An Enhanced Method and Behavioral Model for Noise Cancellation in Audio Devices

Ram Singh Rana [1, 2], Zhang Liang [1, 2], Tang Bin [1, 3] and Garg Hari Krishna [2]
[1]Institute of Microelectronics, 11 Science Park Road, Singapore 117685
[2]Department of Electrical & Computer Engineering, National University of Singapore, Singapore 119260
[3]Division of BioEngineering, National University of Singapore, Singapore 119260

ABSTRACT

In this paper, a beamforming based enhanced method for noise/echo cancellation in audio devices such as hearing aids is presented. A modified two-element beamformer scheme is proposed as a possible solution for better cancellation of noise. An equivalent behavioral model compatible with ADS (advance design system) CAD environment is provided. Simulations are carried out using MATLAB and ADS. Simulation results are observed in compliance with the modified two-element beamformer for noise cancellation and the equivalent behavioral model. The electrical equivalent of the proposed model is found compatible for the whole system simulation of biomedical devices, where noise cancellation is one of the critical concerns.

I. INTRODUCTION

With the advancements in integrated circuits technology the performance improvements of audio device, such as hearing aid devices [1], [2], have been more beneficial to the end users e.g. hearing impaired. However, an input to such device is often associated with the environmental noise. For instance, due to environmental noise, a hearing-impaired person not only feels severe hearing loss but is also unable to perceive desired speech from the environment noise. Thus, the noise cancellation is a primary concern to do the speech enhancement. Adaptive filtering, such as beamforming [10], is generally used as fundamental method for noise cancellation. The beamformer allows the signal pass and cancels the noise at the same time. However, the reported beamforming methods have their own limitations which allow noise cancellation only up to certain extent. Investigations to enhance further the cancellation of noise are desired.

While having better ways of canceling the noise, it needs to be verified before physical implementation. The behavior modeling [3]-[5] has been a conventional method to verify the functionality of the systems during the design and implementation phase.

In this paper, a modified two-element beamformer scheme is proposed as a possible solution for better cancellation of noise. It is supported with an equivalent behavioral model compatible to the industry standard CAD environment. A few behavioral level simulation results are provided.

The paper is structured as follows. The modified two-element beamformer for noise cancellation is provided in section II. The behavior model construction for the noise cancellation stage of a hearing aid like device is addressed in section III. In section IV, the simulation result for this behavior model is presented. The last section outlines the conclusions.

II. TWO-ELEMENT BEAMFORMER

The beamforming method is a more efficient method for noise cancellation in hearing aid device [6]-[8], which is based on the constrained adaptive beamformer of Grifliths and Jim [10]. A two-element microphone array beamformer is shown in Figure 1.

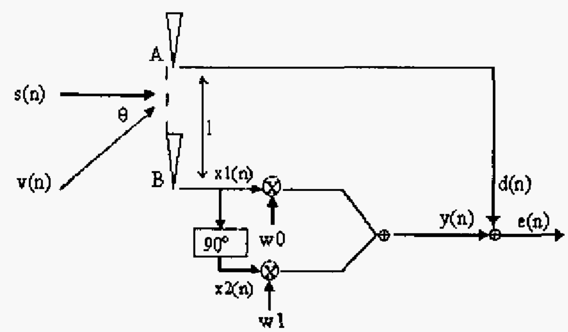

Fig. 1. A two-element microphone array beamformer

In Fig. 1, s(n) is the voice signal and v(n) is the noise. Devices A and B are two omni direction microphones without loss. The voice signal and the noise come to the microphones with various directions. The angel of voice signal is zero. The angel between voice signal and noise is θ. x1(n), x2(n), y(n) and d(n) are signals combined audio signal and noise. w0 and w1 are four coefficients which are determined by output e(n) from solving Wiener-Hope equation [8]. From the theoretic calculation [11], the signal-to-noise power spectral density ratio at the noise canceller output is equal to the inverse of the signal-to-noise power spectral

0-7803-8665-5/04/$20.00 ©2004 IEEE

density ratio at the reference input. This means that if the signal-to-noise power density ratio at the reference input is low, then a good cancellation of the noise at the output can be expected. However, the maxim gain of different arrival angle signal passing through is not at the point of zero degree arrival angle in some cases. As an attempt to handle such cases in a better way, a modified two-element beamformer is provided in Fig. 2.

