Veterans Administration

Journal of Rehabilitation Research
and Development Vol. 24 No. 4
Pages 75–86

Improvement in speech intelligibility in noise employing an adaptive filter with normal and hearing-impaired subjects

ROBERT H. BREY, Ph.D., MARTIN S. ROBINETTE, Ph.D., DOUGLAS M. CHABRIES, Ph.D., and RICHARD W. CHRISTIANSEN, Ph.D.*

Brigham Young University, Provo, Utah, and Mayo Clinic Audiology Department, Rochester, Minnesota

Abstract—It is well known that communication in noisy environments with reverberation present is a difficult problem to solve, particularly for the hearing-impaired listener. Two-microphone noise cancelling using an LMS adaptive filter in real time was used to process speech recorded in the presence of speech-spectrum noise at six different signal-to-noise ratios, -8, -4, 0, 4, 8, and 12 dB. Twelve normal-hearing and 11 sensorineural hearing-impaired subjects were tested. Results indicated mean improvement of 37.34 percent for the 12 normal-hearing subjects and 38.3 percent for 5 of the sensorineural hearing-impaired subjects. Individual data for the 5 remaining hearing-impaired subjects revealed severe speech intelligibility deficits when noise and reverberation contaminated the speech signal. It is proposed that rehabiltative audiological assessments include evaluation of an individual's ability to cope with reverberation and noise.

INTRODUCTION

It is a given that noise is a part of modern life and that noise affects the quality of life. Communication is impaired and sometimes impossible in noise. Stripping noise from the speech signals would greatly improve communication. This would be particularly true for the hearing-impaired listener. Attempts at removing noise from a speech signal are not new, but only recently have adaptive filters been employed for this purpose (2–8,11,13,15,16,20–24). Adaptive filters are self-adjusting based on given performance criteria which can be programmed into the device. The most common filter of this type, first published by Widrow in 1966 (23), is the Least Mean Square (LMS) adaptive filter.

Basically, the filter works by comparing the primary signal consisting of both speech and noise with another signal consisting mostly of noise. This comparison is a statistical correlation of the similarity between the two signals. Signals are correlated if they are linearly related to each other. The adaptive filter adjusts itself until it has minimized the average of the square of the difference between the desired input and the filter output. Mathematically this mean square error is at a minimum when components of the primary input that are correlated with components of the reference input have been removed from the output. For a more complete explanation of the algorithm, see Christiansen (8).

Lim, in the 1982 Vanderbilt Hearing Aid Report (14) summarized a number of approaches that have been used to enhance speech in noise. Included was Widrow's adaptive noise canceller technique (23), used in this study. In summary, Lim stated "Various speech enhancement systems discussed in the previous sections appear to improve speech quality, but not speech intelligibility." In 1982 Chabries et al. (7) presented data showing that intelligibility was

* Robert H. Brey is with the Department of Education Psychology, Brigham Young University, Provo, Utah 84602. Martin S. Robinette is with the Mayo Clinic Audiology Department, Rochester, Minnesota 55905. Douglas M. Chabries and Richard W. Christiansen are with the Department of Electrical Engineering, Brigham Young University, Provo, Utah 84602.

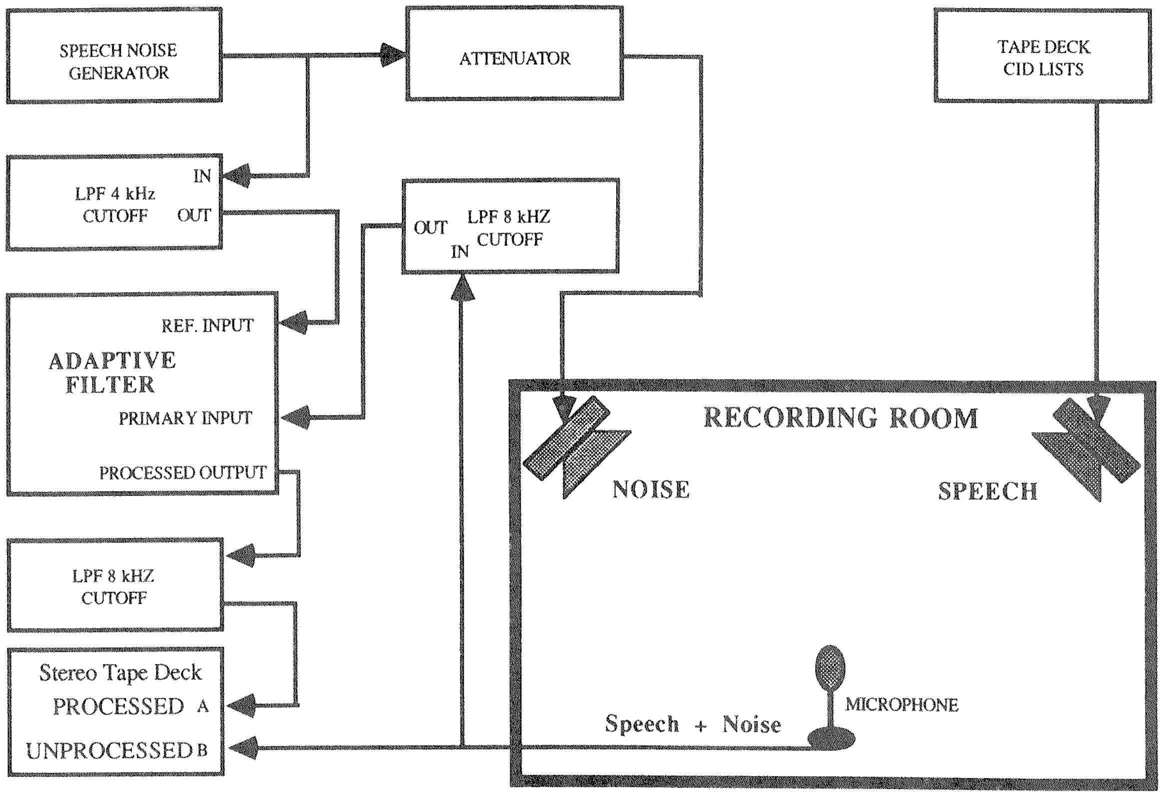

Figure 2.
Equipment diagram for producing the processed and unprocessed tapes in quiet and in the 6 S/N conditions.

