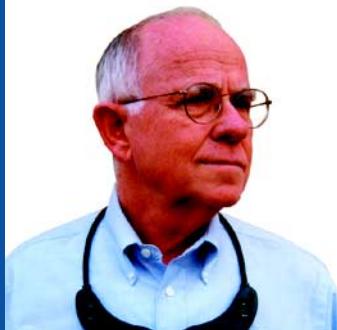




A Microphone Array for Hearing Aids



by
**Bernard
Widrow**



Abstract—A directional acoustic receiving system is constructed in the form of a necklace including an array of two or more microphones mounted on a housing supported on the chest of a user by a conducting loop encircling the user's neck. Signal processing electronics contained in the same housing receive and combine the microphone signals in such a manner as to provide an amplified output signal which emphasizes sounds of interest arriving in a direction forward of the user. The amplified output signal drives the supporting conducting loop to produce a representative magnetic field. An electroacoustic transducer including a magnetic field pick up coil for receiving the magnetic field is mounted in or on the user's ear and generates an acoustic signal representative of the sounds of interest. The microphone output signals are weighted (scaled) and combined to achieve desired spatial directivity responses. The weighting coefficients are determined by an optimization process. By bandpass filtering the weighted microphone signals, with a set of filters covering the audio frequency range, and summing the filtered signals, a receiving microphone array with a small aperture size is caused to have a directivity pattern that is essentially uniform over frequency in two or three dimensions. This method enables the design of highly-directive-hearing instruments which are comfortable, inconspicuous, and convenient to use. The array provides the user with a dramatic improvement in speech perception over existing hearing aid designs, particularly in the presence of background noise, reverberation, and feedback.

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Microphone Array for Hearing Aids

There is a big difference between hearing speech and understanding speech. Most hearing-impaired people will be able to hear speech when given sufficient amplification from their hearing aids. In many cases, however, they will hear but will not understand.

The benefits of amplification alone are limited. In a noisy place, hearing aids will amplify the noise as well as the desired speech signal. In a reverberant place, hearing aids will amplify late multipath arrivals as well as the direct first-arrival signal. Furthermore, feedback associated with high output hearing aids distorts the frequency response of the hearing aid, which was carefully tuned to compensate for the individual's hearing loss; and sometimes causes oscillation.

We describe a microphone array for hearing aids that overcomes some of these limitations and has the capability of enhancing speech understand-



