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A Procedure for Testing Speech Intelligibility in a Virtual Listening Environment
[Articles]

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Article Outline

Figures/Tables

Abstract TOP

Objective: The development of a test of virtual speech intelligibility in noise that enables assessment in typical, everyday listening situations. To eliminate extraneous confounding factors, digital signal processing was incorporated to simulate listening environments and source locations and allow presentation of stimuli via earphones.

Design: Source-to-eardrum transfer functions measured on KEMAR for various source locations in anechoic and reverberant environments were used to process monosyllabic words and speech-spectrum noise. Speech intelligibility was measured for three speech and noise configurations in two environments using an adaptive procedure to determine the signal-to-noise (S/N) ratio for 50% intelligibility.

Results: Normal-hearing listeners achieved 50% intelligibility of monosyllabic words at significantly lower S/N ratios in a virtual anechoic environment than in a virtual reverberant environment. Speech intelligibility improved significantly in both environments when the speech and noise sources were separated, but the intelligibility gain in the anechoic environment was four times larger than in the reverberant environment.

Conclusions: This test is easy to administer and score, and it provides a means for measuring: 1) the effects of separating speech and noise sources and 2) the effects of reverberation on speech intelligibility in noise while eliminating confounding factors such as calibration.

Measures of speech understanding and word recognition were the first procedures used to formally and informally evaluate hearing sensitivity and the impact of hearing impairment on communication **(Penrod, 1994)**. Presently, there are many tests available using various test materials to evaluate speech understanding of children and adults in quiet and in noise. These tests continue to be an integral part of the audiological assessment, providing valuable, objective information about the ramifications of an individual's hearing impairment on communication and the potential benefits of remediation, especially when testing is conducted in the presence of noise.

In general, the speech and noise stimuli used in these clinical tests are presented via earphones or loudspeakers to the listener seated in a sound-treated room. This presents a major drawback because it does not enable an assessment of speech intelligibility in listening environments typical of everyday communication situations. In addition, these traditional test paradigms do not provide a means for assessing the effects of reverberation on speech intelligibility in noise.

It is well known that the ability to understand speech in noise improves when the speech and noise sources are spatially separated (e.g.,**Bronkhorst & Plomp, 1990; Dirks & Wilson, 1969; Hirsh, 1950**). This ability, often referred to as the "cocktail party effect," is based on sensitivity to binaural differences, in particular to interaural differences in the intensity and in the time of arrival of sounds reaching the two ears. When the source locations of the speech and noise are the same, the interaural differences are similar; but, when the source locations are different, the speech and noise stimuli each have distinct interaural differences. These differences are believed to be the basis for improved speech intelligibility when the speech and noise sources are separated **(Bronkhorst & Plomp, 1988; Dirks & Wilson, 1969)**. These investigators have measured speech intelligibility in noise using stimuli created by recording from a dummy head in an anechoic room; then the stimuli are played back to the listeners via earphones. This procedure restricts the possible test conditions and stimuli to those that have been prerecorded. In the procedure described in this paper, it is not necessary to record stimuli for each condition prior to testing.

In this paper a new procedure is described for measuring speech intelligibility in noise in a virtual auditory environment. The approach used in developing this test is similar to that used in our virtual localization test **(Besing & Koehnke, 1995)**. Using state-of-the-art, digital-signal-processing techniques, both anechoic and reverberant listening environments are simulated. In addition, source locations are simulated so that the speech and noise originate from either the same or spatially separate locations. As described for the virtual localization test**(Besing & Koehnke, 1995)**, simulating the listening environments and source locations has many advantages. First, it makes it possible to ascertain the characteristics of the signals being presented to the listeners by analyzing the processed signals. Second, it eliminates the problems of calibration that occur when testing in the sound field. Third, it is no longer necessary to ensure that the sound sources and the listeners are in exactly the same locations for each test session. Fourth, it eliminates problems with other subject variables, such as head movements, that are difficult to control in a sound-field test environment. Finally, the test environments and combinations of test stimuli and locations within an environment are not limited to those described here. Other source locations and environments can be tested by measuring head-related-transfer functions for the desired locations in the particular environment and convolving them with the test stimuli.

Both anechoic and reverberant test conditions are included because it is known that the reverberation encountered in typical listening situations diminishes the binaural cues used in understanding speech in noise. It has been shown that speech perception in noise is degraded by reverberation, especially for individuals with impaired hearing (e.g., **Gelfand & Hochberg, 1976; Hawkins & Yacullo, 1984; Helfer & Wilber, 1990**). However, it cannot be determined from these studies whether the effects of noise and reverberation decrease when the speech and noise sources are spatially separated. The virtual speech intelligibility test described here includes measurement of performance when the speech and noise sources are the same and when they are spatially separate.