METHOD

Production of Test Materials

The equipment employed to produce the tape recordings used in the study is shown in **Figure 2.** The room where the recordings were made measured approximately 3.5 × 4.6 × 3.5 meters. Speech-spectrum noise generated by a Grason Stadler noise generator Model E-5539A was played through one loudspeaker and CID W-22 phonetically balanced word lists were played through the other. The speech and noise were picked up by the "primary input" microphone placed 150 cm equidistant from both loudspeakers, and the loudspeakers were 193 cm apart, thus forming a triangle. The "reference" input to the adaptive filter was routed from the noise generator through a lowpass filter with a 4 kHz cutoff. The purpose of this lowpass filter was to shape the speech-spectrum noise so that it looked more like the acoustical noise being emitted by the loudspeaker used in this study. This configuration provided a realistic environment for noise cancellation because the interference was composed of multiple reflections off the walls and equipment in the room.

Adaptive filter processing was performed in "real-time" using a Modular Adaptive Signal Processor (MASP) with a 12 bit A/D and D/A. The length of the filter was set at 688 taps with a 20 kHz sample rate. The large number of taps were required to compensate for the impulse response of the room. Reverberation measurements made in 1/3 octaves in the test room ranged from .317 to .404 seconds for 250 Hz through 4,000 Hz. A small time delay (48 samples = 2.4 ms) was inserted into the reference channel of the MASP to account for some of the propagation delay. As shown in **Figure 2**, pre and post lowpass aliasing filters with a cutoff of 8 kHz were used for the MASP.

Journal of Rehabilitation Research and Development Vol. 24 No. 4 Fall 1987

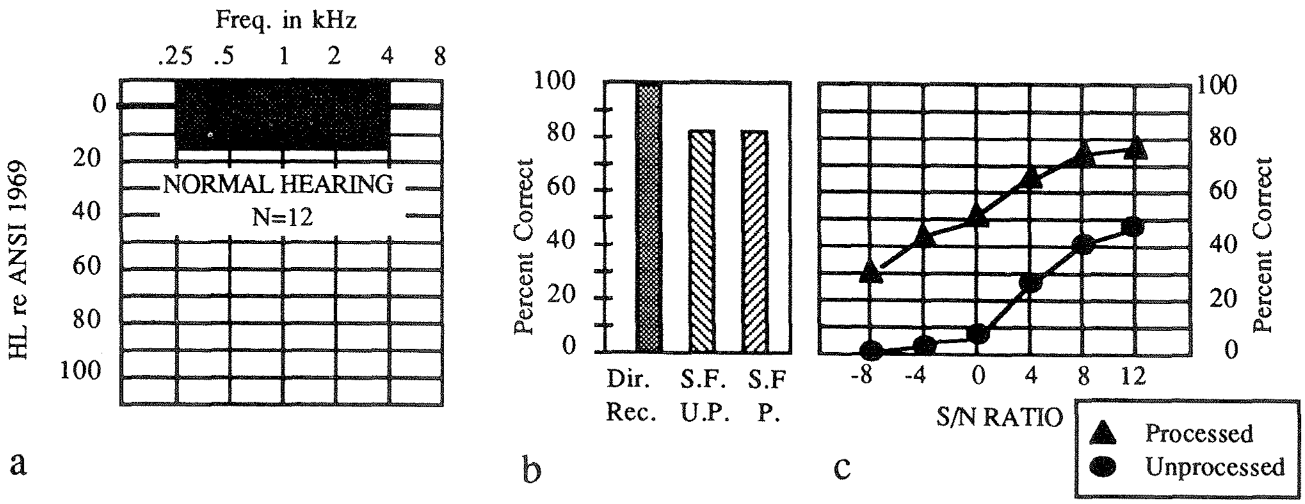

Figure 3.
Summary of data for 12 normal-hearing subjects. (a) Audiogram. (b) Histogram shows mean percent correct in quiet for direct recording, sound-field unprocessed, and sound-field processed. (c) Mean percent correct for 6 S/N ratios for processed and unprocessed tapes.

Thus, the signal recorded on channel A of the stereo tape deck was processed (using the two-microphone adaptive canceller algorithm) and the signal recorded on channel B was unprocessed.

Subjects and Procedures

Twelve subjects, all young adults with a mean age of 23 years, were classified as having normal hearing by meeting the following criteria: a) pure tone threshold equal to or below 15 dB HL in the right ear from 250–4,000 Hz re. ANSI - 1969 Standards (23), and b) obtaining a score of 96 to 100 percent on word list 1A (of CID W-22 word lists) from a direct recording under earphones when presented at 60 dB HL. This test tape, used for screening, was a direct dub from the record with no contamination by room acoustics in the recording. Thus, it was free of reverberation.

The 12 normal-hearing subjects were presented with the test tapes at 60 dB HL to the right ear. Speech intelligibility was measured in quiet and under different S/N conditions varying from -8 to +12 dB, in increments of 4 dB, both with and without adaptive filtering. The order of presentation of the processed and unprocessed sound-field tapes was counterbalanced. Word lists were always presented from the poorest S/N condition (-8 dB) in increasing 4 dB increments to the highest S/N condition (+12 dB) to reduce learning effects from listening to different randomizations of CID lists 3 and 4. The more positive S/N ratios were, of course, easier to understand. Data collection was carried out in two test sessions requiring about 1 hour each, with brief rest periods interspersed between presentations. Responses of subjects to the 50 word lists were written.

Data obtained on 11 hearing-impaired subjects were collected in the same manner as for normal-hearing subjects, with the exception that the test tapes were played at a most comfortable listening level (MCL) instead of at a fixed dB HL level. Also, the right ear was tested only if the hearing was symmetrical. If the hearing was asymmetrical, the poorer ear was tested. All hearing-impaired subjects were adults ranging in age from 19 to 84 who had sensorineural hearing losses.