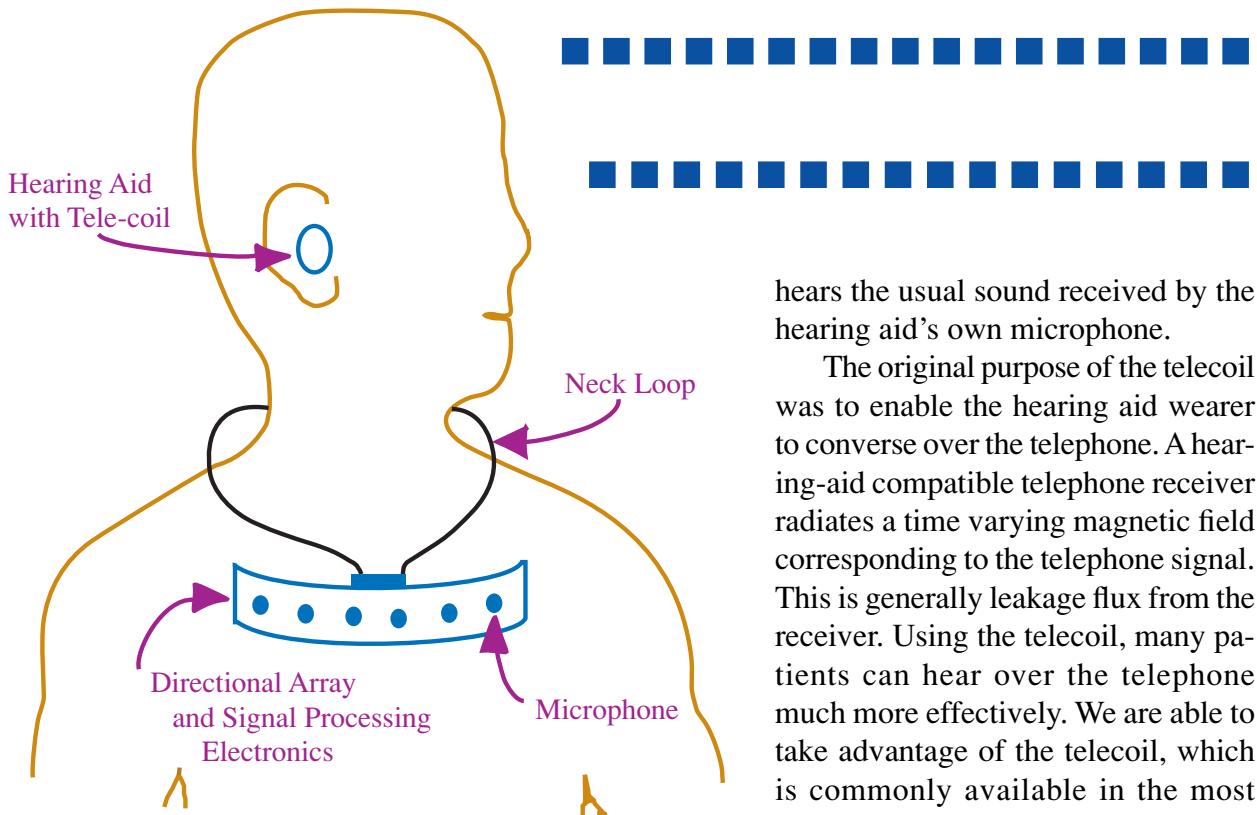


Figure 1. Chest-mounted “necklace”, directional microphone array with neck loop, and hearing aid with telecoil.

ing for hearing-impaired patients. The microphone array is worn on the chest as part of a necklace, in accord with the diagram of Fig. 1. A processed signal from the array drives current through a conducting neck loop thus creating a time-variable magnetic field that is representative of the received sound. The magnetic field provides a wireless means for carrying the sound signal to conventional hearing aid devices located in the ears of the wearer. In order to receive the signal, the hearing aid must be equipped with a “telecoil”, a small induction coil contained within the hearing aid whose output can be switch selected by the wearer to serve in place of the hearing aid’s microphone signal. When switching the hearing aid to telecoil position, the wearer hears the sound received by the array. When switching the hearing aid to the microphone position, the wearer

hears the usual sound received by the hearing aid’s own microphone.

The original purpose of the telecoil was to enable the hearing aid wearer to converse over the telephone. A hearing-aid compatible telephone receiver radiates a time varying magnetic field corresponding to the telephone signal. This is generally leakage flux from the receiver. Using the telecoil, many patients can hear over the telephone much more effectively. We are able to take advantage of the telecoil, which is commonly available in the most powerful behind-the-ear hearing aid types, to provide a wireless link between the chest-mounted array and the hearing aid. Telecoils can be fitted to almost all hearing aids.

Use of the array enhances the patient’s hearing in the following three ways.

Signal-to-noise ratio

The array enhances signal-to-noise ratio. The patient aims his or her body toward the person who is speaking. The array beam is 60° wide in both azimuth and elevation. The sound in the beam is enhanced relative to the surrounding sound. The speech of interest is enhanced relative to omnidirectional background noise by about 10 dB, from about 200 Hz to 6 kHz. The gains of the array sidelobes vary between 20-35 db below the gain at the center of the main beam.

Effects of reverberation

The array reduces the effects of reverberation. Because the array is gen-

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erally steered toward the sound of interest, the direct primary path is thus aligned with the beam. The secondary paths for the most part arrive at angles outside the beam and are thus attenuated by the array. Reducing reverberation enhances sound clarity since the ear and the brain are somewhat confused by multiple arrivals. This is especially the case with hearing-impaired individuals.

Feedback

Use of the array reduces feedback by about 15 dB, because the chest is at a much greater distance from the hearing-aid loudspeaker than is the microphone on the hearing aid itself. Reduction of feedback makes available louder sound for the patient, without oscillation, and allows the hearing aid to function with a frequency response closer to the desired compensation curve.

The current array design and geometry are shown in Fig. 2. The device is comprised of an array of six microphones, four pushbuttons for control, and a plastic case designed to fit both the adult male and female torso. The plastic case was designed by computer, completely specified in software. It contains batteries and all of the signal processing electronics. Two custom ASIC chips were designed for this device, one for signal processing and the other to serve as an interface between a PC computer and the signal processing chip when this chip is being programmed. Custom chips were needed because of the tight space require-

ments and the requirements for low battery drain.

