Veterans Administration

Journal of Rehabilitation Research and Development Vol. 24 No. 4
Pages 111–118

Evaluation of adaptive multimicrophone algorithms for hearing aids

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Abstract—It appears that one of the most important factors limiting the usefulness of hearing aids is the high sensitivity of hearing aid users to interfering sounds. In this work, the possibility of utilizing computing power that could be packaged into a unit the size of a hearing aid in order to cancel out noises before they reach the ear was investigated. Algorithms for combining the outputs of a number of microphones so as to achieve a considerable noise reduction are proposed and evaluated, and some questions related to implementation are studied. Some experimental results based on numerical and listening tests are presented. The computation is started by indicating to the system a silent period of the main speaker. In order to minimize distortion for the main speaker, the algorithms try to produce an output that matches the first few autocorrelations of the main speaker. It is expected that the system should be able to track sufficiently slow variations in the positions of the noise sources and the main speaker.

INTRODUCTION

It seems to be a commonly accepted fact that the presence of noise places a very severe limitation on the performance of hearing aids. A very useful approach is to eliminate the interference before it enters the ear. A great many studies have been carried out on this general problem. Voiced speech may be distinguished from other sounds by the presence of a particular periodicity that changes slowly. This causes the long term spectrum to be concentrated at discrete frequencies. In addition, speech has a short time spectrum that modulates the intensity at each of these frequencies. The work of Sambur (7) uses the periodic structure of the speech signal in an interesting way, by filtering the signal through a pitch dependent adaptive filter. Another approach in the same class is the work of Parsons (6). The spectral peaks of the signal are separated into two groups corresponding to the main and undesired signals, and the signal is reconstructed from the spectral peaks of the desired speaker alone.

The method of spectral subtraction, as studied for example by Boll (1), is based on the fact that the spectrum of the sum of the noise and the interference is the sum of the spectra of the individual signals, if the signals are independent. Therefore, if one possesses a good estimate of the spectrum of the noise, it is possible to subtract it from the spectrum of the signal plus noise to obtain an estimate of the spectrum of the signal. It is then possible to use some spectral reconstruction technique to obtain a signal with the estimated spectrum. This is basically an estimation problem and was treated as such by Ephraim and Malach (2).

All of these techniques attempt to estimate signals in noise, and are subject to estimation errors which may be shown to be bounded away from zero, with the inherent consequent degradation of the original signal. Multimicrophone techniques offer, at least in theory, the prospect of precise reconstruction of the desired signal. Consider, for example, the case where two microphones are placed in the room, where two sine waves of identical frequencies are

present in different locations in the same room. Typically, each of the sine sources will reach each of the microphones with a different phase and magnitude. If one knows the transfer function of the room for this frequency for both locations, one may solve the set of two equations in two unknowns to obtain the amplitudes of both sine waves precisely from the amplitudes and phases of the signals in both microphones. The problem that still remains is the estimation of the transfer function of the room. But in this case, the quantity to be estimated is much less subject to statistical variation then, for example, the particular realization of the noise spectrum. A number of workers have attempted to address the problem of utilizing multimicrophone data to cancel out noises. Mitchell et al. (5) proposed a nonlinear and four-microphone spatial filtering approach. The approach is justified only in free space for delta function inputs.

An adaptive frequency domain approach is presented by Strube (9). Room reverberations are allowed and it is assumed that it is possible to turn off the interference so that only the signal (i.e., the desired speaker) is present. During that stage, filters are found so that the desired speaker is cancelled, when these filters are applied to the microphone outputs and summed. Following the stage when the interference is switched on again the summed outputs of these filters contains only the noise, and may be used as a reference signal to cancel out the noise in the original signals. The adaption scheme is based on a frequency domain approach, using the expected value of the cyclic short time covariance matrix. Since no constraint is placed on the range of influence of the resulting filters, filters that are too long may result. Such filters may depend heavily on the cyclic structure assumed for the signal and may therefore be inconsistent with the true signal, unless the signal is periodic and the transform interval is the period.

Recently, an attempt to apply the principles of noise cancelling to the removal of noise, using microphone arrays, was carried out by Kaneda and Ohga (4). An LMS (least mean square) algorithm was applied to cancel out the noise in such a way that a reference signal consisting of random noise is minimally degraded in some sense. The random noise signal fed into each of the microphones is used to simulate the signals that would be generated at the microphones by a speaker located at the desired position, in a nonreverberant room. The work reports a definite improvement in the signal-to-noise ratio, at the cost of some distortion in the main speaker, when four microphones are used to cancel out a single additional speaker. These results were extended by Sondi and Elko (8) who modified the constraints on the distortion produced by the filters when acting on identical microphone inputs. The modified constraint attempts to fit the spectrum rather than the transfer function of the filter output to the unit spectrum. Eight microphones are used with very good results in anechoic and small rooms.