RESULTS

Normal Hearing Subjects

The Audiogram in **Figure 3a** shows the shaded area used as the criterion for the 12 normal-hearing subjects. The histogram in **Figure 3b** shows that the 12 normal-hearing subjects obtained a mean score of 99.5 percent on the initial screening list (the recording uncontaminated by room acoustics, which will be referred to as the direct recording). Their monaural speech intelligibility scores using the tapes recorded in the sound-field without noise present were 82.5 percent for the unprocessed and 82 percent for the processed recordings. Standard deviations were 3.4 and 5.7 respectively. Therefore, monaural speech intelligibility dropped approximately 17 percent when the tapes were used that had been recorded in a sound-field. The reason for this large difference in intelligibility for the direct vs. sound-field conditions will be discussed later.

It is important to note that in **Figure 3b** the scores were comparable, in the sound-field condition without noise, whether the tapes used had been recorded in the unprocessed condition or had been routed through the adaptive filter in the processed mode. This would indicate that the adaptive filter did not degrade the signal and that the maximum possible score the subjects could obtain in noise, using the processed or unprocessed tapes each contaminated by room acoustics, should be approximately 82 percent.

Of primary importance, however, are the other data, shown in **Figure 3c**. The lower curve shows the mean data for the various S/N ratios in the unprocessed condition. Note that the mean intelligibility scores at -8, -4, and 0 dB S/N conditions were all less than 5 percent. The function then begins to rise to 25, 42, and 46 percent intelligibility for the three positive S/N conditions.

The upper curve represents the same set of S/N conditions after the adaptive filtering process. The mean improvement in speech intelligibility was 37.5 percent across all 6 S/N conditions represented by separate mean improvements of 31, 45, 45, 43, 31, and 30 percent for the S/N conditions of -8, -4, 0 +4, +8, and +12 dB, respectively. Note that an accurate estimate of improved intelligibility could not be obtained at the -8 and -4 S/N ratios since the unprocessed scores were zero percent. The mean data and standard deviations for the normal hearing subjects are shown in **Table 1**.

The maximum mean score for the processed condition of 75.3 percent at a +12 S/N ratio appears somewhat low. However, it should be remembered that the maximum mean score in the sound-field condition without noise present was 82 percent. Close inspection of the standard deviations shows that the maximum score obtained with the noise-cancelled tapes was approaching the range of the scores obtained in quiet.

Direct Recording vs. Sound-field Recording Discrepancy

As previously discussed in **Figure 3b**, the decrease in mean speech intelligibility by normal hearing subjects of 17 percent (99.5 percent to 82.5 percent) between the initial screening (direct recording) and the sound-field recording without background noise, was more than expected. Degradations of speech intelligibility from 6 to 13 percent have been reported for normal hearing subjects listening to monaural recordings constructed in reverberation conditions

Table 1
Mean processed and unprocessed speech intelligibility scores across 6 S/N ratios for normal-hearing subjects.

		SIGNAL-TO-NOISE RATIO						
		−8	−4	0	+4	+8	+12	Combined Mean
Normal-hearing subjects speech intelligibility in percent. N = 12	Unprocessed Mean	0%	1%	5%	25%	42%	46%	19.83%
	SD	0	1.59	4.22	10.21	6.09	6.02	
	Processed Mean	31%	46%	50%	68%	73%	75%	57.17%
	SD	5.75	4.75	6.55	4.52	6.18	6.95	
	Processed Improvement	31%	45%	45%	43%	31%	30%	37.34%

similar to the present study of .3 to .4 seconds (9,10,12,17,18).

In order to determine if the 17 percent intelligibility degradation was due to the acoustic characteristics of the sound-field in which the test tape was recorded or to equipment variables, the following auxiliary study was performed.

Three tape recordings were used. Tape #1 was the same original tape that had resulted in the 17 percent degradation (sound-field in quiet unprocessed). Tape #2 was a new recording replicating precisely the environment, equipment, and procedure that had been utilized in the recording of Test Tape #1. The recording of Test Tape #3 utilized the same equipment and procedure as used for the recording of Tapes #1 and #2, with the exception that an anechoic chamber served as the recording environment.

Thirty normal-hearing subjects, who showed a mean score of 99.3 percent (SD = 1.17) intelligibility from a direct recording of CID W-22 word list IA, under earphones, were randomly separated into 3 groups of 10 each and administered one of the 3 test tapes in the same manner as the 12 subjects in the primary study.

Table 2 shows the results of the auxiliary study. Note that subjects receiving Tape #3, recorded in the anechoic chamber, showed excellent discrimination (97.8 percent) suggesting that technically the recordings were produced correctly.

Test Tape #2 showed a decrease in intelligibility performance relative to Tape #3 of 5.8 percent. This drop is consistent with reported degradation due to sound-field reverberation of .3 to .4 seconds (9,17,18).

Table 2
Auxiliary study—Comparison of the three test tapes for mean percent speech intelligibility performance.

Sound-field tapes	Number of subjects	Mean percent correct	SD
Test tape 1 (Original)	10	82.8%	3.68
Test tape 2 Duplicate protocol of original)	10	92.0%	3.40
Test tape 3 Anechoic chamber)	10	97.8%	2.57

The difference between Tape #2 (92.0 percent) and Tape #1 (82.8 percent) is more perplexing. The score on Tape #1 is similar to scores that had been obtained by the 12 normal-hearing subjects in the primary study (82.5 percent), suggesting that subject variability did not account for the seemingly high sound-field degradation for respondents to Tape #1.

Since Tapes #1 and #2 were recorded with the same equipment, environment, and protocol, one explanation relates to a possible difference in background noise present in the test room on the separate days the tapes were recorded. Two of the walls of the test room are adjacent to a central computer room. Consequently, background noise impinging on the sound-field recording room could vary, depending on the computer activities that took place on the days the tapes were recorded.

During the recording of test tapes the speech materials were held constant at an average of 80 dB SPL at the recording microphone. The overall noise floor was checked to insure that it was at least 10 dB below the speech material. However, the spectral characteristics of the noise floor are unknown. Conceivably, with the recording of Test Tape #1, the background noise may have contained more high frequency components than Tape #2 which resulted in reducing intelligibility of taped consonant sounds. A more complete discussion of this auxiliary study on test room acoustics may be found in the report by Pappas (19).