The ability to measure speech intelligibility in various environments and with the speech and noise sources in either the same or spatially distinct locations has many potential research and clinical applications. We have been using these procedures in the laboratory to both evaluate the effects of recurrent otitis media on understanding speech in noise and to compare the effects of monaural and binaural amplification on understanding speech in noise. After a description of the test development and methodology, some results obtained using this virtual speech intelligibility in noise test with normal hearing adults are presented to illustrate the application of this new test procedure.

Methods TOP

Speech intelligibility is measured in noise in both anechoic and reverberant, simulated, sound-field environments using phonetically balanced, monosyllabic words spoken by a male with a general American dialect. By simulating the listening environments and source locations from which the speech and noise are presented, and delivering them via earphones, extraneous confounding factors such as sound-field calibration and head movements are eliminated. It should be noted that, in the reverberant condition, the stimuli are clearly externalized. Both anechoic and reverberant listening conditions are included because the natural reverberation occurring in everyday listening situations tends to diminish the binaural cues used in understanding speech in noise. Phonetically balanced, monosyllabic words are used because they are the speech stimuli employed most often in clinical testing and because they provide a large number of items for the stimulus pool. With a large stimulus set, multiple tests for various conditions can be conducted with minimal repetition of words. The use of monosyllabic words also allows for rapid test administration.

The phonetically balanced words are selected from a subset of Egan's (1948) 1000-word set that were digitized on a 286 microcomputer using a 20kHz sampling rate and transferred to a DEC micro-VAX for further signal processing. For testing adults, an 880-word subset of Egan's 1000-word set is used; for testing children a 435-word subset of Egan's set is used 1. Prior to the actual testing, all the words were processed for each of the listening conditions described below. When the test is administered, each word is selected randomly from the age appropriate subset of words. Words are selected without replacement for each test run. Although there is some repetition of words across listening conditions, learning is not considered to be a factor because of the large word pool and the random ordering of words within lists.

The background noise used in these measurements is speech-spectrum noise that was digitally generated on a DEC micro-VAX computer with a 20kHz sampling rate. The spectrum of the noise was shaped to be the same as the long-term speech spectrum. Twenty independent samples of noise were generated for each of the experimental conditions described below. Each noise sample has a duration of 500 msec. The noise is gated on first, followed by the target word selected for the particular trial. The speech and noise are gated off simultaneously.

Speech intelligibility is measured in noise for three speech and noise listening configurations in an anechoic and a reverberant environment for a total of six experimental conditions. The speech and noise configurations include the following pairs in the horizontal plane: 1) both speech and noise at 0^o, 2) speech at 0^o and noise at $+90^o$ (near the right ear), and 3) speech at 0^o and noise at -90^o (near the left ear). These configurations were chosen because they sample the most commonly encountered listening situations.

All signals are presented via matched, TDH-49P earphones. The sound-field conditions were simulated using source-to-eardrum transfer functions measured using a broadband stimulus in actual sound-fields for each source location in each environment using the KEMAR manikin (Burkhard & Sachs, 1975). In the reverberant room, reverberation time was approximately 0.25 sec below 800 Hz and 0.4 sec above that frequency2. To compensate for KEMAR's ear canal resonance, an equalization filter was used (Killion, 1979). The signals for all the experimental conditions were either processed on a DEC micro-VAX computer (using ISPUD®) or processed directly on a 486 microcomputer (using DADiSP®) and then transferred to a 386 or 486 microcomputer for presentation to the subjects.

As shown in Figure 1, each digitized speech and noise signal is convolved with the appropriate source-to-eardrum transfer function to generate a single speech and noise output for each ear: S_r and N_r for the right ear and S_1 and N_1 for the left ear. These digital stimuli are transferred to a digital-to-analog converter(Tucker-Davis Technologies, DA1), low-pass filtered, attenuated to the desired level, and presented to the subjects via earphones. When presented binaurally, these stimuli create the virtual locations in the anechoic and reverberant environments.

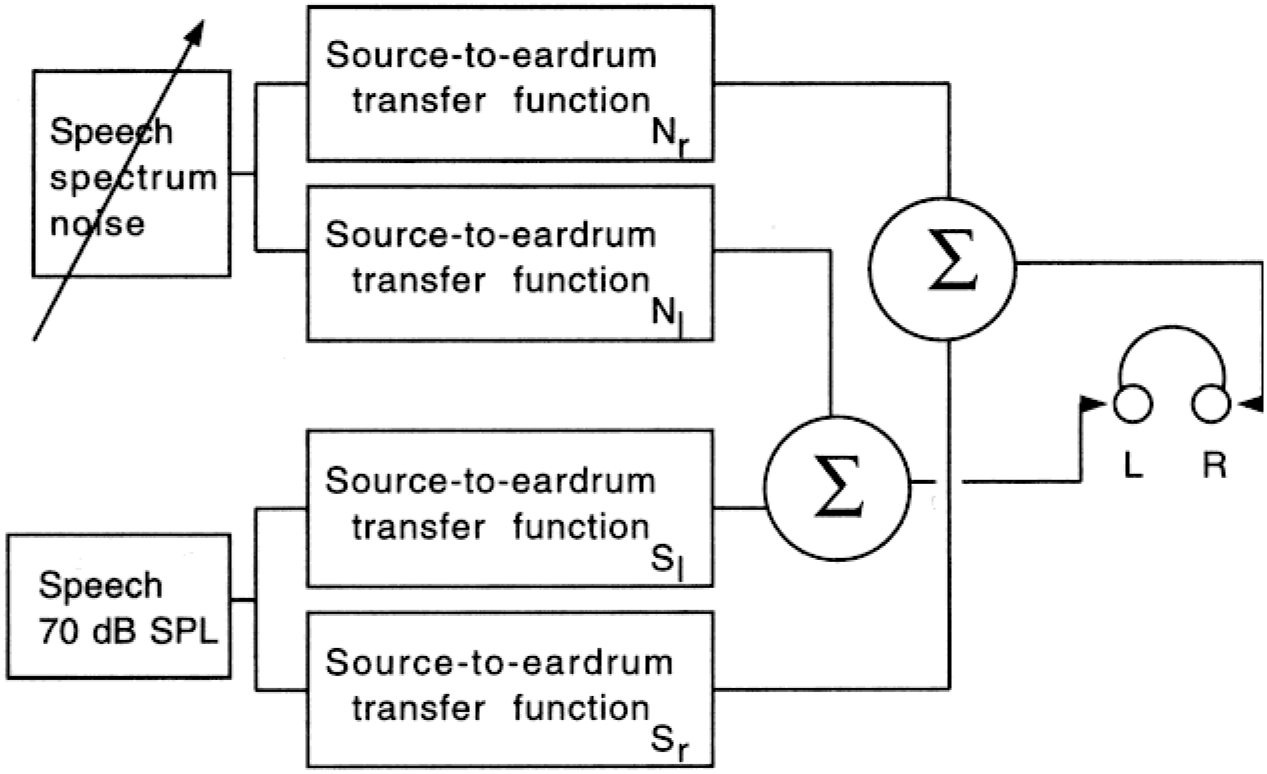

Figure 1. Block diagram of signal processing used to generate the virtual speech and noise stimuli used in the experiment.

In all of the speech-in-noise conditions, the speech is presented at a normal conversational level of 70 dB SPL. Speech intelligibility in noise is measured using a single-interval, adaptive procedure in which the input noise level is varied to find the level needed for 50% correct words (e.g.,**Bode & Carhart, 1974; Elliot, Connors, Kille, Levin, Ball, & Katz, 1979**). Each experimental run includes eight reversals, with the last six averaged to calculate the intelligibility threshold. The noise level is varied in 4-dB steps for the first two reversals and then in 2-dB steps for the remaining six reversals. The subject and experimenter are seated in a sound-treated room. The subject repeats each word he or she hears to the experimenter. The experimenter indicates whether or not the subject said the correct word and also enters the subject's response on the keyboard. If necessary the experimenter can request verification of the word from the subject to be certain that he or she understood the subject's response correctly. An average experimental run takes 2 to 3 minutes and includes 15 to 25 words.

Seven individuals, ages 15 to 33 yr (mean age 24 yr), participated as subjects in this study. All of the subjects had normal hearing (thresholds 20 dB HL or better) at octave frequencies from 250 through 8000 Hz. Each subject completed five repetitions of each of the three speech and noise listening configurations in the anechoic and reverberant environments in random order for a total of 30 experimental runs. Test words for all of the subjects were chosen from the adult list. Data were obtained in one or two sessions lasting from 30 to 90 minutes.

Results TOP

Results obtained for the individual subjects are provided in**Table 1**. The table includes the 50% intelligibility threshold S/N ratios averaged across the five repetitions for each of the three listening configurations for each of the two listening environments. In addition, the standard deviation for each subject in each condition is provided, and the intelligibility advantage obtained by each subject when the speech and noise sources are spatially separated is indicated. Overall it can be seen that the threshold S/N ratios are lower (better) in the anechoic environment than in the reverberant environment. Further, the thresholds are lower when the speech and noise sources are separated (0/+90 and 0/-90) than when the source locations are the same (0/0).

	Anechoic Environment								
	Speech/Noise Condition Threshold S/N (dB)						Intelligibility Gain (dB)		
	0/0		0/+90		0/−90				
Subject	Mean	SD	Mean	SD	Mean	SD	0/0 − 0/+90	0/0 − 0/−90	Average
1	−0.60	3.1	−6.47	3.0	−15.20	4.3	5.9	14.6	10.2
2	0.27	4.4	−12.30	5.7	−16.33	2.5	12.5	15.6	14.1
3	−1.07	4.3	−9.40	3.1	−12.47	5.3	8.2	11.4	9.8
4	2.87	2.9	−11.20	7.3	−14.13	4.1	14.1	17.0	15.5
5	−0.53	2.2	−13.50	4.1	−17.05	2.7	13.0	16.5	14.8
6	−2.00	4.4	−15.70	6.2	−22.20	3.6	13.7	20.2	16.9
7	−0.70	1.0	−14.50	4.9	−14.33	3.8	14.5	14.3	14.4
Mean	−0.25		−11.87		−15.96		11.69	15.66	13.67
SD	1.54		3.17		3.14		3.31	2.72	

	Reverberant Environment								
	Speech/Noise Condition Threshold S/N (dB)						Intelligibility Gain (dB)		
	0/0		0/+90		0/−90				
Subject	Mean	SD	Mean	SD	Mean	SD	0/0 − 0/+90	0/0 − 0/−90	Average
1	7.73	2.0	2.40	1.6	2.53	2.6	5.3	5.2	5.3
2	4.80	5.4	1.53	3.6	−3.47	3.4	3.3	8.3	5.8
3	6.80	3.6	4.73	4.8	4.20	4.2	2.1	2.6	2.3
4	5.00	3.4	0.27	3.0	2.60	2.0	4.7	2.4	3.6
5	0.00	1.3	−4.00	3.4	−5.82	2.3	4.0	5.8	4.9
6	1.07	2.9	−1.33	3.1	−2.80	4.6	2.4	3.9	3.1
7	1.40	3.5	−1.42	3.2	−2.75	4.7	2.8	4.2	3.5
Mean	3.83		0.31		−0.79		3.52	4.62	4.07
SD	3.01		2.88		3.82		1.22	2.04	

TABLE 1. Average threshold signal-to-noise (S/N) ratios (dB) and standard deviations for 50% intelligibility and intelligibility gain (dB) due to signal and noise source separation for individual subjects

TABLE 1. Average threshold signal-to-noise (S/N) ratios (dB) and standard deviations for 50% intelligibility and intelligibility gain (dB) due to signal and noise source separation for individual subjects

A repeated measures analysis of variance (ANOVA) indicates that the listeners performed significantly better in the anechoic environment than in the reverberant environment (F [1, 6] = 185, $p < 0.01$) when all data were collapsed across the different listening conditions. It was expected that performance would be poorer in the reverberant environment because binaural cues are less robust in a reverberant environment.

The ANOVA also reveals that the listeners' overall performance (collapsed across the different environments) changed significantly as a function of the listening configuration (F [2, 12] = 129, $p < 0.01$), but this change in performance was not the same in the two environments(F [2, 12] = 37, $p < 0.01$). Scheffe planned comparisons were made between all possible pairs of noise locations. The mean threshold S/N ratio is significantly better when the noise source is on the right than when it is straight ahead ($p < 0.01$), and it is significantly better when the noise source is on the left than when it is straight ahead ($p < 0.01$). The Scheffe comparisons also indicate a significantly lower average threshold S/N ratio when the noise source is on the left than when it is on the right ($p < 0.01$).

As can be seen in Table 1, there is a wide range of performance among these normal-hearing listeners. Nonetheless, the pattern of results is the same; all the subjects have poorer performance in the reverberant environment and poorer performance when the speech and noise are not spatially separated. The standard deviations indicate that, in general, the intrasubject and intersubject reliability were quite good. In most cases, the individual standard deviations were less than 2 steps or 4 dB; the group standard deviations were all less than 4 dB. Individual variability tended to be larger in the anechoic environment than in the reverberant environment.

The results of the ANOVA indicate no significant effect of repetition(F [4, 24] = 1.5, p = 0.2). This supports the notion that although there is some repetition of words across listening conditions, learning is not affecting the results. This may be due to the large word pool and the random selection of words for each experimental run. To illustrate the absence of a learning effect in this experiment, the data for a representative subject (subject 5) are plotted in Figure 2 for each condition. Clearly, there was no improvement in performance over the course of the experiment in any condition.