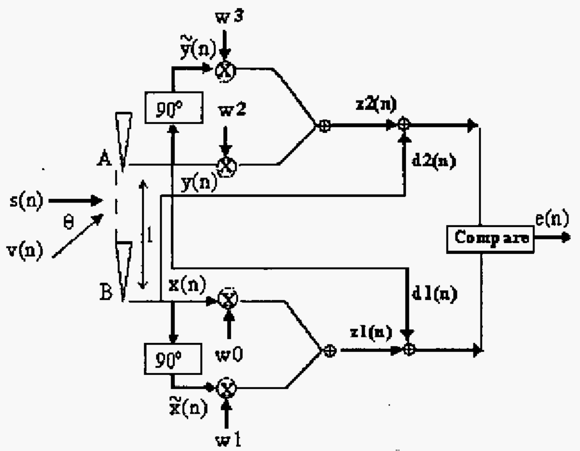

Fig. 2. A modified two-element beamformer

In Fig. 2, two independent beamforming paths are used and then their outputs are compared to select the better one. The comparison is to select the output with smaller signal to noise ratio (SNR). It, thus, enables the maxim gain of different arrival angle signal passing through this beamformer close to the point of zero degree arrival angle. This promises better noise cancellation performance than that in the case of Fig. 1. For the purpose of functional verification, behavioral modeling is considered which is briefly described below.

III. BEHAVIOR MODEL

As a case study, the model is constructed to be amenable with hearing aid devices, the front end of which is shown in Fig. 3. For instance, a typical hearing aid system has its front stage consisting of (i) signal receivers i.e. microphones (ii) amplifiers (iii) A/D converters and (iv) noise cancellation block. Its CAD compatible model enabling behavioral simulation is constructed as follow. It includes following steps: (i) The voice signal generator and interference generator are used to model the received voice and noise respectively, where both sources are defined as random signals in the frequency band of 0 kHz to 10 kHz which falls under audio range. (ii) The voice signal and the noise are combined and then transmitted after amplification into A/D converter to get the digital signals. (iii) The noise cancellation unit is used to process the digital signals using beamforming.

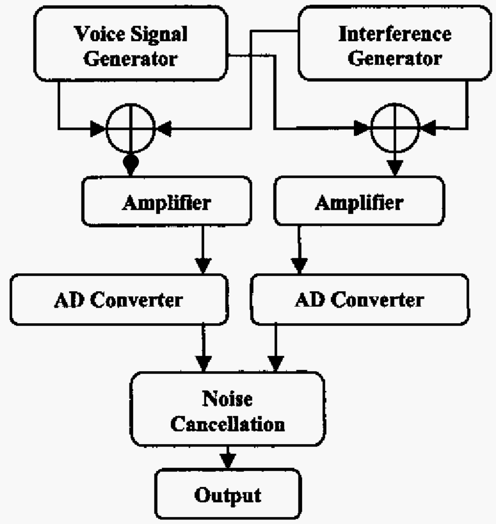

Fig. 3. A typical system using noise cancellation in an audio device

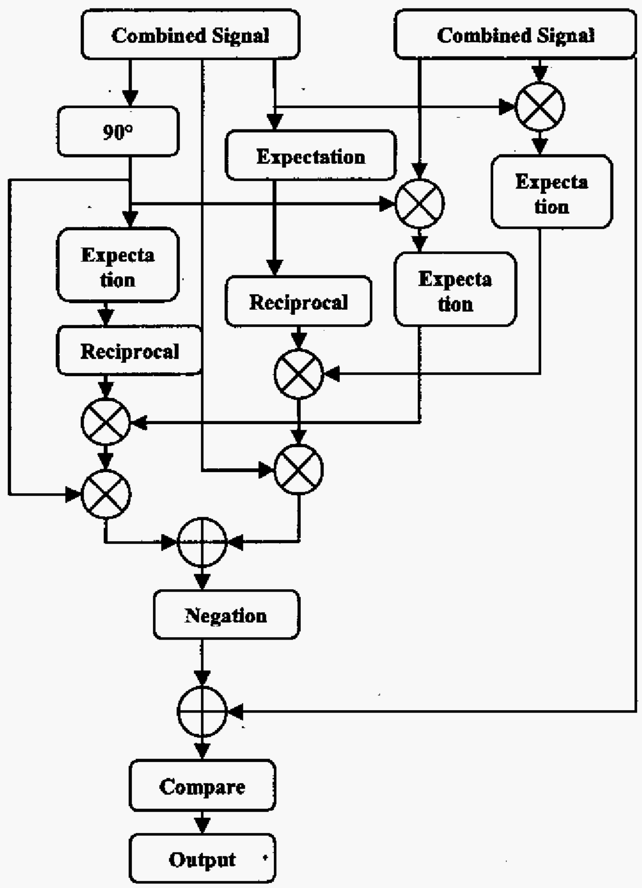

Fig. 4. Extended block diagram of single path beamforming

To develop the behavioral model, for the beamforming as shown in Fig. 2, compatible to MATLAB and ADS, the bock diagram as shown in Fig. 4 is used. Details of the developed models are not included here.

