III. CONCLUSIONS AND RECOMMENDATIONS

The dispersion characteristics of the wave can be readily and efficiently observed by analyzing transient waveforms of an initially impulsive wave using the STFT. The MsSs provide a simple and convenient means for transmitting an initial impulse and detecting the transient waveforms. Since the MsSs require no direct physical contact, they do not obstruct the wave propagation and, thus, are particularly well suited for investigation of wave-propagation properties.

The method used in this experiment can also be applied to investigation of wave dispersion in other structures such as rods and shells with regular and irregular geometries. Additional experimental investigations of wave propagation properties in these other structures are thus recommended to validate theory and numerical calculations. Also, additional sensor and instrumentation development is recommended to further extend the frequency range of the investigation.

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[1] E. Giebe and E. Blechschmidt, "Experimentelle und Theoretische Untersuchungen uber Dehnungseigenschwingungen von Stäben und Röhnen," Ann. Phys. (Leipzig) **18**, 417–457 (1933).

[2] J. H. Heimann and H. Kolsky, "The Propagation of Elastic Waves in Thin Cylindrical Shells," J. Mech. Phys. Solids **14**, 121–130 (1966).

[3] R. W. Mortimer, J. L. Rose, and P. C. Chou, "Longitudinal Impact of Cylindrical Shells," Exp. Mech. **12**, 25–31 (1972).

[4] W. Mohr and P. Höller, "On Inspection of Thin-Walled Tubes for Transverse and Longitudinal Flaws by Guided Ultrasonic Waves," IEEE Trans. Sonics Ultrason. **SU-23**, 369–374 (1976).

[5] H. Kwun and C. M. Teller, "Detection of Fractured Wires in Steel Cables Using Magnetostrictive Sensors," Mat. Eval. **52**, 503–507 (1994).

[6] H. Kwun and C. M. Teller, "Magnetostrictive Generation and Detection of Longitudinal, Torsional, and Flexural Waves in a Steel Rod," J. Acoust. Soc. Am. **96**, 1202–1204 (1994).

[7] C. H. Hodges, J. Power, and J. Woodhouse, "The Use of the Sonogram in Structural Acoustics and an Application to the Vibrations of Cylindrical Shells," J. Sound Vib. **101**, 203–218 (1985).

[8] F. Hlawatsch and G. F. Boudreaux-Bartels, "Linear and Quadratic Time-Frequency Signal Representations," IEEE Sig. Proc. Mag. **9(2)**, 21–67 (April 1992).

[9] T. Önsay and A. G. Haddow, "Wavelet Transform Analysis of Transient Wave Propagation in a Dispersive Medium," J. Acoust. Soc. Am. **95**, 1441–1449 (1994).

[10] M. C. Junger and F. J. Rosato, "The Propagation of Elastic Waves in Thin-Walled Cylindrical Shells," J. Acoust. Soc. Am. **26**, 709–713 (1954).

[11] P. M. Naghdi and R. M. Cooper, "Propagation of Elastic Waves in Cylindrical Shells, Including the Effects of Transverse Shear and Rotatory Inertia," J. Acoust. Soc. Am. **28**, 56–63 (1956).

[12] I. Mirsky and G. Hermann, "Axially Symmetric Motions of Thick Cylindrical Shells," J. Appl. Mech. **25**, 97–102 (1958).

[13] J. E. Greenspan, "Axially Symmetric Vibrations of a Thick Cylindrical Shell in an Acoustic Medium," J. Acoust. Soc. Am. **32**, 1017–1025 (1960).

HRTF measurements of a KEMAR [43.66.Pn, 43.66.Qp]

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An extensive set of head-related transfer function (HRTF) measurements of a Knowles Electronics Mannequin for Acoustic Research (KEMAR) has recently been completed. The measurements consist of the left and right ear impulse responses from a Realistic Optimus Pro 7 loudspeaker mounted 1.4 m from the KEMAR. Maximum length (ML) pseudorandom binary sequences were used to obtain the impulse responses at a sampling rate of 44.1 kHz. In total, 710 different positions were sampled at elevations from −40 deg to +90 deg. These data are being made available to the research community on the Internet via anonymous FTP and the World Wide Web.

It is generally accepted that the auditory cues for sound localization are embodied in the transformation of sound by the torso, head, and external ear. A head-related transfer function (HRTF) is a frequency response describing the pressure transformation from a specific free field source position to the eardrum. Sets of HRTFs are often needed for the study of sound localization and for the synthesis of spatial cues, but measuring HRTFs using human subjects or dummy-head microphones is a laborious task. This technical note describes a set of HRTF measurements made using a Knowles Electronics Mannequin for Acoustic Research (KEMAR) which is being made available to the research community on the Internet.

