.

The test procedure and method of scoring recommended by Nilsson et al (1994b) were followed, except that the initial presentation level for the speech was started 20 dB below the noise level, rather than the recommended -10 dB RTS. This was necessary to avoid starting at an audible speech level for some of the directional hearing aid conditions. For this experiment, two ten-sentence blocks were used for each of the 12 test conditions. Test conditions were counterbalanced across participants. Two ten-sentence practice lists were presented before data collection.

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 75 dB SPL in each ear. The level of the speech was adjusted adaptively to estimate the RTS at which the sentences could be repeated correctly 50% of the time. Correct identification of each sentence was 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 sentences were presented at the same level bilaterally. An incorrect response resulted in the speech presentation level being raised bilaterally, and a correct response resulted in the speech presentation level being lowered bilaterally 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

Table 2. Mean Absolute RTS (dB) across Microphones and Environments

	Environment			
Microphone	Live	R-SPACE	0°/180°	0°/90°
Omnidirectional	-5.7	-6.2	-4.1*	-6.1
Supercardioid	-10.3	-9.8	-12.7*	-11.4
Hypercardioid	-11.7	-12.0	-13.7*	-20.8*

Note: Mean values marked with an asterisk indicate significantly different performance (p = 0.01) in that condition versus the live condition for the same microphone.

the 20th trial, then a hypothetical 21st trial would occur 2 dB higher. To calculate the RTS, the attenuator settings (presentation levels for the sentences) for trials 5 through 21 were averaged, and the audiometer dial setting for the noise was subtracted from this average.

RESULTS AND DISCUSSION

Mean RTS (in dB) and standard deviations for the three hearing aid microphones and four noise environments are shown in Figure 5 and Table 2. Inspection of the figure reveals that in all of the test environments, performance is poorest with the omnidirectional microphone and best with the hypercardioid microphone. Examination of the figure also reveals that the absolute threshold for a given microphone differs as a function of the test environment. Thus the three microphones are ranked similarly in the four environments (i.e., omnidirectional = poorest performance, hypercardioid = best performance, supercardioid = in between), but the RTS obtained by each microphone type differs with the environment.

A decision was made a priori to compare the mean for each microphone in the three experimental evaluation conditions to that in the live condition. The Bonferroni method of multiple comparisons (Dunn, 1961) was employed to determine whether significant differences existed between the mean performance for each microphone in the live condition versus that in each of the other three conditions. For this test, a difference of 1.4 dB was significant. Table 2 shows which means were found to be significantly different from the live condition for each microphone, while Table 3 illustrates the benefit achieved with each microphone type in the R-SPACE, 0°/180°, and 0°/90° conditions when compared to the live condition.

Results showed performance to be statistically identical for the live and R-SPACE conditions. For the live versus 0°/180° condition, significantly better performance was seen for the supercardioid (2.4 dB) and hypercardioid (2.0 dB) microphones in the 0°/180° condition as compared to the live condition. For the 0°/90° (overhead) condition, performance was similar to the live condition for the omnidirectional and directional microphones, but was very different for the hypercardioid microphones. The mean RTS obtained in the 0°/90° condition with the hypercardioid microphones was 9.1 dB better than performance obtained in the live condition. Omnidirectional performance was the same in all situations with the exception of slightly poorer performance (about 1.6 dB) in the 0°/180° 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.

Table 3. Benefit RTS (dB) Achieved with Each Microphone Type Re: the Live Condition

	Environment		
Microphone	R-SPACE	0°/180°	0°/90°
Omnidirectional	0.5	-1.6*	0.4
Supercardioid	-0.5	-2.4*	1.1
Hypercardioid	0.3	2.0*	9.1*

Note: Asterisks indicate a difference from the live condition of more than 1.4 dB (critical difference).