Hearing Impaired Subjects

Results for the 11 hearing impaired subjects will be divided into two groups. All hearing impaired subjects used in the study have test results consistent with sensorineural hearing losses. Only their pure tone air conduction audiograms will be shown for one ear, although bone conduction and immittance testing were carried out on all subjects. Group I consisted of 6 subjects whose intelligibility scores were 80 percent or above on the initial screening list and greater than 55 percent on the unprocessed sound-field tape without noise present. Group II hearing impaired consisted of 5 subjects whose speech intelligibility was less than 80 percent on the initial screening or less than 55 percent on the unprocessed sound-field tape without noise present.

Group I Hearing Impaired Subjects. The composite audiogram for subjects 1-5 in Group I is shown

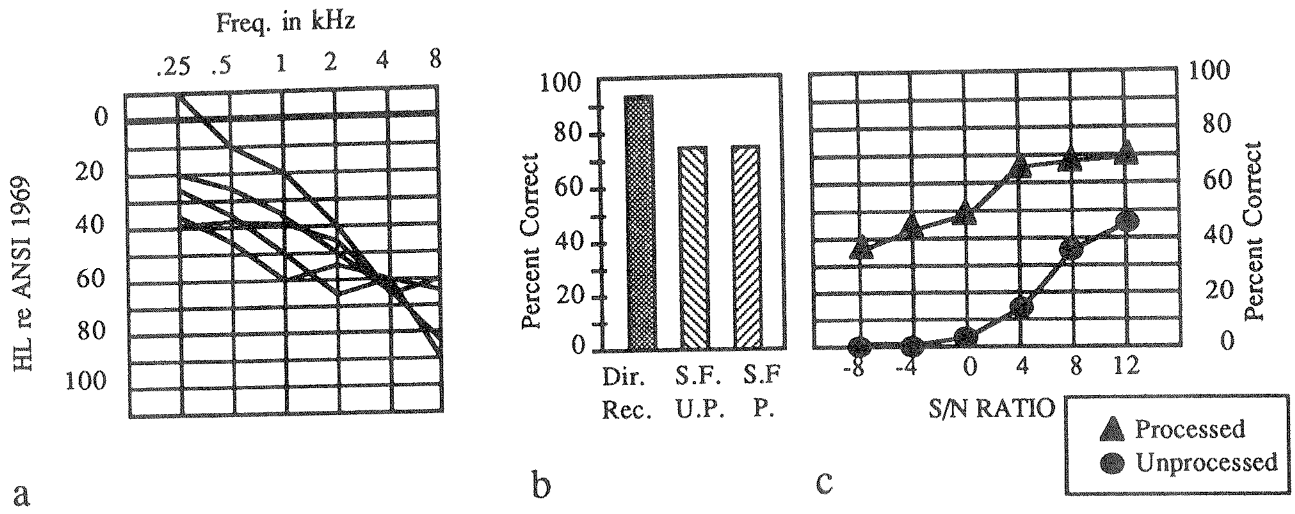

Figure 4.
Summary of data for subjects 1-5 in Group I of the hearing-impaired. (a) Composite Audiogram. (b) Histogram shows mean percent correct in quiet for direct recording, sound-field unprocessed, and sound-field processed. (c) Mean percent correct for 6 S/N ratios for processed and unprocessed tapes.

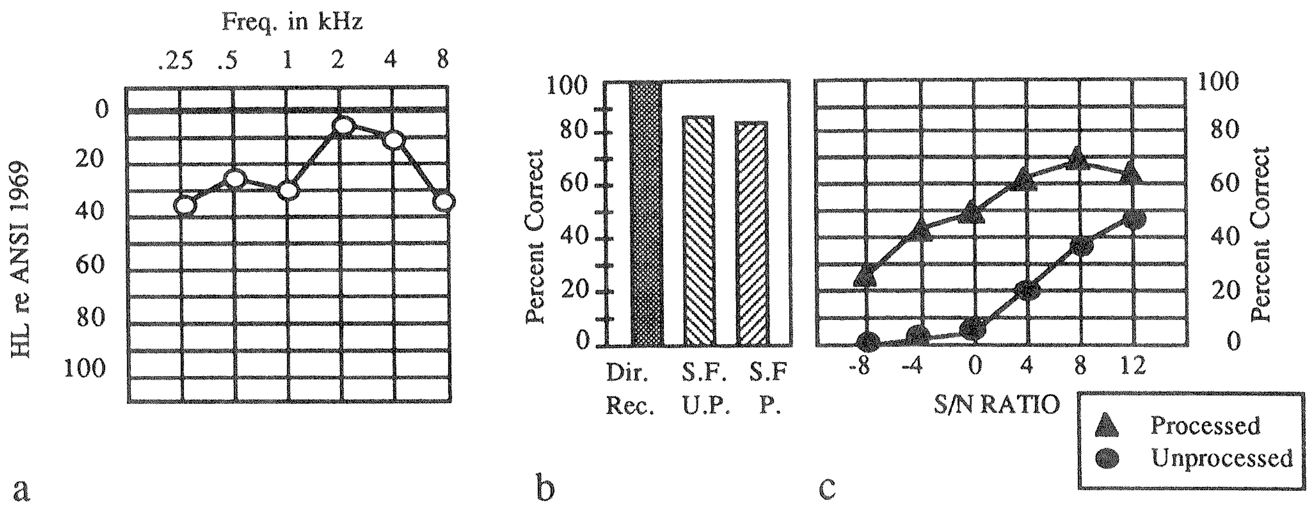

Figure 5.
Summary of data for subject 6 in Group I of the hearing-impaired. (a) Audiogram. (b) Histogram shows mean percent correct in quiet for direct recording, sound-field unprocessed, and sound-field processed. (c) Mean percent correct for 6 S/N ratios for processed and unprocessed tapes.

in **Figure 4a**. The mean slope of the hearing losses from 125 to 8 kHz was 10 dB/octave (SD=4.1 dB). These subjects were grouped together because of their common high frequency sensorineural type hearing losses. The audiogram for subject 6 in Group I is shown in **Figure 5a**. Although he fit the criteria for Group I in terms of speech intelligibility, his sensorineural hearing loss was more low-frequency in nature and thus his data were handled separately.

The histogram in **Figure 4b** for Group I shows that the mean intelligibility score for the direct recording in quiet was 92.8 percent. The mean scores for the sound-field recordings in quiet for both the processed and unprocessed recordings were 76.8 percent. **Figure 4c** shows the mean scores for the processed (upper curve) and unprocessed (lower curve) conditions across the 6 S/N ratios. Note the similarity of these data compared to the normal-

Journal of Rehabilitation Research and Development Vol. 24 No. 4 Fall 1987

hearing data presented in **Figure 3c**. **Table 3** presents the means and standard deviations in percent correct for subjects 1-5 in Group I.

The mean improvement in speech intelligibility seen in **Table 3** across all S/N conditions was 38.3 percent for the 5 hearing impaired subjects which is similar to the 37.5 percent improvement previously reported for the 12 normal-hearing subjects in **Table 1**.

Figure 5a shows the audiogram for subject number 6 of the hearing-impaired Group I. This subject has a low-frequency sensorineural type hearing loss rather than the high-frequency sloping loss shown by the other five subjects in the group. **Figure 5c** shows that the processed versus unprocessed speech intelligibility scores are similar to those of the normal-hearing subjects, as well as the other 5 hearing-impaired subjects of Group I.

Group II Hearing Impaired Subjects. Figures 6 through **10** show the individual data for the 5 hearing-impaired subjects in Group II whose sound-field monaural speech intelligibility was below 80 percent on the direct recording in quiet or below 55 percent (mean 37 percent SD = 13.3) listening to the sound-field recordings without noise present.

Subject JB was a 59-year-old male whose data are displayed in **Figure 6**. His audiogram in **Figure 6a** exhibited a high-frequency sensorineural hearing loss. His score of 44 percent correct listening to the direct recording shown in **Figure 6b** was the lowest score for all our subjects. This would indicate that, even without reverberation and in quiet, he would do poorly. With reverberation added he dropped another 20 percent. His scores listening to the 6 S/N conditions processed and unprocessed never exceeded 20 percent although he did slightly better with the processed materials. Certainly this is an individual with severe communication problems.

Subject RB, a 51-year-old male shown in **Figure 7a**, had a rather flat, moderate, sensorineural hearing loss. **Figure 7b** showed that he did very well listening to the direct recording, i.e., 84 percent, but his listening ability fell to below 30 percent when a moderate amount of reverberation was added. **Figure 7c** shows that he never reached 20 percent correct listening to any of the S/N conditions in either the processed or unprocessed tapes, although he did slightly better at the positive S/N ratios. This subject would also be expected to have extreme communication problems in a real world environment with noise, reverberation, or both present.

Subject EC, a 55-year-old female whose data are shown in **Figure 8a**, had a low-frequency sensorineural hearing loss. She did extremely well listening to the direct recording, as shown by her score of 96 percent in **Figure 8b**. However, she showed a dramatic drop in intelligibility when reverberation was added. **Figure 8c** indicates that she was able to benefit from the signal-processing, as her score rose by 36 percent from the unprocessed to processed condition at a +12 S/N condition.

Subject JG, a 56-year-old female, had a moderately-severe flat sensorineural hearing loss as seen in **Figure 9a**. She did fairly well listening to the

Table 3
Mean processed and unprocessed speech intelligibility scores across 6 S/N ratios for hearing-impaired subjects.

		SIGNAL-TO-NOISE RATIO						
		−8	−4	0	+4	+8	+12	Combined Mean
Hearing-impaired subjects speech intelligibility in percent. N = 5	Unprocessed Mean	0%	0%	3%	14%	37%	47%	16.83%
	SD	0	0	3.7	8.7	8.6	7.1	
	Processed Mean	36%	44%	49%	65%	68%	69%	57.17%
	SD	6.6	4.8	7.7	7.6	10.3	8.4	
	Processed Improvement	36%	44%	46%	51%	31%	22%	38.30%

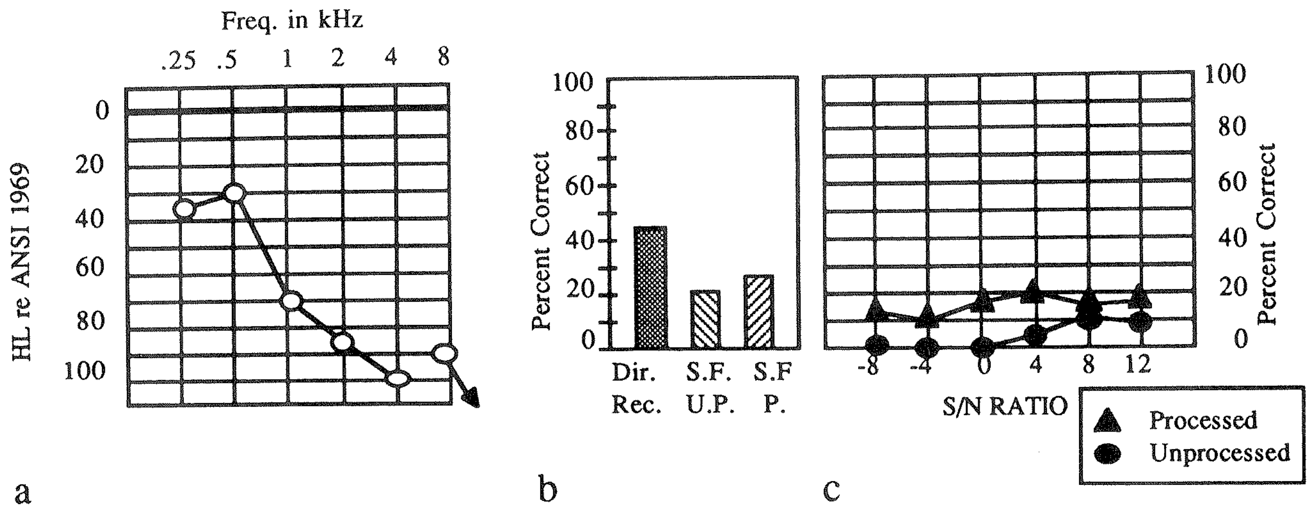

Figure 6.
Summary of data for subject JB in Group II of the hearing-impaired. (a) Composite Audiogram. (b) Histogram shows mean percent correct in quiet for direct recording, sound-field unprocessed, and sound-field processed. (c) Mean percent correct for 6 S/N ratios for processed and unprocessed tapes.