The gain of the amplifier in the behavior model is considered programmable. For simulation purpose, the sampling frequency of the 8 bit A/D converter used is taken 100 kHz, and the reference voltage of AD converter is defined 1.0 V.

In MATLAB, the behavior model is realized by programming the corresponding equations [8]. While in ADS, the behavior model is built using the modeling features of the software of ADS Agilent Ptolemy in behavior level [9]. All the functional blocks such as addition, multiplier and expectation are defined in ADS Agilent Ptolemy using equivalent models in ADS library.

The ADS has the feature for behavior models and circuit model building [9]. The above discussed behavioral models are also made ADS compatible which are found suitable for simulating audio devices consisting of RF, analog and/or digital blocks e.g. wireless hearing aid.

IV. SIMULATION RESULTS

In MATLAB, the voice and noise are all band limited signal. Using only one beamforming path as shown in Fig. 1, the simulated plot is shown in Fig. 5 for a set of input parameters such as the directional difference between noise and signal arrival, variance coefficients of signal and noise and centre frequency etc. The signal source arrival angle is fixed at 0° and the noise source angle is varied from 0° to 360° by a step of 1°. The curve in Fig. 5 shows the gain variation with variation in the noise angles. It is observed that the amplification in noise is less than one when input noise arrival angle is between the range of about 20° to 160° and the gain in signal is more than one when input signal arrival angle is between the range of about 0° to 20° or 160° to 180°. So when the signal and noise are received together from a difference in arrival angles, the amplification in signal and reduction in noise is observed.

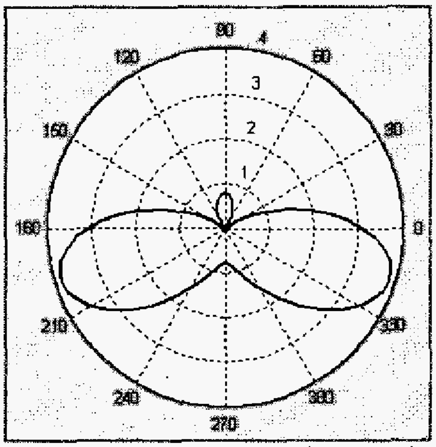

Fig. 5. The gain of different arrival angle noise with one beamforming path

However, as shown in Fig. 5 the noise cancellation performance is not good enough for the arrival noise angles from 180° to 360°. So the two beamforming paths method as shown in Fig. 2 is used and the achieved simulated plot using enhanced method of beamforming is shown in Fig. 6. The noise cancellation performance is symmetrical and is good enough as the gain in noise is not higher than gain in signal that is what is required. It ensures the proposed method overcomes the limitation of one beamforming path method. For instance in Fig. 5, when noise signal angle is 330°, the gain in noise signal is more than 3. However, in Fig. 6, the gain is less 3 for the same angle. The noise and signal sources used in Matlab simulation are defined as random signals i.e. the magnitude and frequency is varied with time.

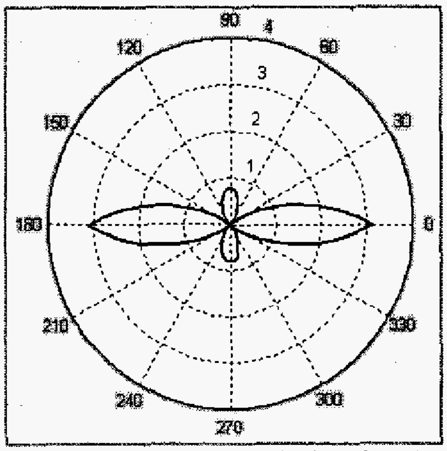

Fig. 6. The gain of different arrival angle noise with two beamforming path

The simulation of two beamforming path method is attempted also in ADS as well keeping in view that the ADS is a better choice for the whole system simulations, in general. Some results are given in Fig. 7 and Fig. 8 which show the differences in output and input SNR with respect to the variation in input signal frequency and input SNR respectively. The noise signal frequency is taken as 1 kHz.

