Session 4aPLa

Plenary Lecture

David T. Blackstock, Chair
Applied Research Laboratories, University of Texas, P.O. Box 8029, Austin, Texas 78713-8029

Chair's Introduction—8:00

8:05

4aPLa1. Fuzzier but simpler analytic models for physical acoustics and structural acoustics. Allan D. Pierce (Dept. of Aero. and Mech. Eng., Boston Univ., 110 Cummington St., Boston, MA 02315, jones@bu.edu)

Acousticians have long been forced to analyze physical systems for which they apparently do not have enough information to make detailed analytical or computational predictions. One could say that their knowledge of such systems is ''fuzzy.'' Alternately, the means of measuring the details or of describing such a system may be in principle possible, but the answers one really desires, those which one can ''deal with,'' are of seemingly far less complexity than the modeled system in all its detail. The trick, of course, is to hypothesize a small number of simple and relevant descriptors of the system and to build a predictive model that uses only these descriptors. The rich and venerable literature of acoustics abounds with successful attempts in this vein, and a few such, taken from diverse subfields such as physical acoustics, architectural acoustics, underwater acoustics, and structural acoustics, are briefly reviewed in this talk. New problems and new questions lead to new searches for appropriate descriptors and for simple models which employ these descriptors. Theoretical tools which help to fill in the gaps caused by the incompleteness of these descriptors are general conservation principles and Jayne's theory of maximum-likelihood based on Shannon's uncertainty function.

THURSDAY MORNING, 25 JUNE 1998 CASCADE BALLROOM II (W), 7:45 TO 10:45 A.M.

Session 4aAA

Architectural Acoustics: Case Study on the New Tokyo Performing Arts Center

Leo L. Beranek, Chair
975 Memorial Drive, Suite 804, Cambridge, Massachusetts 02138-5755

Chair's Introduction—7:45

Invited Papers

7:50

4aAA1. Japan's new National Performing Arts Center. Takahiko Yanagisawa (TAK Associated Architects, Inc., 1-7 Kanda Nishiki-cho, Chiyoda-ku, Tokyo, 101 Japan)

Japan's first National Performing Arts Center, a ''theater city,'' was opened in the Shinjuku district of Tokyo in the fall of 1997. The center is composed of the New National Theatre, which was the winning design in an international architectural competition in 1986, and Tokyo Opera City and covers a total area of 11 acres. The New National Theatre includes an opera house, a drama theater, and an experimental theater, while Tokyo Opera City comprises a concert hall, a skyscraper, and shops. The center promises to become the focal point of theatrical and musical activity in Japan. The architectural goal was to create spaces which would fully engage the human senses, becoming a ''sanctuary of memorable experiences.'' Special focus was placed on the enhancement of the live sound produced by the human voice as well as instrumental sound. The acoustical consultant and associated laboratory were intimately involved from the beginning, and it was therefore possible to integrate their ideas into the design at an early stage. The team also conducted measurements of many of the world's finest halls. Each of the halls boasts its own distinct character, and they are acclaimed by musicians, audiences, and critics alike.

8:05

4aAA2. Acoustical design of the opera house of the New National Theater, Tokyo, Japan. Leo L. Beranek (975 Memorial Dr., Ste. 804, Cambridge, MA 02138), Takayuki Hidaka, and Sadahiro Masuda (TAK Associated Architects, Inc., Tokyo, 101 Japan)

The New National Theater (NNT) project containing three halls started with an open international design competition in 1986 and ended with a grand opening October 10, 1997. The architect is Takahiko Yanagisawa. The opera house, the core of the NNT facility, has a seating capacity of 1810, a volume of 14 500 m^3, and a reverberation time with full audience of 1.5 s (stage curtain open).

Various model experiments, using a CAD model, 1:10 scale model, and full-sized materials samples, were conducted over a 7-year period. As a result, the main floor has an almost rectangular shape, three-layered balconies have a modest fan shape, and the balcony fronts at each level create a rectangular shape. The overhang of each balcony is minimized to take visibility requirements into consideration. The unique design has a large curved reflector in front of the proscenium and over the orchestra pit and curved reflecting surfaces at the fronts of each of the side balconies to reflect the singer's voices uniformly from a large portion of the tremendous stage. The balcony fronts, the side walls, and the ceiling also augment the reflections from those surfaces. The unoccupied values are: EDT=1.6 s, $C_{80}(3)=2.4$ dB, $[I-IACC_{E3}]=0.66$m and bass ratio=1.1.

8:55

4aAA3. Acoustical design of the medium theater of the New National Theater, Tokyo, Japan. Leo L. Beranek (975 Memorial Dr., Ste. 804, Cambridge, MA 02138), Takayuki Hidaka, and Sadahiro Masuda (TAK Associated Architects, Inc., Tokyo, 101 Japan)

The medium theater is mainly intended for the performances of musicals, dramas, dances, and small-size operas. It features use of two types of stage arrangement: a proscenium type with modest fan plan (1038 seats with $V=7150$ m^3 and occupied RT=1.0 s), and an open stage type with wide fan plan (1010 seats with $V=9250$ m^3 and occupied RT=1.3 s). The stage types are changeable from one to the other by operation of a movable floor, which is the front half of the main floor, and of slidable side walls. To cope with a wide variety of performing types, two stage types, two scenery lofts, and extensive lighting and loudspeaker requirements, minimal effective initial reflecting surfaces to enhance raw voice were studied through experiments and studies with CAD models and 10:1 scale model tests. The rear wall is a combination of a Schroeder QRD diffuser and absorbing material to eliminate echo. Satisfactory sound strength without amplification is possible only because of the short depth of the hall.

9:10–9:20 Break

9:20

4aAA4. Acoustical design of the Tokyo Opera City (TOC) Concert Hall. Takayuki Hidaka (Takenaka R & D Inst., 1-5-1 Otsuka, Inzai, Chiba, 270-13 Japan), Leo L. Beranek (Cambridge, MA 02138), Sadahiro Masuda, Noriko Nishihara, and Toshiyuki Okano (Takenaka R & D Inst., Chiba, 270-13 Japan)

The TOC Concert Hall, called ''Takemitsu Memorial'' in honor of the late composer Toru Takemitsu, was opened September 10, 1997. With a seating capacity of 1636 and a volume of 15 300 m^3, the hall is sized to cover the musical range from recitals to orchestral concerts. The plan is rectangular in shape but, by request of the architect Takahiko Yanagisawa, the ceiling is a distorted pyramid, with its peak nearer the stage than the rear of the wall. This unique shape had to be analyzed using a CAD model and a 1:10 scale model so that all interior surfaces would be adjusted in shape and absorption to yield optimum values for RT, EDT, $IACC_{E3}$, surface diffusion, initial time delay gap, and loudness [Beranek, *Concert and Opera Halls* (ASA, New York, 1996)]. To provide a better ensemble condition for the musicians on stage and to provide early reflections to several other regions, a square canopy, almost 10 m on a side, is suspended above the stage. The pyramidal ceiling has diffusing elements added to simulate coffers. Schroeder QRDs on the ceiling surface facing the orchestra are used to control echo and to add sound diffusion. With audience, RT=1.95 s, G=6.4 dB, and $[1-IACC_{E3}]=0.72$.

10:10

4aAA5. Relation of acoustical parameters with and without audiences in concert halls and a simple method for simulating the occupied state. Takayuki Hidaka (Takenaka R & D Inst., 1-5-1 Otsuka, Inzai, Chiba, 270-13 Japan), Leo L. Beranek (Cambridge, MA 02138), and Noriko Nishihara (Takenaka R & D Inst., Chiba, 270-13 Japan)

Six acoustical parameters, RT, EDT, C_{80}, D, G, and $IACC_{E3}$, were measured with and without audiences at six concert and opera halls. The range in volume was 3576 to 15 300 m^3. The upholstering of the chairs ranged from light to heavy. Comparisons were made of these parameters before and after addition of the audience. The conclusions are: (1) Predictions of the occupied reverberation times were found inaccurate when the measured unoccupied values were adjusted by using unoccupied versus occupied data for the chairs obtained in a reverberation chamber. If diffuse sound conditions are assumed, it would be possible to predict EDT, C_{80}, D, and G from the occupied RT values whether measured or calculated as above. Such predictions are also inaccurate. (2) Plots of the six parameters, i.e., unoccupied versus occupied values, are nearly straight lines with correlations generally in excess of 0.9. These high correlations permit the use of empirical equations to predict these parameters for occupied halls from measurements made in unoccupied halls. (3) It was also found that $IACC_{E3}$ is almost the same whether measured with or without audience. This supports a previous report that $IACC_{E3}$, for unoccupied halls, is of importance in assessing their subjective quality when occupied [Hidaka *et al.*, J. Acoust. Soc. Am. 989–1007 (1995)]. This paper also reports on the successful use of a cloth covering over unoccupied seats to simulate occupied conditions.

10:30

4aAA6. Objective and subjective measurement of 15 opera houses in Europe and the USA. Takayuki Hidaka (Takenaka R & D Inst., 1-5-1 Otsuka, Inzai, Chiba, 270-13 Japan) and Leo L. Beranek (Cambridge, MA 02138)

Acoustical measurements were executed at 15 major opera houses in eight countries under unoccupied conditions. The objective parameters determined are RT, EDT, C_{80}, D, G, $IACC_{E3}$, and $IACC_A$, which are common attributes needed for concert hall evaluation. The measured unoccupied RT's range from 1.26 to 2.15s and occupied RTs, usually predicted using the empirical equations of the preceding presentation, range from 1.19 to 1.74 s. The range of the other six parameters are also examined. In particular, $[1\text{-}IACC_{E3}]$ has a range from 0.39 to 0.72, which is similar to that for concert halls [Hidaka *et al.*, J. Acoust. Soc. Am. 998–1007 (1995)]. Questionnaires have been returned from about 20 opera conductors which enable the authors to present subjective evaluations of the halls. Finally, the subjective evaluations are compared with the objective data.

4a THU. AM

Session 4aAB

Animal Bioacoustics: Temporal Patterns and Rhythm

Christopher W. Clark, Chair
Section of Neurobiology and Behavior, Bioacoustic Research Program, Cornell University, Ithaca, New York 14850

Invited Papers

8:30

4aAB1. It's all a matter of timing: Temporal mechanisms for production and encoding of acoustic signals. Andrew H. Bass and Deana A. Bodnar (Neurobiology and Behavior, Cornell Univ., Mudd Hall, Ithaca, NY 14853, ahb3@cornell.edu)

Studies of acoustic communication in teleost fish have identified basic principles underlying both the motor control of acoustic signaling and the sensory encoding of the physical attributes of those signals. Parental male midshipman fish excavate denlike nests under rocky shelters in the intertidal and subtidal zones along the western coast of the United States and Canada. Nesting males generate long duration, multi-harmonic signals with a sinusoidallike appearance that are known as ''hums.'' Observations of courtship behavior together with phonotaxis experiments support the hypothesis that the hum functions as a mate call. Early studies showed that a rhythmically active, pacemaker-motoneuron circuit in the hindbrain establishes the temporal features of hums [Bass and Baker, J. Neurobiol. **21**, 1155 (1990)]. Recent studies suggest that temporal coding in the auditory midbrain contributes to the segregation of, and discrimination between, concurrent acoustic signals which form acoustic beats like those generated by the overlapping hums of nesting male midshipman during the breeding season [Bodnar and Bass, J. Neurosci. **17**, 7553 (1997)]. Hence, the rhythmic timing of action potentials is important for both acoustic signaling and reception. [Work supported by NIH and NSF.]

8:55

4aAB2. Temporal rhythms in the signals of insects. T. G. Forrest (Dept. of Biol., Univ. of North Carolina at Asheville, Asheville, NC 28804, tforrest@unca.edu)

The signals generated by acoustic insects exhibit temporal patterns on several time scales. Annual or seasonal rhythms provide information about temporal changes in insect population size. Knowledge of such changes might be useful in ecological monitoring programs. Circadian or daily rhythms commonly seen in insect signaling are generated by endogenous ''clocks'' that are reset by environmental cues. Understanding these daily patterns is also important for monitoring insect populations. Within insect populations, the signals of neighbors sometimes show temporal patterns of synchrony or alternation with timing on the order of seconds. The phase relationships among individuals are maintained by acoustic resetting, whereby an individual's temporal rhythm will change depending on the phase that neighboring signals are received. On still smaller time scales, temporal patterns of insect songs are diverse and generally differ among species. These small-scale patterns are often essential components of mating signals and are used in mate recognition and mate choice.

9:20

4aAB3. Stochastic resonance in the amphibian auditory system? It's just a matter of time. P. M. Narins (Dept. of Physiological Sci., UCLA, 405 Hilgard Ave., Los Angeles, CA 90095)

Noise has traditionally been a factor to minimize or eliminate for optimum performance of a communication system. Recently, it has been proposed that in some cases, noise may *benefit* information transfer and that its existence in sensory systems may be an adaptation to *enhance* detection of weak signals. One possible mechanism that could take advantage of noise to enhance signal transmission is a physical phenomenon known as *stochastic resonance* (SR). In order to study such effects in auditory neurons, use was made of the ability of amphibians to function over a range of ambient temperatures, and thus over a range of internal noise levels. The goal of the present study was to determine the effect of temperature (and internal noise) on information transmission in the frog. To this end, core temperature shifts were induced experimentally and the resulting changes in signal-to-noise ratio (S/N) were quantified. Although our results do not demonstrate SR in the sense that the S/N passes through a maximum at a particular internal noise intensity, we do demonstrate the profound influence of the internal noise on the S/N derived from the neural spike trains. Moreover, recent field and behavioral data will be presented and interpreted in the framework of stochastic resonance. [Work supported by NIH Grant DC-00222 to PMN.]

9:45–10:00 Break

10:00

4aAB4. Whale voices from the deep: Temporal patterns and signal structures as adaptations for living in an acoustic medium. Christopher W. Clark (Section of Neurobiology and Behavior, Bioacoustics Res. Prog., Cornell Univ., Ithaca, NY 14850, cwc2@cornell.edu)

Whales produce long, rhythmic patterns of sounds and some sounds travel many hundreds of miles underwater. Species can be distinguished by temporal features, but there has been a history of infatuation with melodic qualities as the primary features of measurement. Temporal rates and time-bandwidth products are generally related to bathymetry and transmission properties, suggesting that signal features are adapted for communication and navigation. Pelagic species such as blue and fin whales rely on signals in the 10–30-Hz band, presumably to take advantage of the excellent low-frequency propagation properties of the deep ocean, with 10–200-s patterns of sound delivery. Examples will be presented illustrating the remarkable cadence of signal delivery and that animals retain a rhythm after minutes of silence. Shallow water species produce mid-frequency signals (50–1000 Hz) with temporal patterns on the order of seconds. These species with faster rhythms have greater signal variability covering a greater frequency range. At what level is this relationship between temporal pattern and spectral bandwidth indicative of adaptation to optimize for communication and navigation? Is temporal pattern a retained, conservative feature and spectral variability more of an embellishment by individuals to adjust to local conditions?

Contributed Paper

10:25

4aAB5. Time pattern of Sperm whale codas recorded in the Mediterranean Sea 1985–1996. G. Pavan (Centro Interdisciplinare di Bioacustica e Ricerche Ambientali, Università degli Studi di Pavia, Via Taramelli 24, 27100 Pavia, Italy), T. Hayward (Naval Res. Lab., Washington, DC 20375), J. F. Borsani, M. Priano, M. Manghi, and C. Fossati (Università degli Studi di Pavia, Via Taramelli 24, 27100 Pavia, Italy)

Sperm whales, *Physeter macrocephalus*(*catodon*), emit short click sequences, called codas, with regular time patterns [Watkins and Schevill, J. Acoust. Soc. Am. **62**, 1485 (1977)]. Since codas recorded in different geographical areas have different and stable patterns, they possibly serve to convey regional information. More than 120 codas were recorded in the Central Mediterranean Sea in the years 1985–1996 by several research groups using a number of different detection instruments, including stationary and towed hydrophones, military sonobuoys and passive sonars. All of the recorded codas share the same time pattern 3+1 (///-/) with an overall duration ranging from 456 to 1280 ms and an average value of 910 ms. Even if the coda duration varies, the click pattern remains significantly stable. In the present work, the repetition rate and time pattern of Mediterranean codas are characterized, also taking into consideration their possible biological and geographical role. [Work supported by the Italian Ministry of the Environment, Inspectorate for Sea Protection, and by the Italian Navy. Work of the second author supported by ONR Base funding at NRL.]

THURSDAY MORNING, 25 JUNE 1998 WEST BALLROOM B (S), 9:15 TO 11:55 A.M.

Session 4aAO

Acoustical Oceanography and Animal Bioacoustics: Acoustics of Fisheries and Plankton II

D. Vance Holliday, Chair
Tracor, Inc., 4664 Murphy Canyon Road, #102, San Diego, California 92123-4333

Chair's Introduction—9:15

Invited Paper

9:20

4aAO1. Acoustic scattering models of zooplankton from several major anatomical groups—theory and experiment. Timothy K. Stanton, Dezhang Chu (Dept. of Appl. Ocean Phys. and Eng., Woods Hole Oceanogr. Inst., Woods Hole, MA 02543, tstanton@whoi.edu), and Peter H. Wiebe (Woods Hole Oceanogr. Inst., Woods Hole, MA 02543)

In order to estimate zooplankton abundance with active acoustics methods, it is crucial to understand the acoustic scattering properties of the zooplankton. The major challenges in formulating scattering models include the facts that there are no exact solutions for these complex bounded bodies and that the boundary conditions are difficult to determine due to animal size and heterogeneities. A series of laboratory scattering measurements and associated model development has been conducted involving zooplankton from three major anatomical groups: (1) fluidlike (euphausiids), (2) elastic shelled (marine snails, both benthic and pelagic), and (3) gas-bearing (siphonophores). The measurements include backscatter versus acoustic frequency (part or all of the range 24 kHz to 1 MHz) and angle of orientation (up to one-degree precision). Dominant acoustic scattering mechanisms are identified through both

spectral and time-domain (pulse compression) analyses and scattering models are developed accordingly. Data and modeling results from the past several years are reviewed and the following new results are presented: (1) a new formulation for predicting scattering at end-on incidence by euphausiids and (2) observations and modeling of a strong peak in scattering levels of the shelled bodies in the upper Rayleigh scattering region. [Work supported by ONR and NSF.]

Contributed Papers

9:40

4aAO2. Covariance Mean Variance Classification (CMVC) techniques: Application to the acoustic classification of zooplankton.
Linda V. Martin Traykovski (Dept. of Biol., Woods Hole Oceanogr. Inst., Woods Hole, MA 02543, lmartin@whoi.edu), Timothy K. Stanton, Peter H. Wiebe, and James F. Lynch (Woods Hole Oceanogr. Inst., Woods Hole, MA 02543)

Accurate acoustic classification of zooplankton species has the potential to significantly improve estimates of zooplankton biomass made from ocean acoustic backscatter measurements. Theoretical models have been developed for three zooplankton scattering classes (hard elastic shelled, e.g., pteropods; fluidlike, e.g., euphausiids; gas bearing, e.g., siphonophores), providing a sound basis for a model-based classification approach. The Covariance Mean Variance Classification (CMVC) techniques classify broadband echoes from individual zooplankton based on comparisons of observed echo spectra to model space realizations. Three different CMVC algorithms were developed: the Integrated Score Classifier, the Pairwise Score Classifier, and the Bayesian Probability Classifier; these classifiers assign observations to a class based on similarities in covariance, mean, and variance, while accounting for model space ambiguity and validity. The CMVC techniques were applied to several hundred broadband (~350–750 kHz) echoes acquired from 24 different zooplankton to invert for scatterer class. All three classification algorithms had a high rate of success with high-quality, low SNR data. The CMVC approach was also applied to several thousand echoes from fluidlike zooplankton to invert for angle of orientation using both theoretical and empirical model spaces; excellent success rates were achieved with the empirical model spaces.

9:55

4aAO3. A DWBA-based representation of the extinction cross section of weakly scattering objects: Application to zooplankton.
Dezhang Chu (Dept. of Appl. Ocean Phys. and Eng., Woods Hole Oceanogr. Inst., Woods Hole, MA 02543) and Zhen Ye (Natl. Central Univ., Chung-li, Taiwan, ROC)

Because of the weakly scattering nature of the marine organisms, the Distorted Wave Born Approximation (DWBA) has been successfully used in modeling the acoustic backscattering by zooplankton [Stanton *et al.*, J. Acoust. Soc. Am. **94**, 3463–3472 (1993); Wiebe *et al.*, IEEE J. Ocean. Eng. (in press)]. However, in the cases where the scattering-induced attenuation is noticeable, the previously used DWBA-based solutions fail to predict the attenuation since they ignore the imaginary part of the scattering form function and result in a zero extinction cross section. In this paper, a revised DWBA-based approximation is presented to include the imaginary part of the scattering form function. In deriving the revised solution, phase compensation based on ray paths is used. The solutions are compared with the exact solutions, and the agreements between the approximate solution and the exact ones are reasonably good. The results from this study can be applied to bioacoustic applications where the attenuations due to scattering or/and multiple scattering by zooplankton are significant.

10:10

4aAO4. An *in situ* target strength model for Atlantic redfish.
Stephane Gauthier and George A. Rose (Fisheries Conservation Chair, Memorial Univ., P.O. Box 4920, St. John's, NF A1C 5R3, Canada, sgauthie@caribou.ifmt.nf.ca)

An *in situ* target strength (TS) model for Atlantic redfish is proposed based on a series of acoustic-trawl experiments conducted at the outer edge of the Green and Grand Banks in Newfoundland waters (NAFO div. 3Ps) in July 1996 and January 1997. Acoustic data were collected using a SIMRAD EK500 echosounder with a hull-mounted 38-kHz split beam transducer and a deep-tow system with a 38-kHz dual beam transducer that can be towed down to 400-m depth. Bottom and midwater trawling sets were performed to account for fish species composition and length distributions. Study sites were selected on the basis of low variance in individual fish size and the absence of other fish species. The model tests the dependence of TS on diel migratory behavior, TS methodology (split and dual beam), distance from transducer, fish density and size, and the presence and relative abundance of zooplanktonic organisms (krill).

10:25

4aAO5. Sensitivity of acoustic scattering models to fish morphometry.
J. Michael Jech and John K. Horne (Cooperative Inst. for Limnology and Ecosystems Res., 2205 Commonwealth Blvd., Ann Arbor, MI 48105, jech@glerl.noaa.gov)

Current efforst to model fish backscatter (e.g., Kirchoff ray-mode model [C. S. Clay and J. K. Horne, J. Acoust. Soc. Am. **96**, 1661–1668 (1994)] use digitized images of fish anatomy for a more realistic representation of swimbladder shape, volume, and aspect. However, the degree of image resolution necessary for acoustic characterization and improved predictions, and the variability associated with resolution has not been quantified. Does a more accurate and higher resolution image result in improved correlation between theory and empirical measurements? Backscatter amplitudes display strong relationships with carrier frequency, swimbladder shape, and swimbladder aspect at rather high frequencies. Are predictions of these relationships influenced by image resolution? X-ray images of Atlantic cod (*Gadus morhua*) and brook trout (*Salvelinus fontinalis*) were digitized at fine resolution, and sensitivity analyses were performed to quantify the effect of varying image resolution on predicted backscatter as a function of carrier frequency and swimbladder shape and aspect. Quantifying image resolution is a step towards determining morphometric effects on backscatter variability and estimating fish abundance using multi-frequency data and the inverse approach. [Work supported by ONR.]

10:40–10:55 Break

10:55

4aAO6. Quantifying intraspecies variation in acoustic backscatter models.
John K. Horne and J. Michael Jech (Cooperative Inst. for Limnology and Ecosystems Res., 2205 Commonwealth Ave., Ann Arbor, MI 48105, horne@glerl.noaa.gov)

Fisheries researches are increasing the use of frequency-, length-, and aspect-dependent backscatter models in population abundance estimates of fish and zooplankton. Model predictions are often based on measurements from a single or a limited number of organisms. The resulting model parameters are used to predict backscatter for organisms of any length in the population. At rather high frequencies (i.e., length/wavelength ratios of 1 to 20), choice of carrier frequency, swimbladder shape, and swimbladder aspect influence amplitudes of returned echoes. Morphological differences among individuals will therefore determine the variance observed in amplitudes of echo ensembles. To quantify variance in predicted backscatter within species as a function of carrier frequency, fish length, and swimbladder aspect, we combined data from digitized x rays with Kirchhoff ray-mode backscattering models of brook trout (*Salvelinus fontinalis*) and Atlantic code (*Gadus morhua*). Quantifying backscatter variances within and between species allow us to examine the applicability of multi-frequency data and the inverse approach to estimate fish length abundances. [Work supported by ONR.]

11:10

4aAO7. Estimation of the target strength of juvenile salmonids at any aspect. Thomas J. Carlson (USCE—Waterways Experiment Station, CENPP-PE-E, P.O. Box 2946, Portland, OR 97206-2946)

Many tasks, such as assessment of the state of stocks and observation of migrant behavior, elemental for recovery of Columbia River basin salmonid stocks, are currently performed using active acoustic methods. Detectability models are required to configure acquisition systems, design elements of studies, and process and analyze data. Accurate and precise estimates of juvenile salmonid target strength are necessary input to detectability models. The target strength at 420 kHz of 87 juvenile salmonids was measured within the length range of 45 to 300 mm. The fish were rotated through pitch, roll, and yaw planes. Forty target strength measurements were obtained for each degree of rotation in each plane for each fish. Mean target strengths were computed for 1° and 5° increments of rotation. The mean target strength estimates for each 5° increment of rotation for all fish were fit to the equation $\sigma/\lambda^2 = a(L/\lambda)^b$. Estimates of mean target strength for regions of the sphere encompassing the fish between the planes of rotation were estimated by interpolation to complete a description of the target strength of juvenile salmon at any aspect.

11:25

4aAO8. Target strength measurements of walleye pollock (*Theragra chalcogramma*). Jimmie J. Traynor (Natl. Marine Fisheries Service, NOAA, Seattle, WA, jimt@afsc.noaa.gov)

The results of recent target strength measurements of walleye pollock from the North Pacific are presented. The measurements, made with a lowered transducer system are compared with historical results. The system allows the transducer to be moved closer to the target fish thus reducing the well-known bias of *in situ* target strength measurements due to range-dependent noise thresholds. Results using a conventional system with the transducer mounted on the research vessel and this system are discussed. The AFSC currently uses a target strength-to-length relationship [$TS = 20 \log(L) - 66.0$, L, length in centimeters] to scale echo integration information to estimates of fish density. The appropriateness of this practice is discussed in light of the recent target strength measurements. Caveats regarding the limitations of *in situ* target strength measurement techniques are presented and suggestions for appropriate conditions for such measurements are provided.

11:40

4aAO9. Three-dimensional acoustic measurements of zooplankton swimming behavior in the Red Sea. Duncan E. McGehee (Tracor Appl. Sci., 4669 Murphy Canyon Rd., San Diego, CA 92123, dmcgehee@galileo.tracor.com), Amatzia Genin (H. Steinitz Marine Biol. Lab., 88103 Eilat, Israel), and Jules S. Jaffe (Scripps Inst. of Oceanogr., La Jolla, CA 92093-0238)

Three-dimensional swimming trajectories of several hundred thousand individual zooplankters were measured using the 445-kHz acoustical imaging system FishTV during a three night moored deployment in the Gulf of Eilat, Israel. The sonar examined a 5-m^3 volume at 27 m depth, in water 300-m deep. The image rate was 4 images/s. Targets were tracked from image to image using an automatic three-dimensional tracking algorithm. Over 14 000 targets remained in view for over 5 s, and their trajectories were used in subsequent analysis. Data from net tows indicated that most targets were euphausiids. The horizontal speeds of targets -75 dB and below were highly correlated with flow measurements from an S4 current meter. These targets also exhibited strong vertical motions, apparently due to internal waves. Estimated mean flow was subtracted from each trajectory to compute the swimming speeds of the animals themselves. These were generally much lower than the mean flow. During the night, variance in the vertical component of the flow-removed tracks was much greater than variance in the horizontal. However, the variance became more isotropic as dawn approached. A hop-and-sink foraging behavior offers one possible explanation for this.

THURSDAY MORNING, 25 JUNE 1998 EAST BALLROOM B (S), 9:15 A.M. TO 12:00 NOON

Session 4aBVa

Bioresponse to Vibration/Biomedical Ultrasound: Applications of Microbubble Based Echo Contrast Agents I

Inder Raj S. Makin, Cochair
Ethicon Endo-Surgery, 4545 Creek Road, Cincinnati, Ohio 45242

Nico De Jong, Cochair
Erasmus University, Room Ee2302, 3000 DR Rotterdam, The Netherlands

Invited Papers

9:15

4aBVa1. Specific characteristics of ultrasound contrast agents. Nico De Jong and Peter Frinking (Erasmus Univ., Rm. Ee 2302, P.O. Box 1738, 3000 DR Rotterdam, The Netherlands)

Ultrasound contrast agents possess acoustical characteristics which differ from the surrounding medium. Specific signatures of the agent, consisting of free or encapsulated gas bubbles, will improve the discrimination between the blood containing the agent and the tissue. One of the specific signatures is the initiation of a volume pulsation when the ultrasound wave hits a free or encapsulated gas bubble. Depending on the magnitude of the ultrasound wave the pulsation will be linear to the applied pressure or nonlinear to the applied pressure. Nonlinear vibration can be split into stationary (harmonic) and transient (power) scattering. Besides the nonlinear vibration (harmonics), specific characteristics include enhanced scattering as a function of the acoustic amplitude, transient scattering which is very useful for measuring flow, LOC (loss of correlation) imaging, and the persistence as a function of ambient parameters like pressure and temperature. Recently, a 2-D echomachine has been adapted for imaging of the nonlinear vibration of contrast agents and further developments in both hardware and software will be advantageous for the many specific characteristics of these agents.

9:35

4aBVa2. Nonlinear properties of microbubbles and applications to medical ultrasound imaging. Volkmar Uhlendorf, Thomas Fritzsch, Michael Reinhardt, and Frank-Detlef Scholle (Res. Labs. of Schering AG, 13342 Berlin, Germany, volkmar.uhlendorf@schering.de)

Gas bubbles smaller than 10 μm dissolve within milliseconds even in gas-saturated liquids, but coatings, etc. can prevent dissolution. These stabilized microbubbles serve as transpulmonary contrast agents. When only their linear properties are employed, physical laws limit the diagnostic potential to detection in large diameter vessels. Nonlinear acoustic properties of contrast agents permit very important additional applications. Nonlinearity arises from bubble pulsations, shell properties, finite amplitude waves, and electronic hardware. The first two sources can dominate, allowing new diagnostic imaging modes sufficiently sensitive to detect isolated microbubbles *in vivo*. Harmonic imaging modes detect 2nd harmonics of the transmit frequency, mainly from microbubbles. Acoustic Emission modes destroy bubble shells by one pulse of moderate amplitude. Consequently, the free bubbles respond strongly to this and other pulses before dissolving. Observed lifetimes of 1–20 ms are enough for harmonic or conventional Doppler detection. Typically, bubbles are replenished too slowly to get similar signal intensity for more than one frame at normal rates. Optimal pulse timing should allow new microbubbles to move close to the beam axis before the next pulse arrives. Applications may be counting methods, flow tagging, and quantification of tissue perfusion, possibly combined with single-shot three-dimensional acquisition and HI modes.

9:55

4aBVa3. Harmonic imaging for microbubble contrast agent detection. P. N. Burns (Imaging Res., Sunnybrook Health Sci. Ctr. S660, 2075 Bayview Ave., Toronto, ON M4N 3M5, Canada, burns@srcl.sunnybrook.utoronto.ca), C. T. Chin, D. Hope Simpson (Medical Biophys., Univ. of Toronto), and J. Powers (ATL Ultrasound, Bothell, WA 98041-3003)

The detection of microbubble contrast agents in the vessels of the microcirculation is currently limited by the relatively low concentrations attainable without causing acoustic attenuation, set against the high clutter signal imposed by surrounding soft tissue. By inducing nonlinear resonant oscillation in a population of microbubbles in transit in the circulation, echoes at the second harmonic can be detected preferentially, thus segmenting the blood signal from that due to tissue. Real-time harmonic imaging and Doppler have been implemented using a clinical array imaging system, and it is shown that with bubble agents it can detect flow in 40 μm vessels of the kidney. At higher incident peak pressures (above about 0.5 Mpa at about 3 MHz), some microbubble contrast agents are irreversibly disrupted by ultrasound. They then produce transient echoes which are high in amplitude and rich in harmonics. Furthermore, transient echoes can be obtained repeatedly over a period of milliseconds, allowing correlation imaging. Such ''transient Doppler'' imaging can demonstrate microcirculatory blood in the capillaries of the moving myocardium, a new development with significant clinical potential. [Work supported by Medical Research Council of Canada.]

10:15

4aBVa4. Acoustic characterization of contrast agents for medical ultrasound imaging. Lars Hoff (Dept. of Telecommunications, The Norwegian Univ. of Sci. and Technol., N-7034 Trondheim, Norway, hoff@tele.ntnu.no) and Per C. Sontum (Nycomed Imaging AS, N-0401 Oslo, Norway)

Nycomed's ultrasound contrast agent *NC100100* has been investigated by *in vitro* acoustic measurements. Acoustic attenuation spectra were used to determine resonance frequencies of the particles. The spectra were correlated with size distributions, and it was found that the shell-encapsulated gas bubbles can be described as viscoelastic particles with bulk modulus 700 kPa. When exposed to hydrostatic overpressures mimicking those found *in vivo* during the systolic heart cycle, the resonance frequency increased, as expected by the particles' increased stiffness. This effect was reversible: After the pressure was released, the particles went back to giving the original attenuation spectrum. This shows that the particles are not destroyed or otherwise changed by the pressure. Acoustic backscatter measured as a function of distance through a contrast agent was used to estimate the backscatter efficiency of the particles, that is, the ratio between scattered and absorbed ultrasound. Results from these measurements agree with theoretical estimates based on the attenuation spectra. Measurements on *NC100100* were compared with earlier results from measurements on *Albunex*® and measurements on an experimental polymer-encapsulated contrast agent, showing how different shell materials cause differences in particle stability and stiffness.

10:35–10:45 Break

Contributed Papers

10:45

4aBVa5. Nonlinear response of microbubbles to pulsed diagnostic ultrasound. Sascha Hilgenfeldt, Detlef Lohse (Fachbereich Physik, Univ. of Marburg, Renthof 6, D-35032 Marburg, Germany), and Michael Zomack (Schering AG, Berlin, Germany)

The sound radiation from micrometer-size bubbles driven by short ultrasound pulses emitted by diagnostic ultrasonography devices is investigated. The frequency spectrum is analyzed in order to predict useful parameter ranges for the application of the second harmonic method.

11:00

4aBVa6. Acoustical nonlinearity parameter of liquids with microbubbles. X. F. Gong, S. G. Ye, D. Zhang (Inst. Acoust., State Key Lab. Modern Acoust., Nanjing Univ., Nanjing, 210093, PROC), S. S. Feng, R. Q. Zhang, R. T. Wang (Acad. Sinica, PROC), Z. Z. Xu, L. M. Liu (Shanghai Zhong Shan Hospital, China), and K. L. Ha (Pukyoung Natl. Univ., Pusan, Korea)

The gas bubbles in liquid are the strong scatters of the sound propagation in such liquid. A liquid containing microbubbles is used as an effective ultrasound contrast agent in medical diagnosis to improve the contrast of the ultrasonic image. However, the existence of bubbles in liquid may enhance its nonlinearity parameter. In our previous paper, the preliminary results of some ultrasound contrast agents were reported and their large values of the nonlinearity parameter were obtained. This paper is devoted to an experimental demonstration of the influence of microbubbles on the nonlinearity parameter B/A values. These include measuring the dependence of B/A values of Echovist 300 on time. Results show that B/A values decrease with time (from 2678 to 16 over 45 min). B/A values of human blood with a different portion of Echovist were also studied. Larger B/A values are obtained with more Echovist. B/A values of a kind aqua, such as 76% Injectio Meglumini Diatrizoatis Composita (MDC), were studied before and after sonication with different bubble contents. Results indicate that the values of nonlinearity parameter depend on the presence of microbubbles obviously. Some explanation of this effect is discussed. [Work supported by NSF of China.]

11:15

4aBVa7. Bubble dynamics of ultrasound contrast agents. Michalakis A. Averkiou, Matthew F. Bruce, and Jeffry E. Powers (ATL Ultrasound, P.O. Box 30003, Bothell, WA 98041)

The bubble dynamics involved with microbubble contrast agents under insonification is investigated. The acoustic field of an ATL HDI-3000 diagnostic ultrasound system in a contrast specific harmonic imaging mode is reviewed first, and its major features that are related with microbubble behavior are discussed. Issues relating to sound attenuation, mechanical index, and bubble destruction are addressed. The nonlinear oscillatory behavior of contrast microbubbles is modeled with the Gilmore equation. The acoustic pressure field of a short pulse utilized in harmonic imaging is measured with a hydrophone and used as the driving pressure of the Gilmore model. Radius-time $[R(t)]$ and bubble wall velocity-time $[U(t)]$ curves are shown. Frequency domain analysis of $U(t)$ indicates transient resonance characteristics in both the fundamental and second harmonic components that are somewhat different from what one would expect with a continuous-wave steady-state response. The times for complete solution of microbubbles in water are calculated and correlated to observations seen in ultrasound images with contrast agents. Radio frequency (rf) data of scattered pulses from contrast agent microbubbles in an *in-vitro* experiment were collected with a phased array. This data is used to support and explain the contrast microbubble behavior.

11:30

4aBVa8. Estimation of contrast agent concentration using spectrum analysis. Cheri X. Deng (Riverside Res. Inst., 330 W 42nd St., New York, NY 10036, cheri@rrinyc.org), Frederic L. Lizzi (Riverside Res. Inst., New York, NY 10036), Ronald H. Silverman, and D. Jackson Coleman (Cornell Univ. Medical College, New York, NY 10021)

A theoretical scattering model was formulated to calculate the calibrated power spectrum of ultrasound contrast agents. The analysis incorporates the scattering coefficient from an encapsulated contrast agent bubble and includes the effects of realistic focused beams. The results relate specific spectral features to the characteristics of contrast agents, including their concentration and size distribution. Analytical calculations and experimental studies were conducted to analyze the calibrated power spectra of backscatter data using Albunex® and Aerosomes™ over frequency bands employed in typical medical ultrasound (3–12 MHz) and very-high-frequency ultrasound (VHFU, 15–50 MHz). The results showed that contrast agent concentration can be estimated from the spectral features such as the spectral slope and intercept values, which are measured using linear regression analysis. These studies also demonstrated the feasibility of employing contrast agent with VHFU, using wide-band transducers with center frequencies near 40 MHz. The VHFU scans of *in vivo* rabbit eyes showed the time history of contrast agent activity and revealed increased backscatter (as much as 8 dB) within subsegments of the ciliary body. Estimation of contrast agent concentration as a function of time promises to provide key information about perfusion in a contrast agent perfused organ.

11:45

4aBVa9. Detection and estimation of microbubble size distribution in blood. Klaus V. Jenderka, Georg Dietrich, Ulrich Cobet (Inst. of Medical Phys. and Biophys., Medical Faculty, