Evaluation of a real-time adaptive-beamforming hearing aid

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10:05

II3. Addressing Hugh Knowles' fundamental question: What would "high fidelity" mean for a hearing aid wearer? Mead C. Killion (Etymotic Research, 61 Martin Lane, Elk Grove Village, IL 60007)

First assigned by Hugh Knowles to update Hugh's earlier estimate of the difference between real-ear and 2-cm^3 coupler frequency-response curves for hearing aids, the author later did his Ph.D. research on the topic of the design and evaluation of high-fidelity hearing aids. Throughout this research, Hugh's probing questions were both useful and sometimes embarrassing. He made it impossible to give superficial answers to difficult questions about (1) the occlusion-effect ("hollow voice") problem with closed-mold fittings, (2) what gain and frequency response would be required for a *perception* of high-fidelity sound by someone with a substantial high-frequency hearing loss, and (3) what directional/frequency-response accuracy in the hearing aid would be required for a "high fidelity" sense of the auditory space. Partly, as a result of Hugh's questions and contributions, a good solution has now been obtained to the occlusion-effect problem (reinventing Zwislocki's 1953 solution that he thought probably impractical), there are now transducers and "acoustic plumbing" techniques permitting 40-Hz to 16-kHz reproduction in hearing aids with fidelity comparable to that of professional studio monitors, in-the-canal hearing aids with directional properties comparable to that of the unaided ear, and a new integrated-circuit hearing-aid amplifier chip with level-dependent high-frequency emphasis that is designed to provide nearly "transparent" amplification to the wearer. Recorded demonstrations will be played if time permits.

10:30

II4. Electroacoustic test methods for evaluation of hearing-aid performance. L. B. Beck (Veterans Administration, Medical Center, 50 Irving Street, N.W., Washington, DC 20422), E. D. Burnett (National Institute for Standards and Technology, Gaithersburg, MD 20899), and G. D. Causey (Catholic University of America, Washington, DC 20064)

The purpose of this paper is to describe the development of electroacoustic test procedures for evaluation of hearing aids by the Veterans Administration (VA). These methods have evolved from the use of the 2-cm^3 coupler with pure tones as the input stimuli to the use of the KEMAR manikin with various types of complex noise as input stimuli. Hugh Knowles served as a consultant to the VA and his efforts, both for the VA and the research and development community at large, formed the framework for the evaluation program in place today. His contribution and leadership have resulted in procedures that permit the evaluation of a hearing aid in a manner very similar to the way it will be used by the wearer. Current use of FFT analysis and strategies for evaluating noise reduction circuits and in-the-ear hearing aids will be highlighted.

10:55

II5. The acoustics of the external ear: Old problems and fresh perspectives. E. A. G. Shaw (Division of Physics, National Research Council Canada, Ottawa, Ontario K1A 0R6, Canada)

A coherent picture of the external ear, operating both in the free field and as an enclosed receiver, came into focus more than a decade ago though several troublesome problems remained, particularly at frequencies greater than a few kHz. These included uncertainty about the geometry and acoustical characteristics of the ear canal, uncertainty about the dynamics of the eardrum and the reflection of sound from the eardrum, uncertainty about the role of the external ear in sound localization and, above all, uncertainty about the acoustical interactions between earphones and ears. Considerable progress has been made in most of these areas during the past few years but some problems remain intractable. In the meantime, the scientific and technological advances of the 1970s and early 1980s have provided us with a legacy of valuable instruments: eardrum simulators, artificial heads especially the Knowles Electronic Manikin (KEMAR), microphones for probing ear canal pressure, and improved insert earphones.

Contributed Papers

11:20

II6. Evaluation of a real-time adaptive-beamforming hearing aid. Julie E. Greenberg and Patrick M. Zurek (MIT Research Laboratory of Electronics, Room 36-761, Cambridge, MA 02139)

A real-time two-microphone monaural hearing aid, based on the constrained adaptive beamformer proposed by Griffiths and Jim [IEEE Trans. Antennas Propag. **AP-30**, 27–34 (1982)], has been implemented and evaluated. The beamformer adapts to preserve the target signal (assumed to be straight ahead) and to minimize the power of jammer signals arriving from all other directions. The basic Griffiths-Jim algorithm is augmented with a method to inhibit adaptation in the presence of strong target signals [Greenberg *et al.*, J. Acoust. Soc. Am. Suppl. 1 **85**, S26 (1989)]. The real-time system employs the Motorola DSP56001 and was evaluated using a speech target and a single jammer for a variety of free-space and head-mounted microphone configurations in anechoic and reverberant environments. The system demonstrates very good performance (30- to 40-dB gain from input to output in an intelligibility-weighted spectral average of target-to-jammer ratio) for several broadside microphone configurations in the anechoic environment. Adaptation inhibition provides a robust insensitivity to target misalignment at high-input target-to-jammer ratios. As expected, performance degrades with increasing reverberation; at the critical distance, the system provides 5- to 10-dB gain. In extreme reverberation, the system performs at least as well as a delay-and-sum beamformer. [Work supported by NIH.]