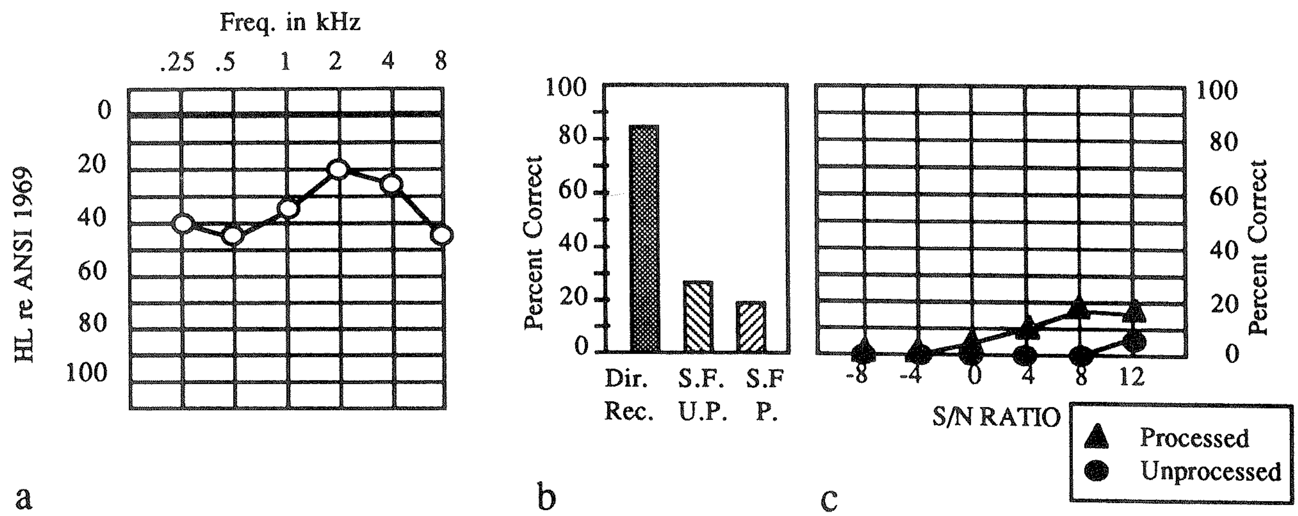

Figure 7.
Summary of data for subject RB in Group II of the hearing-impaired. (a) Composite Audiogram. (b) Histogram shows mean percent correct in quiet for direct recording, sound-field unprocessed, and sound-field processed. (c) Mean percent correct for 6 S/N ratios for processed and unprocessed tapes.

direct recording, scoring 82 percent as seen in **Figure 9b**. She too showed a dramatic drop in intelligibility when reverberation was added, dropping to below 50 percent. **Figure 9c** shows that the noise cancelling improved her scores by 30 to 40 percent for the poor S/N conditions.

Subject LS, an 84-year-old male shown in **Figure 10a**, exhibited a high frequency sensorineural hearing loss. **Figure 10b** shows that he did quite well listening to the direct recording, scoring 86 percent. However, when reverberation was added he dropped to 50 percent. The data in **Figure 10c** indicate that the noise cancelling improved his scores only at the 0 and +4 S/N conditions.

In summary, all subjects in the normal-hearing group and in Group I hearing-impaired subjects showed improved scores for processed tapes over the unprocessed tapes across the 6 S/N conditions tested. Group II's hearing-impaired subjects data cannot be directly compared with the other subjects

Journal of Rehabilitation Research and Development Vol. 24 No. 4 Fall 1987

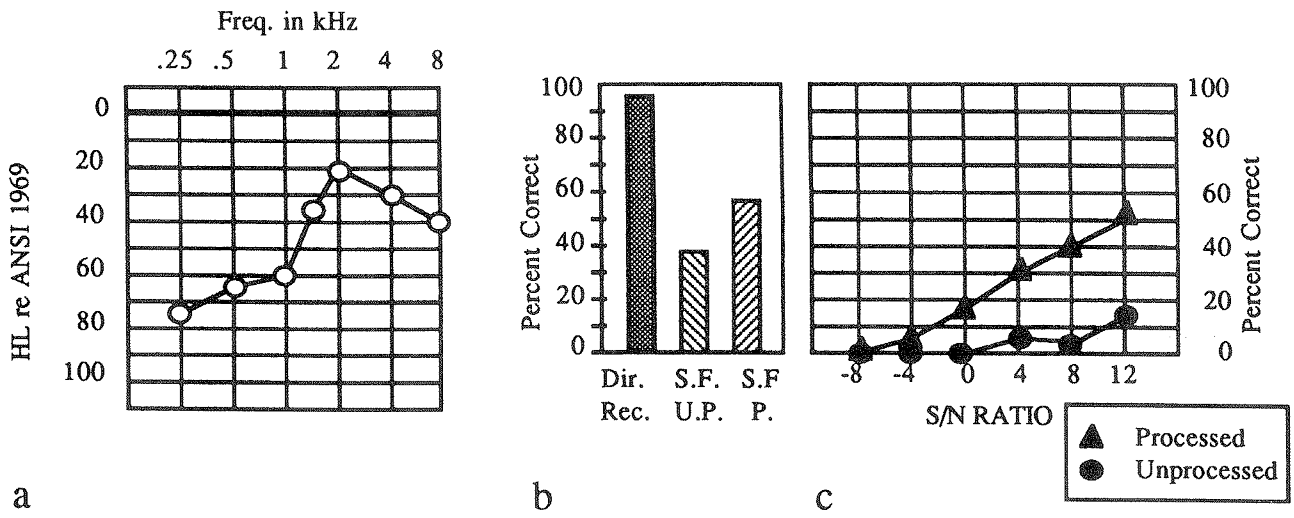

Figure 8.
Summary of data for subject EC in Group II of the hearing-impaired. (a) Composite Audiogram. (b) Histogram shows mean percent correct in quiet for direct recording, sound-field unprocessed, and sound-field processed. (c) Mean percent correct for 6 S/N ratios for processed and unprocessed tapes.

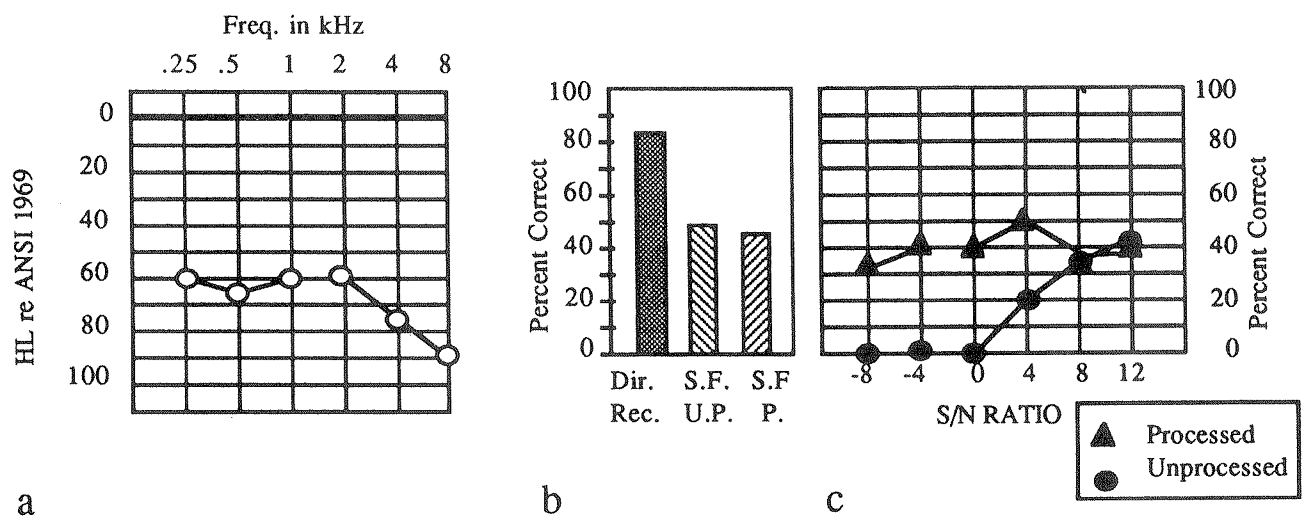

Figure 9.
Summary of data for subject JG in Group II of the hearing-impaired. (a) Composite Audiogram. (b) Histogram shows mean percent correct in quiet for direct recording, sound-field unprocessed, and sound-field processed. (c) Mean percent correct for 6 S/N ratios for processed and unprocessed tapes.

presented because of their poor sound-field speech intelligibility prior to receiving the processed and unprocessed materials. However, each of the 5 hearing-impaired subjects in Group II exhibited data that should be handled on an individual basis.

It is interesting to note that, for each subject, the best processed intelligibility score was: 1) an improvement over the unprocessed score, and 2) similar to the maximum sound-field intelligibility results prior to the inclusion of the six S/N conditions.

CONCLUSIONS

The results of this study indicate the advantages of a noise cancellation system utilizing a reference sample of the noise that is independent from the primary speech signal mixed with the noise. Three findings of this investigation are of particular interest. First, normal-hearing individuals can certainly benefit from noise cancelling using a two-microphone algorithm. Second, the data for most hearing-

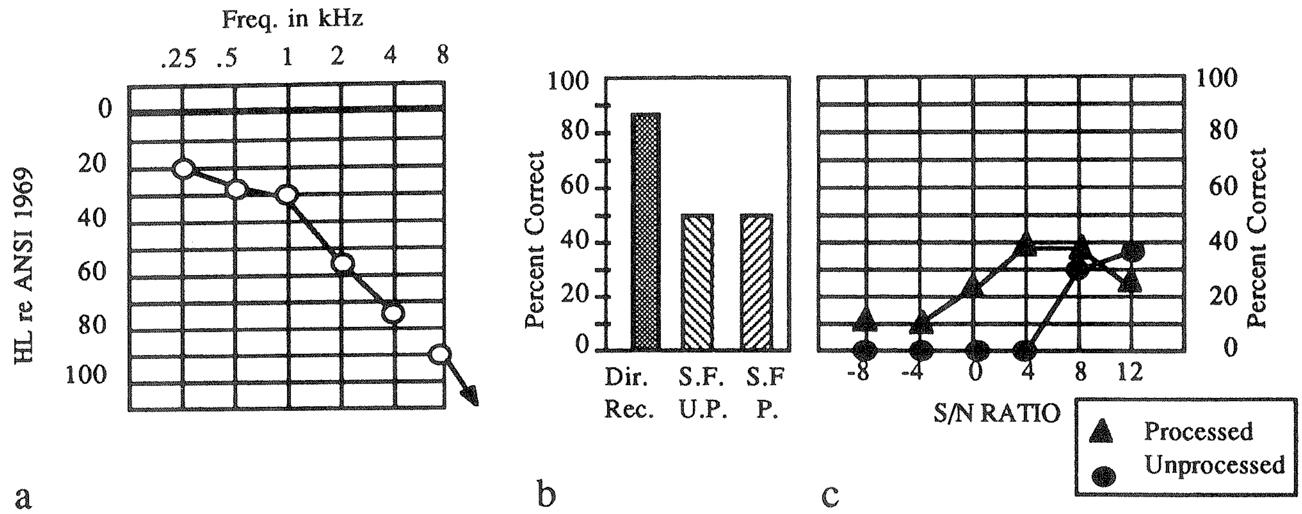

Figure 10.
Summary of data for subject LS in Group II of the hearing-impaired. (a) Composite Audiogram. (b) Histogram shows mean percent correct in quiet for direct recording, sound-field unprocessed, and sound-field processed. (c) Mean percent correct for 6 S/N ratios for processed and unprocessed tapes.

impaired subjects (Group I) showed speech intelligibility improvement similar in pattern and magnitude to that of the normal-hearing subjects. Generally, subjects with moderate hearing impairment report severe difficulty in understanding speech in poor S/N ratios. With this noise cancellation procedure, one might expect the background noise reduction to enhance speech intelligibility for the hearing-impaired subjects to an extent that would approach the maximum intelligibility they would experience for an optimal speech signal in a quiet background. Third, the individuals in Group II exhibited severe communication problems even without the presence of noise. In some cases, however, the noise cancelling did at least help them to do as well with the noise as they did in quiet. As a side benefit to this study, the authors feel it is important to point out some noteworthy data shown by the 5 subjects in Group II. Although the mean score for the subjects when listening to the direct recording in quiet was 87 percent, their mean score dropped to 46 percent when a moderate amount of reverberation was added. Most extreme is subject RB in **Figure 7** who dropped from 82 to 28 percent, a drop of 54 percent. Subject JB in **Figure 6** was even worse off as she scored only 46 percent listening to the direct recording and 24 percent with reverberation. These individuals exhibit the difficult types of problems with which the hearing health team must begin to deal.