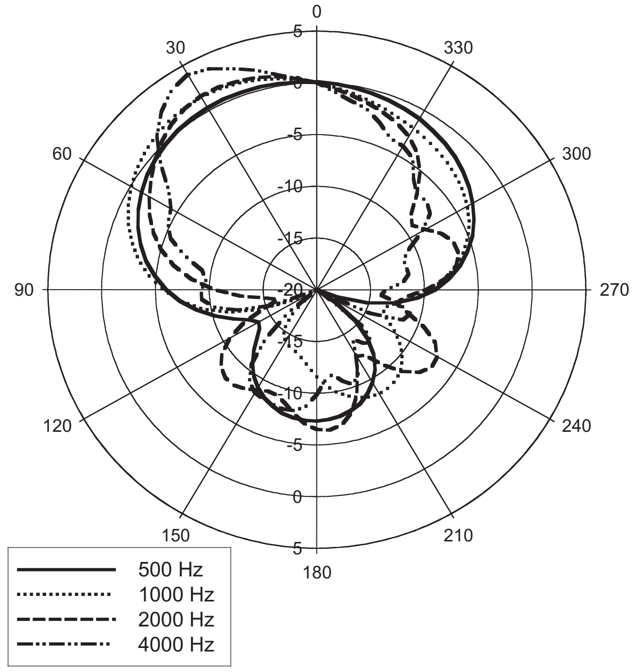

Figure 6. In situ polar plot at 500, 1000, 2000, and 4000 Hz of left supercardioid microphone (all microphones on KEMAR); measurement performed 24 inches from the loudspeaker.

Figures 6 and 7 show the in situ polar directional patterns of the super- and hypercardioid microphones used in this study (left microphone shown). Both microphones have noticeable rear lobes. The supercardioid microphone (Figure 6) has a very large pick-up pattern in the front hemisphere, and its nulls are located approximately at 120° and 265°, while the hypercardioid microphone (Figure 7) has a narrower pick-up pattern in the front hemisphere and deeper nulls at 90° and 270°.

According to research by Ricketts, "in an environment without reverberation, and given a particular hearing aid's polar directivity pattern with a signal of interest

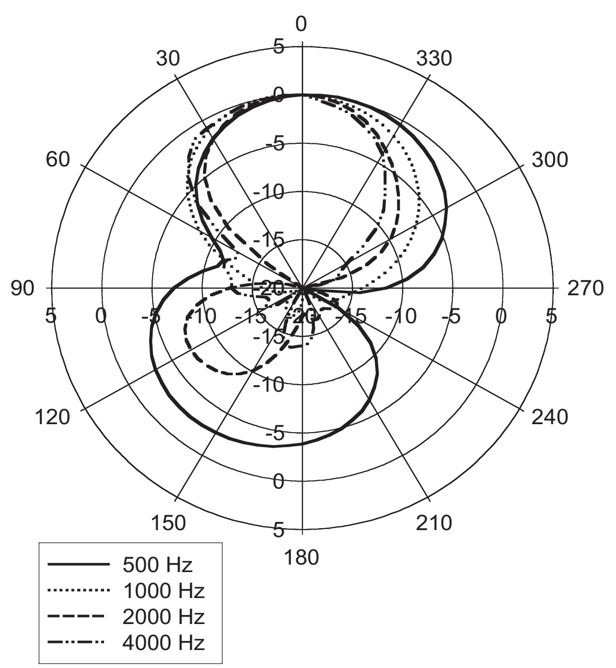

Figure 7. In situ polar plot at 500, 1000, 2000, and 4000 Hz of the left hypercardioid (array) microphone (all microphones on KEMAR); measurement performed at 24 inches from the loudspeaker.

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" (2000, p.202). Thus, it seems reasonable to assume that, in the case of the 0°/180° condition, better performance for both directional microphones occurred because noise was not present in the front hemisphere as it was in the live and R-SPACE conditions.

The slightly poorer performance (about 2 dB) of the omnidirectional microphones in the 0°/180° as opposed to the 0°/90° condition was initially puzzling, since manufacturer's specifications (in situ AI-weighted polar plot) for the supercardioid showed that, on the average, it should provide more noise rejection from directly behind (1.5 dB) as compared to 90° 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, it was assumed that the omnidirectional microphone would show no noise rejection overhead. However, post hoc spectral analysis of identical omnidirectional noise tokens recorded from behind and overhead revealed approximately 1 dB greater amplitude when the tokens for each channel were presented from behind versus from above (Figure 8).

As a crosscheck, anechoic AI-weighted one-third octave band polar responses were obtained for the left omnidirectional ITE microphone, using a chirp stimulus and with the test loudspeaker positioned 24" (same distance used for calibration and recording) above (90°) and behind (180°) KEMAR. Results (Figure 9) revealed about 1 dB more sensitivity in the behind condition, thus explaining most of the discrepancy. The authors suspect that the additional 1 dB difference is due to test-retest variability or some other unaccounted-for factor.