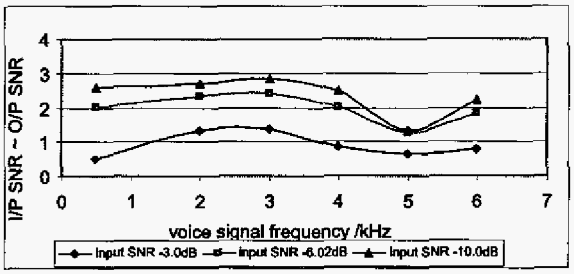

Fig. 7. The difference between input SNR and output SNR

Fig. 8 reflects that the difference between input SNR and output SNR decreases with increase in the input SNR. It may, however, be noted from the above results that there is an improvement in output SNR when input SNR is less than certain limit i.e. in this case about 0 dB. These results are observed in compliances with the modified method of noise cancellation as discussed above and also with the results achieved using MATLAB as shown in Fig. 6.

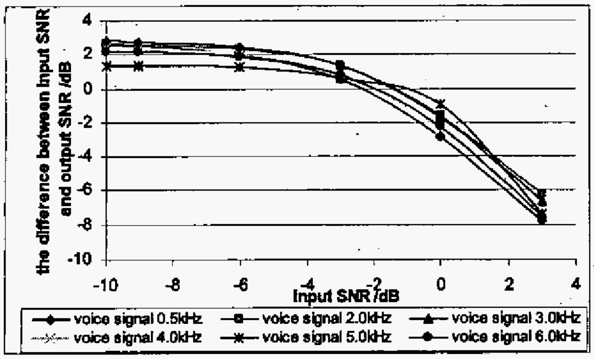

Fig. 8. The difference between input SNR and output SNR for different frequency

V. CONCLUSIONS

In this paper, an enhanced method for noise cancellation is discussed. It is supported with a behavior model for noise cancellation in audio devices such as hearing aid. Equivalent behavioral models are generated for carrying out simulation in MATLAB and ADS. The simulation results show the enhancement in the noise cancellation using the proposed scheme for noise cancellation. The model developed has been used in the whole system simulation of a RF link based hearing aid using ADS.

ACKNOLOGEMENT

The IME-NUS Joint Microelectronics Lab is acknowledged for scholarship support.

REFERENCES

[1] Armin Deiss and Qiuting Huang, "A low-power 200-MHz receiver for wireless hearing aid devices," *IEEE J. Solid-State Circuits*, vol. 38, pp. 793-804, May 2003

[2] D. George Gata, Walter Sjursen, et al., "A 1.1-V 270uA mixed-signal hearing aid chip," *IEEE J. Solid-State Circuits*, vol. 37, pp. 1670 - 1678, Dec. 2002

[3] Ram Singh Rana, "Computer aided system simulation of micro power CMOS analog hearing aid", *IEEE ASIC Conference, Portland USA,* pp. 343 - 347, Sept. 1997

[4] Matti Kajala and Matti Hamalainem, "Broadband beamforming optimization for speech enhancement in noisy environments", *Proc. 1999 IEEE Workshop on Applications of Signal Processing to Audio and Acoustics*, New Paltz, New York, pp.1881 - 1884, Oct. 17 - 20, 1999

[5] Min Li, Hector G. McAllister, Norman D. Black, and Trina Adrian De Perez, "Perceptual time-frequency subtraction algorithm for noise reduction in hearing aids", *IEEE Trans, Biomedical Engineering*, vol. 48, pp. 979 -988, Sept. 2001

[6] Desloge, J.G., Rabinowitz, W.M., Zurek, P.M., "Microphone-array hearing aids with binaural output. I. Fixed-processing systems", *IEEE Trans, Speech and Audio Processing,* vol. 5, pp. 529 - 542, Nov. 1997

[7] Welker, D.P., Greenberg, J.E., Desloge, J.G., Zurek P.M., "Microphone-array hearing aids with binaural output. II. A two-microphone adaptive system", *IEEE Trans, Speech and Audio Processing,* vol. 5, pp. 543 - 551, Nov. 1997

[8] B. Farhang-Boroujeny, *Adaptive Filters—Theory and Application*, pp. 75 - 81, McGraw-Hill, 1998

[9] *Agilent Ptolemy Simulation Manual*, Agilent Technologies, Sept. 2002

[10] Lloyd J. Grifliths and Charles W. Jim, "An alternative approach to linearly constrained adaptive beamforming", *IEEE Trans, Antennas and Propagation,* vol. 30, pp. 27 - 34, Jan. 1982

[11] B. Widrow and S. T. Stearns, *Adaptive Signal Processing*, Prentice-Hall, 1985