11:35

II7. Optimization of hearing-aid gain and frequency response for cochlear hearing losses. Arne Leijon (Department of Information Theory, Chalmers University of Technology, S-412 96 Gothenburg, Sweden)

A new theoretical procedure was developed for optimizing the acoustic characteristics of hearing aids. The algorithm increases predicted speech intelligibility while keeping the estimated loudness of amplified speech at a predetermined level. The intelligibility criterion was the mutual information between spoken phonetic messages and the corresponding streams of auditory patterns in a psychoacoustic model of the listener's auditory system. The speech signal was modeled as a sequence of allophone transitions. Each transition was represented by three auditory patterns, each containing ten cepstrum coefficients, one voice and one duration parameter. The shape of optimized hearing-aid frequency responses depended strongly on the desired loudness of amplified speech. When normal loudness was desired for speech at 65 dB SPL, and the optimization was initialized with the NAL prescription, hearing-aid gain was reduced, but predicted speech intelligibility was nearly unchanged. Predicted intelligibility of amplified speech in a noisy background was similar for several different frequency responses. The mutual information was highly correlated with a modified version of the articulation index.

THURSDAY MORNING, 30 NOVEMBER 1989 ST. LOUIS BALLROOM H, 9:00 TO 11:45 A.M.

Session JJ. Underwater Acoustics V: The Sea Surface—Noise and Scattering

Eric I. Thorsos, Chairman

Applied Physics Laboratory, University of Washington, Seattle, Washington 98195

Chairman's Introduction—9:00

Contributed Papers

9:05

JJ1. Further studies of the underwater noise produced by rainfall. Paul A. Elmore, Hugh C. Pumphrey, and Lawrence A. Crum (National Center for Physical Acoustics, University of Mississippi, University, MS 38677)

A study of the sound produced by water drops striking a water surface has confirmed some earlier results [Pumphrey *et al.*, J. Acoust. Soc. Am. **85**, 1518 (1989)]. In particular, for a certain well-defined range of drop sizes and impact velocities, drops will predictably and repeatedly entrain bubbles; this phenomenon has been named regular entrainment. In the present study, various fixed drop diameters have been used to investigate how bubble frequency, dipole strength, and time between drop impact and bubble formation vary with impact velocity. It is found that as impact velocity is increased through the point where entrainment begins, both frequency and dipole strength decrease to a minimum value and then rise again as the highest velocity at which entrainment occurs is approached. Both terms show increased variability near the critical upper and lower velocities. The frequency tends to increase monotonically as drop size is reduced; drops that entrain bubbles at their terminal velocities tend to produce frequencies near 14 kHz, which is also the peak frequency of the natural rainfall spectrum. Finally, the time between drop impact and bubble formation was found to increase monotonically as drop size or impact velocity increases. [Work supported by Office of Naval Research.]

9:20

JJ2. An experimental investigation of bubble clouds as sources of ambient noise. S. W. Yoon,[a] L. A. Crum (National Center for Physical Acoustics, University of Mississippi, University, MS 38677), and A. Prosperetti (Department of Mechanical Engineering, The Johns Hopkins University, Baltimore, MD 21218)

Collective motions of bubble clouds are experimentally investigated as an ocean ambient noise source. The experiments, performed in a water tank, show that the frequency due to the bubble column resonance depends inversely on the logarithm of the void fraction of the water/bubble mixture. It is also observed that the frequency depends inversely on the volume of the bubble column. These experimental results indicate that the volume resonance of the bubble column is one of the more likely mechanisms for an ocean ambient noise source between 100 and 1000 Hz. [Work supported by ONR and KOSEF (SWY).] [a] On leave from the Department of Physics, Sung Kyun Kwan University, Republic of Korea.

9:35

JJ3. Underwater sound generated by impacts and bubbles from raindrops at oblique incidence. Armagan Kurgan,[a] Herman Medwin, and Jeffrey A. Nystuen (Departments of Physics and Oceanography, Naval Postgraduate School, Monterey, CA 93943)

The sound generated by rainfall at sea is caused by raindrops of a wide size range that fall at their terminal velocities and generally strike the ocean surface at local, oblique incidence. Laboratory experiments have been conducted to evaluate the energy spectrum of sounds caused by both the impact and the bubble formed by single raindrops. The results, using terminal velocities and oblique trajectories, are very different from the published normal incidence, nonterminal-velocity characterizations. For example, bubble frequencies other than the well-known 14-kHz peak are found. Also, the energy of the impact sound increases significantly for larger drops and for large deviations from normal trajectories. These observations provide specific reasons for the known broadening of the 14-kHz spectral peak of rain noise in the presence of winds at sea. [Work supported by ONR.] [a] Lt., Turkish Navy.