A large, divergent result from the live condition was obtained with the array microphones in the 0°/90° condition. This 9.1 dB improvement in threshold was due to the location of the single noise source directly above a wide, deep null (20 dB) at the midline of each array microphone (Figure 7). A listening check verified dramatic signal attenuation in both the horizontal and vertical planes of the microphones' midlines. For these particular microphones, placing a signal noise source overhead produced a contrived advantage. This type of test arrangement would be good 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 were not for the contrived array performance, it could be said that the 0°/90° condition produced similar

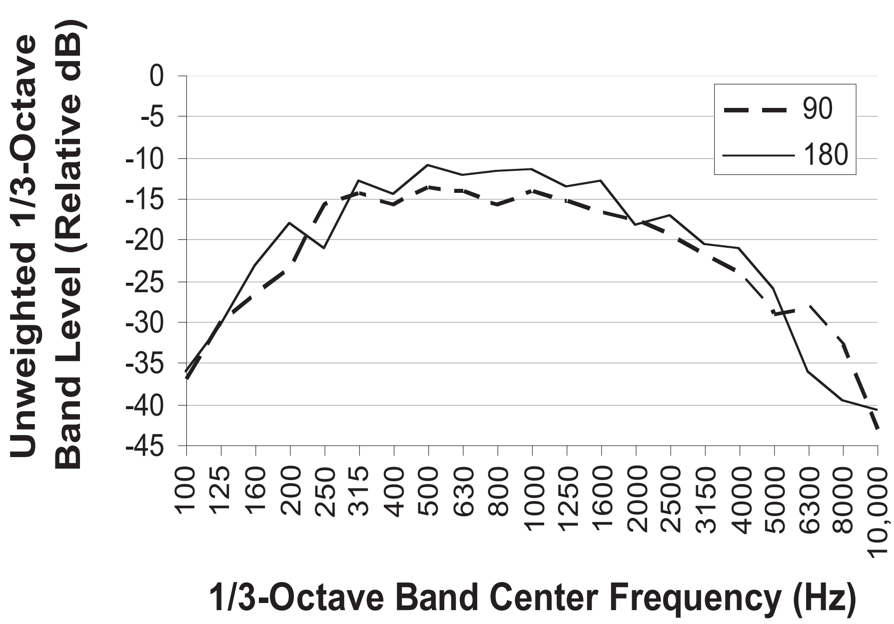

Figure 8. Comparison of spectra for left omnidirectional ITE noise tokens as recorded through KEMAR at 90 vs. 180 degrees.

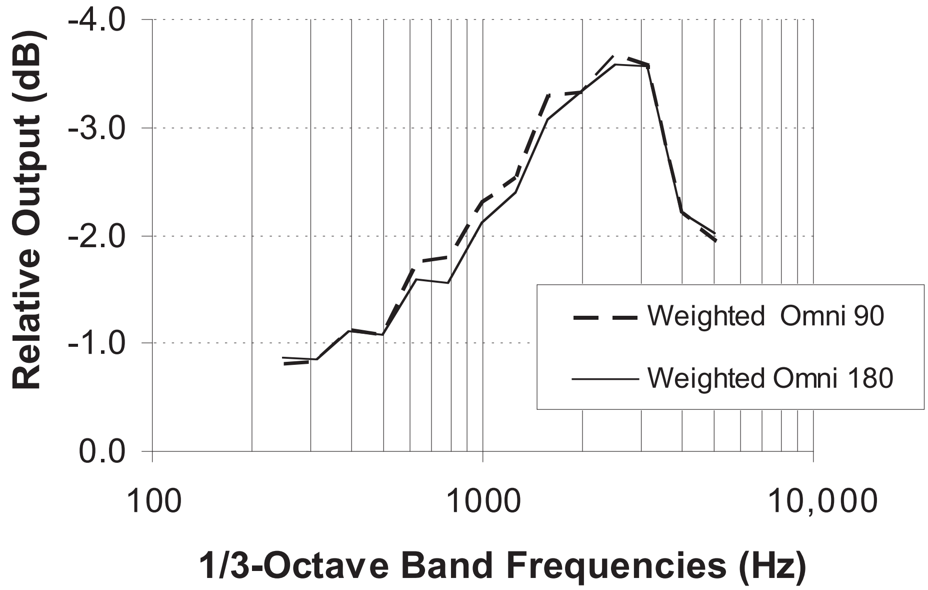

Figure 9. AI-weighted one-third octave band polar frequency responses for the left omnidirectional ITE microphone at 90° and 180° (IAC booth).