Approached formally, there are two separate problems that should be addressed in the context of multimicrophone noise cancellation. The first is whether the filters needed to carry out the cancellation exist, and what is their size. The second question relates to algorithms for identifying those filters. Some procedure must be provided for "pointing out" the location of the noises, and of the main speaker to the system, so that this identification could be carried out. Both of these question form the subject of this work. In order to point out the location of the noise sources, a silent interval of the main speaker must be identified to the system. On the basis of this information, FIR (finite-impulse-response) filters that minimize the sum of their outputs subject to a constraint on the filters that forces their sum to be the unit gain filter, are found. Following that stage, a second optimization step is carried out that attempts to fit the summed outputs of some new filters to the desired signal in some sense. To that effect, a measure of fit between the desired signal and the filter output is defined, and some possible algorithms for achieving this fit are described.

In the next section a general description of the approach will be given, and some of the limiting factors and tradeoffs described. This is followed by a mathematical statement of the problem, concluded by the experimental results and a discussion.

METHOD OF SOLUTION—MOTIVATION AND GENERAL DESCRIPTION

As mentioned in the introduction, it is assumed that there exists an interval of time, which is pointed out to the system, during which the main speaker

is silent. The natural thing to do when one is presented with just noise data is to try to find a set of filters (as many as there are microphones) so that their summed output, when they are applied respectively to each of the microphones, is as small as possible. In order to make this a well-defined problem, it is necessary to constrain the filters in such a way that their sum is the identity filter. Thus stated, the problem may be converted into the solution of a set of linear equations which may be solved in any number of ways.

In what follows, a set of filters obtained to act together on the vector of inputs will be viewed as a single (vector) filter, for convenience. Thus, the filter obtained contains the information about the noise that will be used in later periods to cancel the noise when speech is present. In order for this information to be useful, it is necessary that the filter be independent of any extraneous noise properties, such as its short-term spectrum. Otherwise, random variations in the noise spectrum will make the information irrelevant to later intervals. Therefore the interval over which the noise minimization must be carried out must be long enough to provide a spectrum with as large a support (the set of frequencies over which it is non zero) as is ever likely to occur.

Now, it is quite evident that the larger the support of the spectrum, the longer the filter needed to cancel the noise, since the filter has to fit over more and more points. It follows that not only do minimizations need to be carried out over long intervals, but also the filters must be quite long if a considerable noise reduction is to take place. The filters obtained at this stage may be applied to the incoming (noisy) signal to obtain a reduced noise signal. Since however, the filters are both long and not optimized to fit the signal, they may produce a considerable distortion of the signal, possibly making it sound more reverberant than it originally was. At this point two possible strategies will be described. Each of them has certain advantages in particular circumstances.

One possible approach is to use the (matrix) of noise autocorrelations obtained during the silent stage as the basic guide as to the direction of the noise, with the output of the filter obtained in that stage as an auxiliary or instrumental variable. The other approach does not use the autocorrelation matrix, only the instrumental variable. Both approaches depend on formulating a criterion of fit to the desired speech signal of the output of a new filter. To understand how such a criterion can be devised, it must be remembered that the first-stage filter is capable of producing a signal with a reduced noise content. Since the output of any one of the microphones is the sum of the desired signal and the noise, the scalar product of the instrumental variable with any of the microphone outputs is the sum of the products of the instrumental variable with the desired signal and with the noise. Assuming, however, that the instrumental variable is fairly noise free and that the signal is independent of the noise, the terms containing the cross product between signal and noise are small. This is also true when the scalar products of relative shifts of the two signals are considered. The criterion of fit that is used requires that the scalar product of the instrumental variable, and its shifts with the output of any new filter to be evaluated, matches its scalar product with the sum of the microphone outputs. This means that the cross correlation of the new filter output with the old filter output matches the cross correlation of the desired signal with the old filter output. The quality of the fit may be controlled by the number of cross-correlation coefficients which are used, i.e. the number of shifts of the instrumental variable whose scalar product with the signal is matched by the desired filter output. Thus, a constraint on possible choices of the desired filter has been generated. In addition to satisfying this constraint, this filter may now be required to minimize its noise output. This may be done in either of two ways. One way, which puts less weight on the instrumental variable, is for the filter to minimize the noise output that the new filter would have generated during the last silent period of the main speaker. The problem that is solved in this way has the same criterion as the first stage, but a new constraint.

Another approach is to minimize the total current output energy. This approach relies on the fact that the signal and noise are independent hence the total output energy is the sum of the filtered signal output energy and that of the filtered noise. Since the constraint does not allow the signal output to go to zero, the noise output will be minimized. The second scheme has the disadvantage that the criterion function does not distinguish between noise and signal, hence it attempts to minimize both. The signal that

is attenuated in the process is the one that has no effect on the constraint. If the instrumental variable is fairly noise free, the noise has little effect on the constraint, hence the computation attempts to minimize it. It seems intuitively clear that only by chance can the noise reduction of the calculation just proposed exceed that already present in the instrumental variable. One reason for pursuing this approach might be that the fitting may be done over shorter intervals of time. Since the spectrum of a signal over a short interval is sparser, the same degree of noise reduction may be accomplished with a shorter filter, resulting in a smaller distortion of the desired speaker.

The purpose of the discussion above was to present a collection of tools, which could be of use in removing unwanted noise from multimicrophone signals when it is not possible to turn off the noise source. The test of the theory rests with the experimental results that may be obtained for realistic conditions. Some such tests will be described in the sequel.

MATHEMATICAL STATEMENT

A shorthand notation, which will make the discussion more precise without being burdened with unnecessary detail, will now be introduced. Time signals will be denoted by capital letters. Thus the time signal $y_1(t)$ will be denoted by the letter Y_1 representing column vector of the values of y(t) arranged in ascending order. Two such columns may be concatenated to produce a two column matrix signal Y

$$Y = Y_1, Y_2 = \begin{pmatrix} y_1(1), y_2(1) \\ y_1(2), y_2(1) \\ \cdot \quad \cdot \\ \cdot \quad \cdot \\ \cdot \quad \cdot \\ y_1(T), y_2(T) \end{pmatrix} \qquad [1]$$

In this work only FIR filters will be considered. A filter is specified by its vector of coefficients and its degree of prediction. A filter is nonpredictive if its current output depends only on past values of input. In this application it was assumed that a certain delay is tolerable in producing the output, as long as it does not interfere with lipreading. Thus all filters are predictive, with the output depending on p/2 future inputs where p is the filter order. The output of filter F_1 when its input is Y_1 will be denoted as $F_1\ Y_1$. The filter output signal is padded with zeroes so that the vector of outputs will always be aligned with the vector of inputs. Since the number of points over which a filter output is defined is smaller by p-1 then the number of points in the filter input, a projection operator T will be defined. If $Z = T_{a,b}\ Y,\ 0 \leq a \leq b$, Z will be identical to Y for indices (times) in the interval a,b and Z will be zero elsewhere. The norm of a signal is the usual Euclidian norm. The norm of Y_1 will be denoted by $|Y_1|$. For vector filters the norm of the filter output is the norm of the sum of its component outputs. Finally, let the shift operator P_i be defined on any time function Y in the natural way

$$P_1\ Y = \begin{pmatrix} 0 \\ Y \end{pmatrix}$$

$$P_{-1}\ Y = \begin{pmatrix} Y(2) \\ Y(3) \\ \cdot \\ \cdot \\ Y(t) \\ 0 \end{pmatrix} \qquad [2]$$

$$P_i = P_1 P_{i-1}$$

Focusing on the case of two microphones, let the signals arising in microphone i, (i = 1,2), from the signal and noise be $s_i(t)$ and $n_i(t)$ respectively—yielding the actual output $y_i(t) = s_i(t) + n_i(t)$. The problem is to find a filter F, (F = F_1, F_2) so that

$$T_{a,b}(F_1 S_1 + F_2 S_2) = T_{a,b}(0.5(S_1 + S_2)) \quad \textit{and} \quad T_{a,b}(F_1 N_1 + F_2 N_2) = 0 \qquad [3]$$

Where a,b is the largest time interval over which all signals are defined. If a filter satisfying the equation above is found, then

$$T_{a,b}(F_1 Y_1 + F_2 Y_2) = T_{a,b}(0.5(S_1 + S_2)) \qquad [4]$$

where the noise terms cancelled. Assuming that the transfer function of the room distorts the signal and noise differently, one might expect such filters to exist. For example, in free space the difference between the times of arrival of the signals from the main speaker will typically be different from that difference for the noise source. This will serve as a basis for discrimination.

It is now possible to state the equations for the proposed algorithm. During the first stage of the

algorithm, the noise exists without the signal hence it is possible to compute the energy output of any filter acting on it. It is therefore possible to find a filter that minimizes the noise output energy. In order to keep the energy of the desired speaker at some nonzero level, the filters are constrained to add up to the unit filter. This means that if the main speaker presents an identical signal to both microphones, his speech will not be affected by the resulting filter.

$$\min\{|T_{a,b}FY| : F_1 + F_2 = \delta(t - p/2)\} \qquad [5]$$

where $\{a,b\}$ is the largest interval over which the filter output does not depend on non existing inputs. Since this is the only chance to capture the noise, the interval $\{a,b\}$ must be made long enough so that a faithful picture of the support of the spectrum of the noise is obtained. The length of the filter F is chosen so that noise is sufficiently reduced during that interval. Since this is a quadratic minimization problem, a linear set of equations must be solved to obtain F. Because the main speaker signal is not roughly equal at both microphones, the main speaker may also be attenuated by this operation, resulting in a less than optimal signal-to-noise ratio. In order to correct this, a second stage optimization may now be applied, using information on an interval $\{c,d\}$ over which the signal exists. A new filter G is sought which solves the problem

$$\min\{|T_{a,b}FY| : (T_{c,d}P_iFY)^tT_{c,d}GY = (T_{c,d}P_iFY)^tT_{c,d}0.5(Y_1 + Y_2)\ s.t.\ I_1 \le i \le I_2\} \qquad [6]$$

Note that the criterion function of this optimization is identical with that of the previous one. There are $1 + I_2 - I_1$ constraints in this problem (as opposed to p in the first stage).

In order to understand this choice of constraints, let it be noted that if the result of applying the filter to the original signal is to be equal to the original signal represented as the sum of the microphone outputs, the desired filter must satisfy the constraint:

$$T_{c,d}GY = T_{c,d}0.5(S_1 + S_2) \qquad [7]$$

If both sides of this equation are multiplied by shifts of the output of the first filter, the equation below results

$$(T_{c,d}P_iFY)^tT_{c,d}GY = (T_{c,d}P_iFY)^tT_{c,d}0.5(S_1 + S_2) \qquad [8]$$

Since the first filter output contains mainly the signal, its scalar products with N must be small because of independence of the signal and noise. Since Y = S + N, it follows that the scalar products of shifts of the first filter output with Y are good estimators of its products with S. When these estimates are placed in Equation [8] Equation [6] results. What was shown so far is that the constraints [6] are roughly satisfied for "good" filters. Filters for which [8] holds, try to match in some sense I_1, I_2 correlation coefficients of the filtered signal to those of the true one. Usually I_1 will be chosen to be equal to $-I_2$ making the first I_2 correlation coefficients of the filter output (with a signal which resembles the desired speaker) have their true value. Since it is commonly assumed that speech signal is characterized by its first 10–15 correlation coefficients (for example, as in linear predictive coding), this is a reasonable thing to do.

The value of I_2 is usually quite small, on the order of 20 to 40. Thus the problem solved in the second stage has many fewer constraints than the first stage. They are also more suitably arranged to discriminate signal from noise. The result is that besides having a better fit to the true signal, the ratio between energy originating in the signal to energy originating in the noise in the resulting signal is typically 2 to 4 times higher then in the first stage signal. Still, this processing stage relied on the assumption that the spectrum of the noise during the first, speaker-free, period is a good representative of the spectrum during the current period. For that reason, the filters that are found should be good over a wide spectral range, which is not likely to change over the whole interval. To accomplish this the filters should be sufficiently long. This may result in a reverberant quality of desired-speaker filtered signal. It is now possible, however, to formulate a slightly different optimization problem where the fitting is carried out over shorter time intervals using much shorter filters.

$$\min\{|T_{c,d}FY| : (T_{c,d}P_iFY)^tT_{c,d}GY = (T_{c,d}P_iFY)^tT_{c,d}0.5(Y_1 + Y_2)\ s.t.\ I_1 \le i \le I_2\} \qquad [9]$$

Note that Equation [9] is almost identical to [6] except that the criterion function is now defined over the present time interval. Using the independence of the signal and the noise, the criterion of [9] may be rewritten as

$$|T_{c,d}FY|^2 = |T_{c,d}FS|^2 + |T_{c,d}FY|^2 \qquad [10]$$

Despite the fact that a signal component is now present in the criterion, the constraint prevents the

signal from being cancelled by the filter, to the extent that the second stage filter output is noise free. Since the current noise output of the new filter is now being minimized, the interval $\{c,d\}$ is not constrained to be long, as it was in the previous stage. It is possible to pick relatively short time intervals for which the spectrum is much sparser, hence smaller filters are capable of producing good results. Such filters do not have a noticeable reverberant quality, and therefore they may improve intelligibility by a considerable factor.

In order to limit the computational effort required, the linear equations that result from the constrained optimization problem are solved for blocks of data, not recursively. Furthermore, fast inversion routines, which utilize the near Toepliz nature of the resulting equations, are used (3). The parameters used in the equations are correlations coefficients of the various signals that occur in the problem. These are obtained using Fast Fourier Transforms (FFTs). The complexity of the computation is therefore $O(n \log n) + O(p^2)$ where n is the block size and p is filter size.

EXPERIMENTAL AND COMPUTATIONAL RESULTS

In order to test the ideas discussed above, the computations were carried out on a main-frame computer (an IBM 4381) in nonreal time. In all tests the signal and noise were recorded separately so that separate evaluations of the filtered noise and signal output energies could be computed, thus producing a signal-to-noise ratio for the result. In addition to informal listening tests on the results, a formal listening test was carried out on a particular set of data. The following objectives were set for the testing:

1. Find the size of the interval over which a sufficiently representative spectrum for the signal and noise source may be obtained.
2. Find the size of the filters that are capable of producing a given signal-to-noise ratio.
3. Determine the tradeoff between the number of microphones and filter size.
4. Obtain estimates for the optimal distances between microphones.
5. Determine how effective the various stages of optimization are in reducing the noise.
6. Determine the extent to which degradation caused by the filtering itself affects intelligibility.

Because of computational limitations and the preliminary nature of the study, the scope of the work was limited. All the data were collected in medium size rooms, with the speaker within 1.5 m of the microphones and the disturber at around the same distance. The signal-to-noise ratio which was assumed desirable was around 6 to 10 dB. It was also assumed that, initially, the signal was roughly equal to the noise. Both speakers were male, with the same pitch range, which tended to produce a considerable overlap between their spectra.

It is expected that for large enough intervals the support of the noise spectrum will be filled out. Since, in all tests, every filter that was found was applied to the full set of data for that experiment, it was possible to determine the extent by which a filter found using one set of data was suitable for the same experimental conditions at different times. For some of the data, 1-second intervals seemed to be enough. For others, it was not possible to use filters obtained at one stage for later stages unless the time interval was at least 3 seconds. The effectiveness of the filters also seemed to depend on experimental conditions such as the size of the room. For the data used in the listening tests, filters of size 450 taps reduced the noise to a barely audible level from a signal-to-noise ratio of 1. Thus a reduction by a factor exceeding 7 was obtained. For test data in other rooms, the same size filter resulted in only a fourfold reduction. The size of the filter to be used would seem to be limited by the degradation it produces in the desired speech signal. It appears that filters beyond 450 taps produce an intolerable degradation.

The usefulness of a device based on this approach could thus be categorized in terms of the range of environments for which it would be useful. One way, found to be very effective in improving this tradeoff, is to increase the number of microphones. Thus it was found that for the examples tried, and for a given noise reduction, it was possible to reduce filter size by a factor of 3 through the use of four microphones instead of two. Since the number of equations to be solved is proportional to both the number of microphones and the filter size, and the amount of numerical work increases with the square of the number of equations (because of the special nature of the equations), the result is a net twofold

reduction in computational effort. In addition, the amount of distortion in the filtered speech signal was reduced considerably.

The distance between the microphones did not seem to have a definite effect on the quality of the filtered signals or on the needed filter size for a given signal-to-noise ratio. Evidently, when the microphones are very close, the problem becomes singular as the signals in both microphones converge to each other. As the distance becomes very large, one might expect the filters needed to convert the input from a given sound source to one of the microphones to that of the other microphone to become large. It appears that at the range of distances tried (10–40 cm), the fact that the noise signals become more distinct as the distance grows counter-balanced the increases in the complexity of the filter needed to cancel the signals.

It may be noted here that the noise reduction that may be obtained in the first stage is independent of the initial signal-to-noise ratio. For the second stage, it is of some importance for the noise to be not too large, since the constraint depends to some extent on the first-stage filtered noise being nondominant. For the third stage, the noise content of the instrumental variable is critical since it determines directly the accuracy of the result. In addition, however, if the signal is large, the biasing term in the criterion function, containing the signal, places a lower bound on the amount of noise reduction that is possible. All these considerations will therefore affect the final improvement observed with each stage. The second stage of optimization resulted quite consistently in a twofold to threefold improvement over the first stage. This improvement depends on the extent to which the signals induced by the main speaker in both microphones are close to each other. If they are identical, the first stage is optimal and one would not expect any improvement. The first stage typically resulted in a reduction by a factor of two to four. Thus a combined effect of fourfold to tenfold was obtained. For many of the cases, particularly with the use of four microphones, the disturbance becomes inaudible and only a slight distortion can be heard.

In order to estimate the effect of the distortion, a listening test was carried out on a particular set of data. Two microphones were used with an 8 kHz sampling rate. The speaker and the noise source were at about 1.5 m away from the microphones in an intermediate size room. The original signal-to-noise ratio was 0 dB and a 9 dB improvement was obtained. A very definite degradation could be heard, however, in the main speaker signal. A list of 50 high frequency words was presented to the test group. They were allowed to familiarize themselves with the written list, and then listened to the original list, followed by the noisy list (in a different order) followed by the filtered list. Scoring was based on correct recognition counts. The subjects were divided into two groups, one consisting of hearing-impaired and the other of normal-hearing persons. The test confirmed the existence of the cocktail party effect. Both the degradation due to the noise and the improvement due to filtering were greater for the hearing-impaired group. Whereas for the normal-hearing group the scores for the noiseless, noisy, and filtered signals were 96.6 percent, 53.2 percent, ands 68.7 percent, respectively, the corresponding scores for the hearing-impaired were 68.4 percent, 19.2 percent, and 45.1 percent. The use of four microphones would most probably yield improved performance gain for both groups.

Another possible way to improve the performance might be to use the output of the second stage as an instrumental variable for a third stage estimation of a shorter filter, which would be adapted separately on much smaller time intervals, and would be able to take advantage of the small support of the spectrum over such intervals. It was, however, not possible to obtain consistent results using that technique. It worked well for some sets of data but there were conditions where the noise reduction obtained was very limited, with signal-to-noise much worse than for the second stage. The technique has the further drawback that short intervals are more likely to be silent periods of the main speaker. In that case, the constraints are meaningless, and no filter can be obtained. It is, therefore, necessary to couple the approach with an algorithm that silences the noise during long silent periods of the main speaker.

To conclude, it may be noted that no evaluation of the approach can be complete without constructing a portable version of the device and evaluating it for actual field condition. It may also be noted here, however, that with present signal processors the computational requirements are on the far limit of existing technology. It may therefore be some time before practical arrays of processors capable of tackling this task become available.

REFERENCES

1. **Boll SF:** Suppression of acoustic noise in speech using spectra subtraction. *IEEE Trans* ASSP-27:113–120, 1979.
2. **Ephraim Y and Malach D:** Adaptive speech signal dereverberation. *11th Convention of Electrical and Electronics Engineers in Israel, Tel-Aviv,* Oct. 1979.
3. **Friedlander B, Morf M, Kailath T, and Ljung L:** New Inversion Formulas for Matrices Classified in Terms of Their Distance from Toiplitz Matrices. In *Linear Algebra and Its Applications,* Elsevier North Holland, Inc., 1978.
4. **Kaneda Y and Ohga J:** Adaptive microphone-array system for noise reduction. *J Acoust Soc Am* (76), 1984.
5. **Mitchell OMM, Ross CA, and Yates GH:** Signal processing for a cocktail party effect. *J Acoust Soc Am* (50):656(b), 1971.
6. **Parsons TW:** Separation of speech from interfering speech by means of harmonic selection. *J Acoust Soc Am* (60):911–918, 1976.
7. **Sambur MR:** Adaptive noise cancelling for speech signals. *IEEE Trans Acoust Speech Signal Processing* ASSP-26:419–423, 1977.
8. **Sondi MM and Elko GW:** Adaptive optimization of microphone arrays under a nonlinear constraint. *Proceedings ICASSP* 86(2):981–984, 1986.
9. **Strube W:** Separation of several speakers recorded by two microphones (cocktail party processing). *Signal Processing* (3):355–364, 1981.