Measurements were made using an Apple Macintosh computer equipped with a Digidesign Audiomedia II DSP card, which has 16-bit stereo A/D and D/A converters that operate at a 44.1-kHz sampling rate. One of the audio output channels was sent to an amplifier which drove a Realistic Optimus Pro 7 loudspeaker, a small two-way loudspeaker with a 4-in. woofer and 1-in. tweeter. The KEMAR was Knowles Electronics model DB-4004 and was configured with two neck rings and a torso. The left pinna was the "small" model DB-061 and the right was the "large red" model DB-065. The KEMAR was equipped with Etymotic ER-11 microphones, Etymotic ER-11 preamplifiers, and DB-100 occluded ear simulators with DB-050 ear canal extensions. The outputs of the microphone preamplifiers were connected to the stereo inputs of the Audiomedia card.

From the standpoint of the Audiomedia card, a signal sent to the audio outputs results in a corresponding signal appearing at the audio inputs. Measuring the impulse response of this system yields the impulse response of the combined system consisting of the Audiomedia D/A and A/D converters and anti-alias filters, the amplifier, the speaker, the room in which the measurements are made, and most importantly, the response of the KEMAR with its associated microphones and preamps. Interference due to room reflections can be avoided by ensuring that any reflections occur well after the head response time, which is several milliseconds.

The measurements were made in MIT's anechoic chamber. The KEMAR was mounted upright on a motorized turntable which could be rotated accurately to any azimuth under computer control. The speaker was mounted on a boom stand which enabled accurate positioning of the speaker to any elevation with respect to the KEMAR. Thus the measurements were made one elevation at a time, by setting the speaker to the proper elevation and then rotating the KEMAR to each azimuth. With the KEMAR facing forward toward the speaker (0 deg azimuth), the speaker was positioned such that a normal ray projected from the center of the face of the speaker bisected the interaural axis of the KEMAR at a distance of 1.4 m. It is believed that the speaker was always within 1.5 cm of the desired position, which corresponds to an angular error of ±0.5 deg.

The impulse responses were obtained using a maximum length (ML) sequence measurement technique.[1,2] The sequence length was 16383 samples, corresponding to a 14-bit generating register. This sequence length was chosen to yield a good signal-to-noise ratio (SNR) without excessive storage requirements or computation time. Because the measurements were performed in an anechoic chamber and the ML sequence was sufficiently long, time aliasing in the impulse responses was not significant. The measured SNR for frontal incidence was 65 dB.

The spherical space around the KEMAR was sampled at elevations from −40 deg (40 deg below the horizontal plane) to +90 deg (directly overhead) in 10-deg increments. At each elevation, a full 360 deg of azimuth was sampled in equal sized increments. The azimuth increment sizes were chosen to maintain approximately 5-deg great-circle increments. Table I shows the number of samples and azimuth increment at each elevation (all angles in degrees). In total, 710 locations were sampled.

TABLE I. Number of measurements and azimuth increment at each elevation. All angles are in degrees.

Elevation	Number of measurements	Azimuth increment
−40	56	6.43
−30	60	6.00
−20	72	5.00
−10	72	5.00
0	72	5.00
10	72	5.00
20	72	5.00
30	60	6.00
40	56	6.43
50	45	8.00
60	36	10.00
70	24	15.00
80	12	30.00
90	1	x

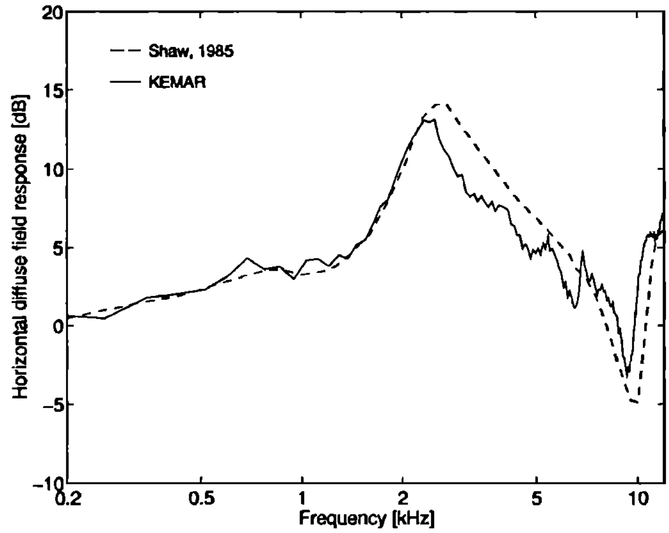

FIG. 1. Comparison of horizontal diffuse field response from Shaw's data and KEMAR data.