In this device, the audio spectrum from 200 Hz to 6 kHz is divided into twelve bands, each with its own digital gain control. The six microphone signals are amplified and weighted and then fed to each of the twelve band-pass filters. Different microphone-signal weightings were designed for each frequency band so that the beam width was able to be held at approximately 60° over the entire frequency range of interest. The microphone weights were designed off-line by using adaptive beamforming techniques to achieve the desired beam shape and to achieve a specified robustness to inherent variations in microphone characteristics. A least square error criterion was used for the design. Anechoic chamber testing was used to verify the de-

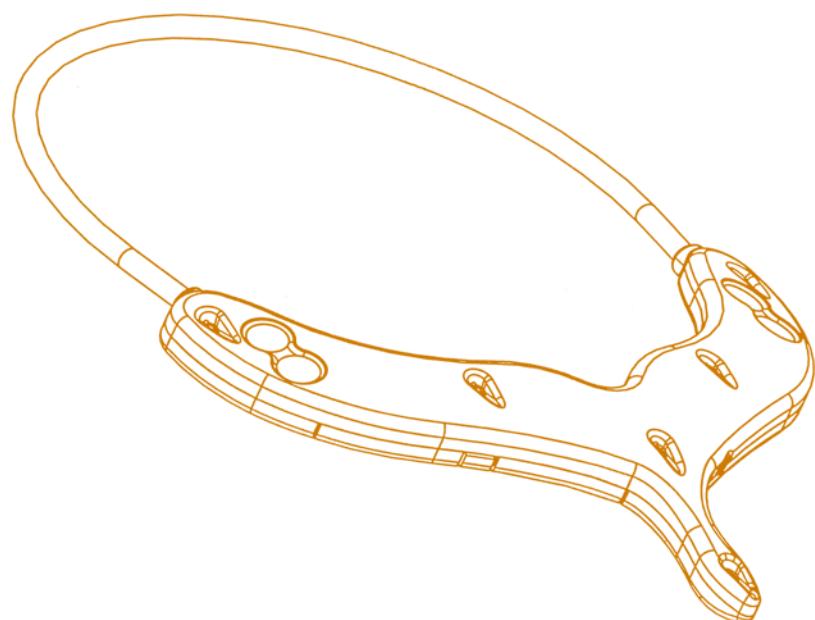


Figure 2: The current array design geometry.

sign. Theoretical and measured beam patterns turned out to be remarkably close.