It would certainly make sense to include reverberation and different noise measurements as routine clinical work, when trying to assess an individual's ability to communicate in the real world and whether or not he/she can benefit from amplification. All who work closely with hearing-impaired persons have often heard the comment that understanding the spoken word is difficult. These data certainly support that notion and further crystallize the fact that reverberation and noise can be devastating to those with hearing impairment.

A drawback to current communication systems designed to aid hearing-impaired individuals in classrooms or industrial and military environments is the need for a positive S/N ratio at the speakers microphone. The results of this study showed maximum or near maximum speech intelligibility improvement when the S/N ratio at the primary microphone was zero or negative.

Future work is under way to apply the two-microphone adaptive canceller system in different types of noise environments, including speech babble and real world industrial settings.

REFERENCES

1. **American National Standards Institute.** *American National Standards Specifications for Audiometers S3.* 6-1969. New York: American National Standards Institute Inc., 1970.
2. **Berouti M, Schwartz J, and Makhou J:** Speech corrupted by acoustic noise; enhancement method. *International Conference of Acoustics, Speech and Signal Processing.* pp 208-211, 1979.
3. **Boll SF:** Adaptive noise cancelling in speech using the short time transform. *Proceedings of ICASSP* (3):692-695, 1980.
4. **Boll SF:** Suppression of acoustic noise in speech using spectral subtraction. Institute of Electrical and Electronic Engineers, *Transaction on Acoustics, Speech and Signal Processing.* ASSP 27:113-120, 1979.
5. **Boll SF and Pulsipher DC:** Suppression of acoustic noise in speech using two microphone adaptive noise cancellation. *IEEE Transactions on Acoustics, Speech, and Signal Processing* 28(6), 1980.
6. **Brey RH and Robinette MS:** Noise reduction in speech using adaptive filtering: II. Speech intelligibility improvement with normal and hearing subjects. Paper presented at the *Acoustical Society of America Convention* 71(1):S8, 1982.
7. **Chabries DM, Christiansen RW, Brey RH, and Robinette MS:** Application of the LMS adaptive filter to improve speech communication in the presence of noise. *IEEE International Conference on Acoustics, Speech and Signal Processing* 1:148-151, 1982.
8. **Christiansen RW, Chabries DM, and Lynn D:** Noise reduction in speech using adaptive filtering: I. Signal processing algorithms. *J Acoust Soc Amer* 71(1):S7, 1982.
9. **Finitzo-Hieber T and Tillman T:** Room acoustics effects on mono-syllabic word discrimination ability for normal and hearing-impaired children. *J Speech Hear Res* 21:440-448, 1978.
10. **Gelfand SA and Hochberg I:** Binaural and monaural speech discrimination under reverberation. *Audiology* 15:72-84, 1976.
11. **Griffiths L:** An adaptive lattice structure used for noise-cancelling applications. *Proceedings of the IEEE Conference on Acoustics, Speech and Signal Processing.* Tulsa, Oklahoma, pp 87–90, 1978.
12. **Harris RW and Reitz ML:** Effects of room reverberation and noise on speech discrimination by the elderly. *Audiology* 24:319-324, 1985.
13. **Lim JS:** Signal processing for speech enhancement. Prentice-Hall Inc., 1983.
14. **Lim JS:** Signal Processing for Speech Enhancement. *The Vanderbilt Hearing Aid Report* (GA Studebaker, FH Bess, Editors), pp 124–129, 1982.
15. **Lim JS, Oppenheim AV, and Braida LD:** Evaluation of an adaptive comb filtering method for enhancing speech degraded by white noise addition. *IEEE Transactions on Acoustics, Speech and Signal Processing.* ASSP 26:354-358, 1978.
16. **McAulay RJ and Malpass ML:** Speech enhancement using a soft-decision noise supression filter. *IEEE Transactions on Acoustics, Speech, and Signal Processing.* ASSP 28:137-139, 1980.
17. **Moncur JP and Dirks D:** Binaural and monaural speech intelligibility in reverberation. *J Speech Hear Res* 10:186-195, 1967.
18. **Nabalek AK and Robinson PK:** Monaural binaural speech perception in reverberation for listeners of various ages. *J Acoust Soc Am*, 71:1242-1248, 1982.
19. **Pappas KAK:** Comparison of speech discrimination performance under three sound-field conditions. Unpublished thesis, University of Utah, 1982.
20. **Pulsipher D, Boll SF, Rushforth CK, and Timothy LK:** Reduction of nonstationary acoustic noise in speech using LMS adaptive noise cancelling. *Proceedings of the IEEE Conference on Acoustics, Speech, and Signal Processing.* Washington, D.C., pp 204–207, 1979.
21. **Sambur M:** Speech processing adaptive noise cancelling using LMS approach. *IEEE Transactions on Acoustics, Speech and Signal Processing* 26:354-358, 1978.
22. **Tremain TE, Fussell JW, Dean RA, Abzug BM, Cowing MD, and Boudra PW:** Implementation of two real time narrow band speech algorithms. *EASCON 78 Record.* pp 698–708, 1978.
23. **Widrow B:** Adaptive filters: 1. Fundamentals. Stanford Electronics Lab, Stanford University, Rep. *SU-SEL*-66-126, 1966.
24. **Widrow B, Glover JR, McCool JM, Kaunitz J, Williams CS, Hearn RH, Zeidler JR, Dong E, and Goodlin RT:** Adaptive noise cancelling: Principles and application. *Proceedings of the IEEE* 63(12):1692-1715, 1975.