It was desired to obtain HRTFs for both the "small" and "large red" pinna styles. If the KEMAR had perfect medial symmetry, including the pinnae, then the resulting set of HRTF measurements would be symmetric within the limits of measurement accuracy. In other words, the left ear response at azimuth θ would be equal to the right ear response at azimuth 360-θ. It was decided that an efficient way to obtain symmetrical HRTF measurements for both the "small" and "large red" pinnae was to install both pinnae on the KEMAR simultaneously, and measure the entire 360-deg azimuth circle. This yields a complete set of symmetrical responses for each of the two pinna, by associating each measurement at azimuth θ with the corresponding measurement at azimuth 360-θ. For example, to form the symmetrical response pair for the "small" pinna (which was mounted on the left ear), given a source location at 45 deg right azimuth, the left ear response at 45 deg (contralateral response) would be paired with the left ear response at 315 deg azimuth (simulated ipsilateral response). Such a symmetrical set will not exhibit interaural differences for sources in the median plane, which has been shown to be a localization cue.[3] Assuming an HRTF is negligibly affected by the shape of the opposite pinna, these symmetrical sets should be the same as sets obtained using matched pinnae.

Each HRTF measurement yielded a 16 383-point impulse response at a 44.1-kHz sampling rate. Most of these data are irrelevant. The 1.4-m air travel corresponds to approximately 180 samples, and there is an additional delay of 50 samples inherent in the playback/recording system. Consequently, in each impulse response, there is a delay of approximately 230 samples followed by the head response, which persists for several hundred samples and is in turn followed by reflections off objects in the anechoic chamber (such as the KEMAR turntable). In order to reduce the size of the data set without eliminating anything of potential interest, the first 200 samples of each impulse response were discarded and the next 512 samples were saved. Each HRTF response is thus 512 samples long.

The impulse response of the Optimus Pro 7 speaker was measured in the anechoic chamber using a Neumann KMi 84 microphone at a distance of 1.4 m. The measurement technique was exactly the same as the HRTF measurements. The speaker response was used to create an inverse filter, which can equalize the HRTF measurements to compensate for the nonuniform speaker response, particularly at low frequencies. However, this equalization will not compensate for other nondirectional elements of the measurements, including the ear canal resonance and the response difference between the Etymotic and Neumann microphones. To prevent excessive noise amplification, the gain of the inverse filter was limited to +20 dB at very low frequencies.

Figure 1 shows the average magnitude response of all the HRTF measurements made in the horizontal plane using the "small" pinna, and includes the speaker equalization described above. This planar diffuse field response is compared with that derived from Shaw's data, obtained from HRTF measurements of human subjects.[4] The two curves have the same general shape, exhibiting a peak at 2–3 kHz and a notch near 10 kHz, and are in close agreement for frequencies below 2 kHz.

The HRTF data are available on the Internet via anonymous FTP from the machine "sound.media.mit.edu" (Internet address 18.85.0.105) in the directory "pub/Data/KEMAR". The data are organized into binary archives accompanied by a document that describes in detail the format of the data. The data may also be retrieved via the World Wide Web page "http://sound.media.mit.edu/KEMAR.html". Any correspondence regarding the data may be sent to the authors via the Internet mail address "kemar@media.mit.edu".

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[1]D. D. Rife and J. Vanderkooy, "Transfer-Function Measurements using Maximum-Length Sequences," J. Audio Eng. Soc. 37, 419–444 (1989).

[2]J. Vanderkooy, "Aspects of MLS Measuring Systems," J. Audio Eng. Soc. 42, 219–231 (1994).

[3]C. L. Searle, L. D. Braida, D. R. Cuddy, and M. F. Davis, "Binaural pinna disparity: another auditory localization cue," J. Acoust. Soc. Am. 57, 448–455 (1975).

[4]E. A. G. Shaw and M. Vaillancourt, "Transformation of sound-pressure level from the free field to the eardrum presented in numerical form," J. Acoust. Soc. Am. 78, 1120–1123 (1985).

Advanced-degree dissertations in acoustics

Editor's note: Abstracts of Doctor and Master's theses will be welcomed at all times. Please note that they must be double spaced, limited to 200 words of text, must include the appropriates PACS classification numbers, and formatted as shown below (don't make the editor retype them, please!). The address for obtaining a copy of the thesis is helpful. Please submit two copies.