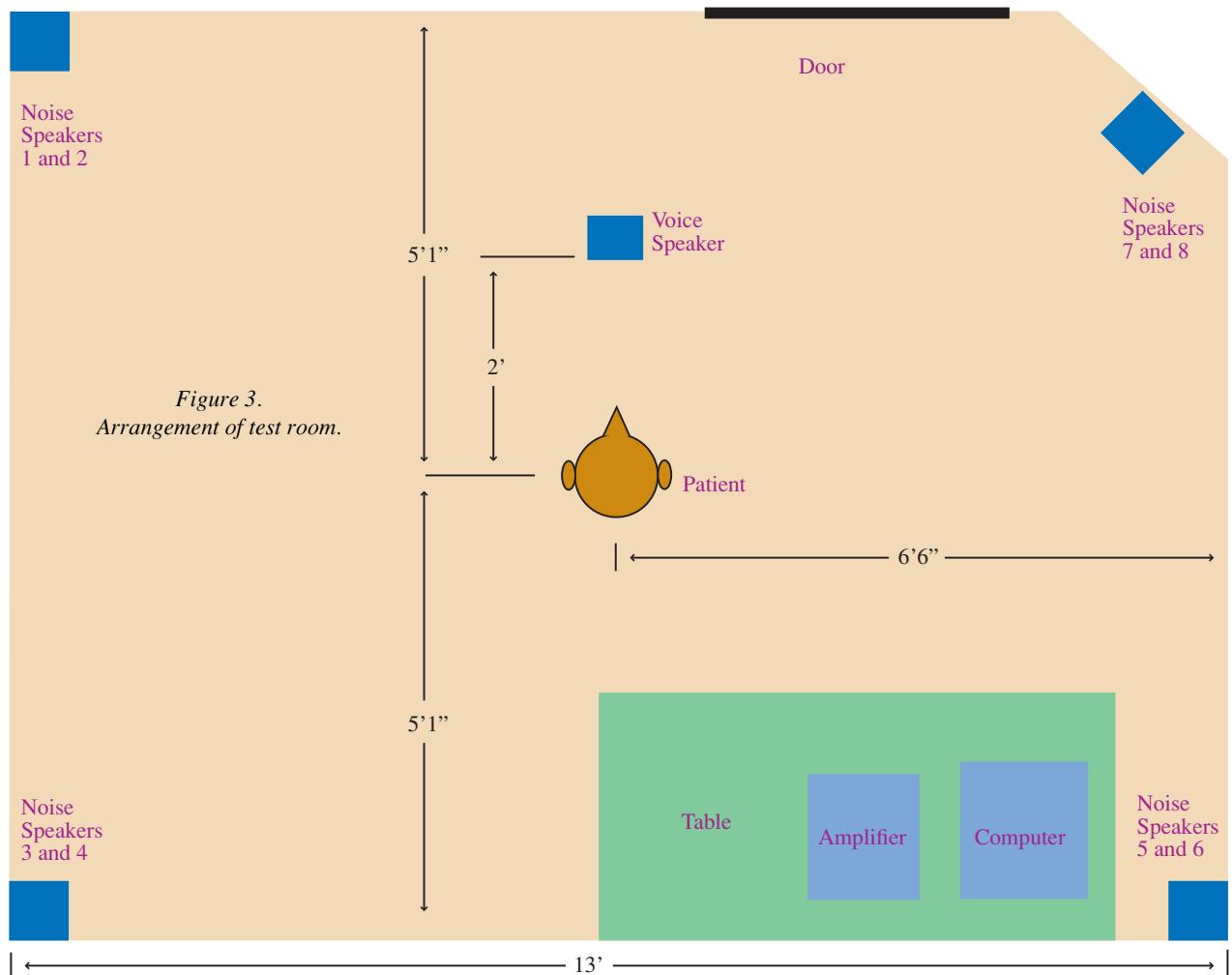
U.S. Patent number 5,793,857 has been granted to Michael A. Lehr and Bernard Widrow for this technology. Canadian, European, and Australian patents have been granted, and patents are pending in other countries.

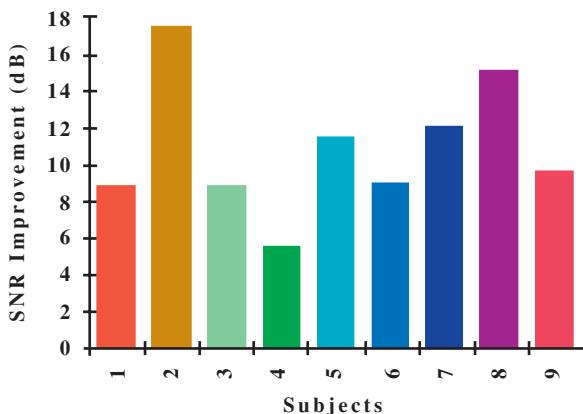
Patient testing was performed to evaluate the effectiveness of the microphone array and to compare listening with the hearing aid alone with listening to the array and hearing aid in telecoil mode. Fig. 3 shows the floor arrangement of the test room. The patient was seated before a loudspeaker that carried the sound of a male test voice. Four loudspeakers on the floor in the four corners of the room carried

spectrally weighted bandpass noise. Four additional loudspeakers in the four corners at the ceiling were also used to carry the same noise. The room was not anechoic but had some sound damping. The noise carried by the eight corner loudspeakers produced a noise field that was approximately isotropic.