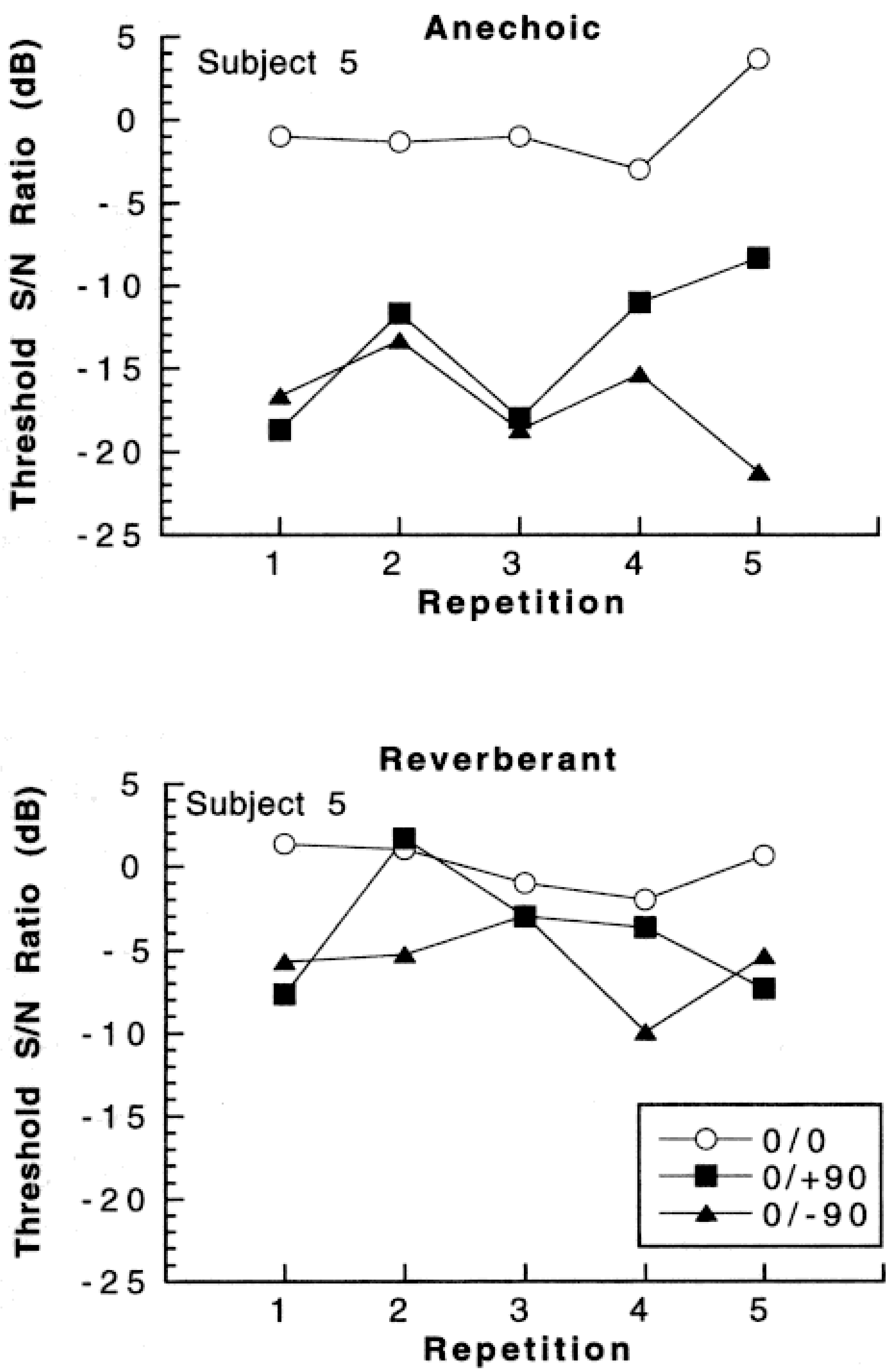

Figure 2. Threshold signal-to-noise (S/N) ratios needed for 50% correct as a function of repetition for Subject 5. The top panel shows results for the anechoic environment and the bottom panel shows the results for the reverberant

environment. The unfilled circles are for the 0/0 condition when the speech and noise sources are in the same location; the filled squares and triangles are for the 0/+90 and 0/-90 conditions, respectively, when the speech and noise sources are separated.

The average and individual subject intelligibility gains achieved when the speech and noise sources were separated are shown in**Table 1**. These values were calculated using the average threshold S/N ratio for each condition for each subject (or for the mean across subjects) across the five repetitions. Although this is a derived measure, it is the value typically reported by investigators. Therefore, the effects of listening condition and subjects on the intelligibility gain were examined with a 2-way, repeated measures ANOVA. As expected, the results are consistent with the findings of the ANOVA described above for the actual threshold S/N ratios measured in the experiment. There is a significantly larger intelligibility gain obtained in the anechoic environment than in the reverberant environment (F [1, 6] = 76, $p < 0.01$). The results also indicate a significant difference between the gain in intelligibility when the noise source is moved to the right (0/0 to 0/+90) than when the noise source is moved to the left (0/0 to 0/-90) (F[1, 6] = 15, $p < 0.01$) for all data collapsed across environment. Six of the seven subjects obtained a considerably larger intelligibility gain in the anechoic environment when the noise source was moved to the left than when it was moved to the right; five of the seven subjects obtained a larger intelligibility gain in the reverberant environment when the noise source was moved to the left. This was an unexpected result and we have no explanation for it.

The effects of listening environment and speech and noise location are apparent in **Figure 3**, which shows the average threshold S/N ratios across subjects for each of the three listening conditions in each environment. Clearly, performance is better in the anechoic condition than in the reverberant condition regardless of the noise source location.**Figure 3** also illustrates the improvement in performance and gain in intelligibility achieved by separating the speech and noise source locations. As the bar graph shows, thresholds are lower when the noise source was located at -90^{0} or $+90^{0}$ than they are when the noise source was located at 0^{0} in both environments.**Table 1** also indicates the intelligibility gain in decibels when the noise source is moved to the left (-90^{0}) or to the right ($+90^{0}$) side. The gain was clearly greater in the anechoic environment than in the reverberant environment, and, as indicated above, this difference is significant.

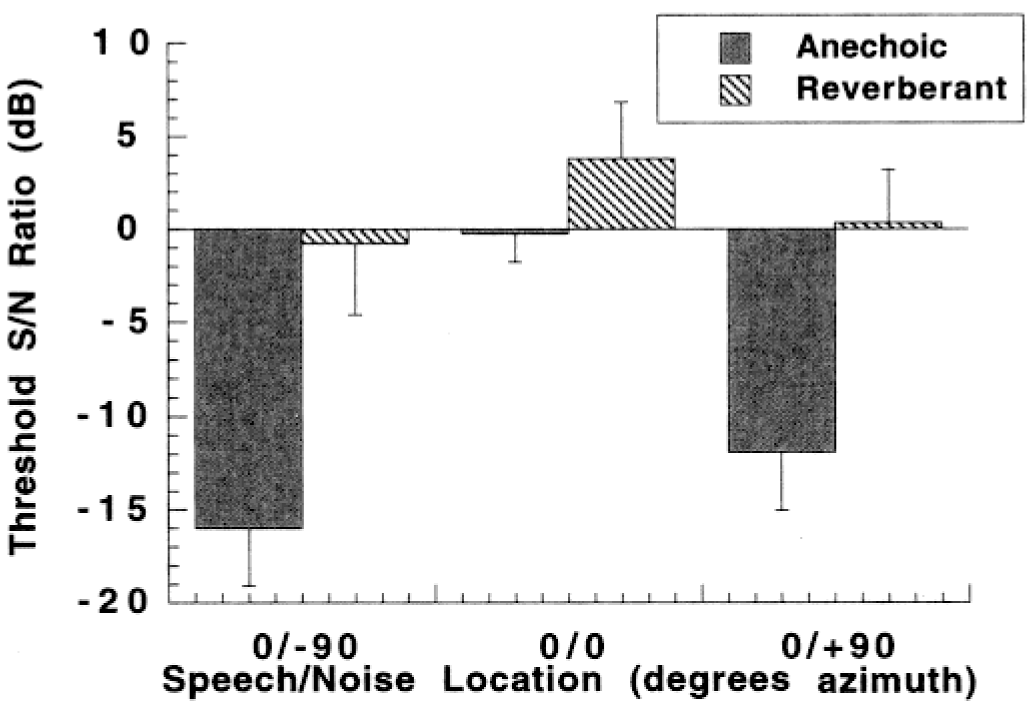

Figure 3. The threshold signal-to-noise (S/N) ratio(dB) averaged across subjects is plotted as a function of the speech/noise location in the anechoic (dark bars) and the reverberant (slashed bars) environments.

Discussion TOP

The virtual speech intelligibility in noise test described here is easy to administer, score, and interpret, and it provides a measure of speech intelligibility in conditions comparable to typical everyday listening situations. By processing the stimuli to simulate different listening environments and various speech and noise source locations and then presenting them via earphones, many of the problems inherent in free-field testing are avoided. As indicated by the results obtained with a group of normal-hearing listeners, the test can effectively assess the impact of both reverberation and signal and noise source location on speech intelligibility.

As expected, performance was significantly affected by listening environment, with higher S/N ratios required for 50% correct word intelligibility in the reverberant environment than in the anechoic environment. This is in agreement with other studies in which speech perception was measured as a function of increasing reverberation time (e.g.,Gelfand & Hochberg, 1976; Hawkins & Yacullo, 1984; Helfer & Wilber, 1990; Nabelek & Pickett, 1974a, 1974b; Nabelek & Robinson, 1982). In most of these studies, speech intelligibility was compared for monaural and binaural listening conditions(with and/or without hearing aids) to determine the effect of reverberation on the binaural advantage (the difference in the S/N ratio needed for equivalent performance when listening with one versus two ears) (Gelfand & Hochberg, 1976; Hawkins & Yacullo, 1984; Nabelek& Pickett, 1974a, 1974b; Nabelek & Robinson, 1982). In the present study, however, stimuli were always presented binaurally and performance with the same speech and noise source location was compared with performance when the speech and noise sources were separated.

The results reveal a fairly wide range of performance for this group of normal-hearing listeners as evidenced by the standard deviations shown inTable 1. However, the pattern of results was the same for all the subjects. The variability among subjects has also been found to be fairly large in other studies in which speech intelligibility was measured using monosyllabic words (Dirks & Wilson, 1969; Gelfand & Hochberg, 1976; Nabelek & Robinette, 1978). However, it is difficult to make a direct comparison because these studies measured performance in percent correct rather than by determining the S/N ratio required for 50% correct intelligibility. It is interesting to note that a wide range of performance among normal-hearing listeners has also been observed for other binaural tasks (Koehnke, Colburn & Durlach, 1986; McFadden, Jeffress & Russell, 1973).

The overall results of this study are in good agreement with other investigations of the effect of separating speech and noise sources on speech intelligibility in noise. Nonetheless, there are some differences in the magnitude of the intelligibility gains and the threshold S/N ratios obtained in the various studies (Bronkhorst & Plomp, 1988, 1990; Dirks & Wilson, 1969; MacKeith & Coles, 1971). It is difficult to compare the results obtained byMacKeith and Coles (1971) with the present study because they measured binaural (both ears open) and monaural (one ear plugged and muffed) speech intelligibility in noise and reported only the binaural listening advantage, not the measured threshold S/N ratios. However, the studies of Bronkhorst and Plomp (1988, 1990) andDirks and Wilson (1969) are comparable.

The intelligibility gains obtained in the reverberant environment by the normal-hearing listeners in the present study are 1.5 to 2 dB smaller than those measured by Bronkhorst and Plomp (1990) for their normal-hearing listeners. The threshold S/N ratios measured by Bronkhorst and Plomp are also 9 to 12 dB lower than those measured in this study. There are a number of factors that may account for these differences.Bronkhorst and Plomp (1990) made their measurements in a real reverberant room, whereas we simulated the environment using signal processing. It is possible that the signal processing affected the signals in ways that were not apparent to us. Therefore, the results obtained in this study might not be obtained if the measurements were made in a real sound field. To address this issue, a study is presently underway comparing speech intelligibility in noise measured with the processed stimuli with speech intelligibility in noise measured in a real field with equivalent reverberation. Another possible explanation for the differences is that Bronkhorst and Plomp used Dutch sentences as speech stimuli, whereas we used monosyllabic words. Also, Bronkhorst and Plomp varied the level of the speech to find the 50% intelligibility threshold, whereas we varied the noise level, and they used a lower overall presentation level than was used in the present study.

In another study, Bronkhorst and Plomp (1988) measured speech intelligibility in noise of normal-hearing adults using stimuli recorded through KEMAR's ears in an anechoic room. Results of this experiment reveal a maximum gain in intelligibility of 10.1 dB when the speech and noise sources were separated by 90^o (and the speech source was located at 0^o). This gain is somewhat smaller than the average 13.7 dB gain(11.7 dB gain when the noise was at $+90^o$ and 15.7 dB gain when the noise was at -90^o) measured for the same environment and source locations in the present study. The difference in the intelligibility gains measured in the two studies results primarily from the fact that there is a 6 dB difference between the present study and the Bronkhorst and Plomp study in the average threshold S/N ratio when the speech and noise were both located at 0^o; but there is little difference between the thresholds in the other comparable listening conditions. This 6 dB difference is probably due to the way in which the stimuli for the two studies were prepared (direct recording by Bronkhorst and Plomp versus digital signal processing using KEMAR's source-to-eardum transfer functions in the present study).

Both the threshold S/N ratios and the intelligibility gains measured by**Dirks and Wilson (1969)** differed from those in the present study. For phonetically balanced words, their overall 50% threshold S/N ratios are higher than those obtained in the present study and their intelligibility gain is only 3 to 4 dB compared with our measured gains of 11.7 to 15.7 dB, although, as in the present study, they kept the intensity level of the speech constant at 70 dB SPL and varied the noise level. But, although they measured performance for both speech and noise at 0^{o}, the only condition they used that is comparable to our condition with the speech at 0^{o} and the noise at $+90^{o}$ or -90^{o} is with the noise at 0^{o} and the speech at -90^{o}. This is more like a monaural listening condition with the speech audible in only one ear, and thus, the absolute thresholds and gains obtained in their study probably cannot be directly compared with our results. Dirks and Wilson recorded their speech and noise stimuli through a manikin's head in an anechoic room.

In summary, the results obtained for the normal-hearing listeners in this study are in reasonably good agreement with other studies in the literature, taking into account differences in stimulus type, stimulus presentation mode, and signal processing procedures. Clearly, the virtual speech intelligibility in noise test described here provides a means for evaluating the effects of listening environment and the effects of speech and noise source separation on speech intelligibility in noise without the confounding effects involved in free-field testing. Further testing is presently being conducted to develop this test for clinical use.

Acknowledgments: TOP

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References TOP

Besing, J., & Koehnke, J. (1995). A test of virtual auditory localization. *Ear and Hearing, 16*, 220-229.
[Context Link]
Bode, D. L., & Carhart, R. (1974). Stability and accuracy of adaptive tests of speech discrimination. *Journal of the Acoustical Society of America, 56*, 963-970.
[Context Link]
Bronkhorst, A. W., & Plomp, R. (1988). The effect of head-induced interaural time and level differences on speech intelligibility in noise. *Journal of the Acoustical Society of America, 83*, 1508-1518.
[Context Link]
Bronkhorst, A. W., & Plomp, R. (1990). A clinical test for the assessment of binaural speech perception in noise. *Audiology, 29*, 275-285.
[Context Link]
Burkhard, D. M., & Sachs, R. M. (1975). Anthropometric manikin for acoustic research. *Journal of the Acoustical Society of America, 58*, 214-222.
[Context Link]
Dirks, D. D., & Wilson, R. A. (1969). The effect of spatially separated sound sources on speech intelligibility. *Journal of Speech and Hearing Research, 12*, 5-38.
[Context Link]
Egan, J. P. (1948). Articulation testing methods. *Laryngoscope, 58*, 955-991.
[Fulltext Link] [CrossRef] [Context Link]
Elliot, L., Connors, S., Kille, E., Levin, S., Ball, K.,& Katz, D. (1979). Children's understanding of monosyllabic nouns in quiet and in noise. *Journal of the Acoustical Society of America, 66*, 12-21.
[Context Link]
Gelfand, S. A., & Hochberg, I. (1976). Binaural and monaural speech discrimination under reverberation. *Audiology, 15*, 72-84.
[Medline Link] [Context Link]
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.
[Context Link]

Helfer, K. S., & Wilber, L. A. (1990). Hearing loss, aging, and speech perception in reverberation and noise. *Journal of Speech and Hearing Research, 33*, 149-155.
[Context Link]
Hirsh, I. J. (1950). The relation between localization and intelligibility. *Journal of the Acoustical Society of America, 22*, 196-200.
[Context Link]
Killion M. (1979). Equalization filter for eardrum-pressure recording using a KEMAR manikin. *Journal of the Audio Engineering Society, 27*, 13-16.
[Context Link]
Koehnke, J., Colburn, H. S., & Durlach, N. I. (1986). Performance in several binaural-interaction experiments. *Journal of the Acoustical Society of America, 79*, 1558-1562.
[Context Link]
MacKeith, N. & Coles, R. (1971). Binaural advantages in hearing of speech. *Journal of Laryngology & Otology, 85*, 213-232.
[Context Link]
McFadden, D., Jeffress, L. A., & Russell, W. E. (1973). Individual differences in sensitivity to interaural differences in time and level. *Perception and Motor Skills, 37*, 755-761.
[Context Link]
Nabelek, A. K., & Pickett, J. M. (1974a). Reception of consonants in a classroom as affected by monaural and binaural listening, noise, reverberation and hearing aids. *Journal of the Acoustical Society of America, 56*, 628-639.
[Context Link]
Nabelek, A. K., & Pickett, J. M. (1994b). Monaural and binaural speech perception through hearing aids under noise and reverberation with normal and hearing-impaired listeners, *Journal of Speech and Hearing Research, 17*, 724-739.
[Context Link]
Nabelek, A. K., & Robinette, L. (1978). Reverberation as a parameter in clinical testing. *Audiology, 17*, 239-259.
[Context Link]
Nabelek, A. K., & Robinson, P. K. (1982). Monaural and binaural speech perception in reverberation for listeners of various ages. *Journal of the Acoustical Society of America, 71*, 12142-1248.
[Context Link]
Penrod, J. P. (1994). Speech threshold and recognition/discrimination testing. In J. Katz (Ed.), *Handbook of clinical audiology*. Baltimore: Williams and Wilkins.
[Context Link]
[1] A subset of words considered to be familiar and appropriate for children was chosen from the 880-word list used for the adults. For example, the words *creed* and *debt* are on the adult list but not on the children's list, whereas the words*book* and *dog* are on both lists. [Context Link]

[2] A typical conference room, approximately 16.5′ × 20′ × 9′ with a linoleum floor, tables, chairs, and an acoustically tiled ceiling, was used to measure the HRTFs for the reverberant environment. [Context Link]