Acoustic intensity measurements in the presence of low Mach number flow [43.58.Fm, 43.28.Ra]—Toby D. McNeal, *Graduate Program in Acoustics, The Pennsylvania State University, University Park, PA 16802, August 1993 (M.S.).* Acoustic intensity provides wave propagation directionality along with absolute magnitude, and it can be measured in the near or far field of a source. Acoustic intensity measurements acquired in the presence of a mean flow are susceptible to errors due to the effects of the flow noise on the sensor. To determine if this error could be quantified, intensity measurements were acquired with the standard two-microphone cross spectral technique in a sound field that contained both mean flow and an independent random broadband noise source. The microphones were flush mounted at several different separation distances in the test section of a wind tunnel that provided the desired flow conditions, while a large speaker provided the independent random noise source. The error calculations were based on a technique that had already been derived theoretically and published, but had not been proven experimentally. The experiments performed validate that the error is indeed a bias error and that it can be quantified accurately. In addition, accurate quantification of the error is not limited to one-dimensional sound fields that contain only plane waves, so the method can be easily extended to two or three directions with complex wave propagation.

Advisor: Gerald C. Lauchle.

Lattice gas models for sound propagation simulation [43.20.Bi]—Yasushi Sudo, *Graduate Program in Acoustics, The Pennsylvania State University, University Park, PA 16802, May 1994 (Ph.D.).* New lattice gas models for sound propagation are studied in this thesis. The one dimensional (1-D) model has zero truncation error, and the group velocity is independent of wave number as is required from the continuum limit. Conventional finite difference approaches do not have these properties in general. Boundary condition treatments, applicable to the 1-D model, are also given. An extension of the 1-D model to two-dimensional (2-D) models is also discussed. A model for a 2-D square lattice and a model for a 2-D hexagonal lattice are obtained. In arbitrary directions of propagation the methods are at least second-order accurate. However, the accuracy of the 2-D square lattice model is greatly increased for narrow angle propagation along the coordinate axes. Dissipation effects can be included into these lattice gas wave models. To simulate dissipation effects, lattice gas particles are assumed to take a random walk. A different formulation of the above 1-D and 2-D models is also developed. This formulation requires less computer memory and also provides a way to include fluid dynamic nonlinear effects into the lattice gas wave models. Using this formulation, lattice gas models for Burgers' equation and the 1-D Euler equations are obtained.

Thesis advisor: Victor W. Sparrow.

A method for predicting noise reduction from a vibrating plate using modal analysis and a point source acoustic model [43.20.Tb, 43.40.At, 43.40.Dx, 43.40.Rj]—Daniel L. Battazzo, *Graduate Program in Acoustics, The Pennsylvania State University, University Park, PA 16802, December 1993 (M.S.).* Modal analysis is used to develop a model to evaluate the acoustic output power from various damped and undamped configurations of a vibrating plate. The residue values determined during the modal analysis yield a shape for the plate's vibration during a given mode. The shape is scaled by assuming a fixed input excitation force and determining the resulting velocity at a reference point on the plate using a frequency response function between the excitation point and the reference. The data taken during the modal analysis and the scaling factor are used as the input for an acoustic model based on a representation of the plate as a collection of point sources. The output acoustic power is determined for each configuration using the model and compared against the output power measured using the two-microphone intensity technique in an anechoic chamber. Comparable conclusions can be drawn about which damping configurations are most effective in treating noise emissions using either the model results or the intensity results. Insights into the limitations discovered regarding the application of selective damping are discussed, with the findings that selective damping is useful only in limited circumstances.

Thesis advisor: Oliver H. McDaniel.

Design and analysis of a cylindrical near-field array [43.30.Sf, 43.30.Yj, 43.38.Fx]—Philip A. Ferlino, *Graduate Program in Acoustics, The Pennsylvania State University, University Park, PA 16802, May 1994 (M.S.).* A near-field calibration array (NFCA), which can be used to calibrate large transducer arrays, has been designed and analyzed. The NFCA can either be of the transmitting or receiving type, which allows for calibration of hydrophones as well as projectors and acts as a plane-wave synthesizer in the transmit mode, or as a filter in the receive mode. The design process has been outlined and explained step by step according to guidelines and rules of thumb that were established by Trott and Van Buren. The performance of the NFCA design was analyzed for plane wave uniformity via FORTRAN and C routines over a prescribed frequency range and volume.

Advisor: W. Jack Hughes.