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. However, it would be important for the clinician to know ahead of time if the microphones being tested were devoid of midline nulls in the vertical plane.

In addition to providing information about the validity of various test environments, this study also provides information about the improvement that can be obtained with three different microphones of varying directivity. Killion et al (1998) suggested that the AI-DI value should predict the improved SNR to be obtained with a particular microphone. The improvement seen in the live and R-SPACE conditions with the supercardioid microphones as compared to the omnidirectional microphones was between 3.5 and 4.6 dB and is in rough agreement with the average improvement of AI-DI values for the directional microphones of 3.68 dB (anechoic). This also agrees with the results of Killion and his colleagues (1998) for the same restaurant. Benefit (over omnidirectional microphones) with the array microphones was 5.8 dB for the R-SPACE and 6.0 for the live condition. This was found to be similar in magnitude to the AI-DI of 6.9 dB obtained in the anechoic condition. Listener performance was similar to that obtained by Soede et al (1993b) with normal listeners. In that study, monaural SNRs of normal-hearing listeners improved approximately 5 dB with the array microphone compared to unaided performance. In the current study, the improvement from the omnidirectional ITE condition (used to simulate an unaided condition) was approximately 6 dB in the R-SPACE and live conditions.

In conclusion, the recording/simulation technique used in this study (R-SPACE) provided a reasonably accurate simulation of the live condition yielding equivalent performance in the real versus the reproduced restaurant. It is clear from the data that the R-SPACE technique is superior to traditional methods of evaluating directional microphones that use single loudspeakers to simulate a noisy environment. Of the three methods evaluated, the R-SPACE recording technique did the best job of predicting performance in the real restaurant. While the overhead loudspeaker simulation of a diffuse noise field was an adequate predictor of performance for the supercardioid microphone, it drastically overestimated the benefit of the array microphone. As described above, this was because the array had a large null in the center of the vertical plane. Thus, in evaluating directional microphones clinically, one should use a single overhead noise source configuration only with full knowledge of the three-dimensional polar pattern of the microphone in question; that is, the microphone does not have a midline null in the vertical plane. 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.

This investigation was limited in scope. Because the simulation was found to yield RTS performance through directional microphones consistent with real-world performance for normal-hearing listeners, further studies of this approach are warranted. Important issues not considered in this study should be investigated.

First, subjects with sensorineural hearing loss should be tested with hearing aids containing directional microphones to verify the validity of the technique on the population for whom it is intended.

Second, in several recent studies (e.g., Ricketts and Dhar, 1999; Ricketts, 2000), attempts have been made to simulate the real world using multiple loudspeakers and uncorrelated noise recordings. The recording/simulation technique used in the present study is unique in that multimicrophone recordings were obtained in the real world and then were reproduced in the clinical/laboratory setting. A validation study comparing the two approaches with recordings of the live condition would reveal the accuracy required for predicting benefit in the real world.

Third, because the noise recordings employed in the R-SPACE simulation technique were specific to a particular restaurant, the findings of this investigation cannot necessarily be generalized to other restaurants or other noisy environments. Future research is needed to catalogue a range of daily typical listening environments for listeners with hearing loss. A recorded "library" of sound using R-SPACE or other recording techniques could be developed to represent these environments.

Fourth, in the present study the recordings were purposely 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 of worship, lecture halls).

Acknowledgments. The first author would like to thank her dissertation committee, Arlene C. Neuman, chair, and Harry Levitt and Mead Killion, committee members. Many thanks are also due to Larry Revit and Robert Schulein who provided the audio engineering expertise, to David Preves who served as the outside examiner, to Jonathan Siegel of Northwestern University who provided access to his IAC booth for recording purposes, and to those individuals who served as listeners for several pilots and the main investigation. Finally, sincere appreciation is extended to Ruth Bentler and two anonymous peer reviewers for their helpful suggestions on an earlier version of this manuscript.

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