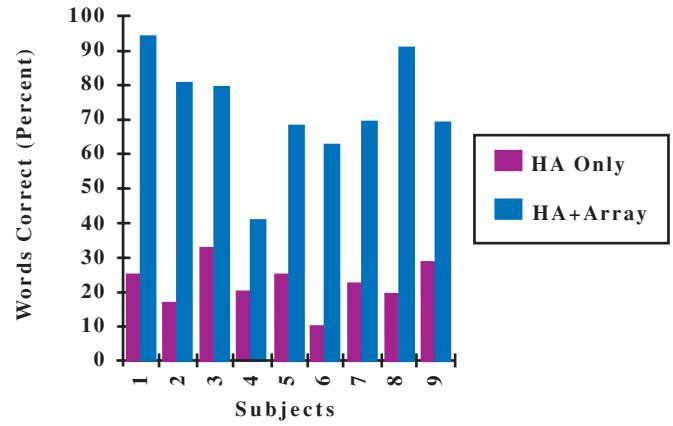
The test voice and the test noise were stored in a PC computer. The voice and noise data were obtained from Dr. Sig Soli of the House Ear Institute in Los Angeles. We performed a modified version of his HINT test (hearing and noise test).

With the patient seated at a prescribed location marked on the floor, the volume control of the hearing aid and the volume control of the array





(a)



(b)

Figure 4. Results obtained on HINT and Intelligibility tests. (a) Magnitude of improvement in sentence speech recognition threshold in noise (HINT) with the microphone array, in comparison with the hearing aid alone. (b) Percent speech intelligibility in the presence of noise. For each test subject, the purple bar is the result obtained using the hearing aid and the blue bar is the result using the array.

were set so that the measured volume delivered to the patient's ear would be the same when listening to the test voice through the hearing aid and through the array. The volume level of the test voice was set to be comfortable for the patient, in the absence of noise.

Word phrases were spoken to the patient by the test voice, with some noise applied. The patient was asked to repeat the words. If any word in the phrase was repeated incorrectly, the response was considered to be incorrect. The noise level was reduced by 2 dB, and another randomly chosen phrase was read. If the response was incorrect again, the noise was lowered by another 2 dB and so forth. When a correct response was obtained, the noise level for the next phase was raised by 2 dB. If another correct response was obtained, the noise level was raised by another 2 dB and so forth. The noise level went up and down, and the average noise level was observed over ten or twenty phrases.

The average noise level when using the hearing aid was compared to that when using the array. The improvement in signal-to-noise ratio

when using the array is plotted in Fig. 4(a) for nine test patients. This improvement averages more than 10 dB, which is consistent with anechoic chamber measurements and theoretical calculations.

Other testing was done with the noise volume fixed and the volume level of the test voice fixed. Individual words randomly selected were presented by the test voice. The responses of the patients were observed when using the hearing aid, and when using the array. The results are shown for the same nine patients, in Fig. 4(b). Patient #1 had a 25% correct response with the hearing aid, and a 95% correct response with the array. Patient #2 had a 15% correct response with the hearing aid, and an 80% correct response with the array. And so forth. These improvements are rather dramatic.

One young woman in Palo Alto, California, has been wearing one of these devices on a daily basis over the past five years. As the design evolved, she always had the latest for testing. She is totally deaf in one ear and is 95–105 db below normal in her “good” ear. Using her hearing aid and with



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profound ones.

The microphone array devices are now being manufactured and marketed by Starkey Laboratories, 6700 Washington Ave, Eden Prairie, MN, 55344, U.S.A. The trade name for the device is Radiant Beam Array (RBA). It is the most powerful hearing device on the market. It remains to be seen how well it will be accepted by the hearing-impaired community.

Bernard Widrow received the S.B., M.S. and Sc.D. degrees in electrical engineering from the Massachusetts Institute of Technology in 1951, 1953, and 1956, respectively. He joined the MIT faculty and taught there from 1956–1959. In 1959, he joined the faculty of Stanford University, where he is currently Professor of Electrical Engineering.

Dr. Widrow is a Life Fellow of the IEEE and a fellow of AAAS. He received the IEEE Centennial Medal in 1984, the IEEE Alexander Graham Bell medal in 1986, the IEEE Neural Networks Pioneer Medal in 1991, the IEEE Signal Processing Society Medal in 1998, the IEEE Millennium Medal in 2000, and the Benjamin Franklin Medal of the Franklin Institute in 2001. He was inducted into the National Academy of Engineering in 1995, and into the Silicon Valley Engineering Council Hall of fame in 1999.

Dr. Widrow is a past president and currently a member of the Governing Board of the international Neural Network Society. He is associate editor of several journals, and is the author of about 100 technical papers and 15 patents. He is co-author of *Adaptive Signal Processing* and *Adaptive Inverse Control*, both Prentice Hall books. A new book, *Quantization Noise*, is in preparation.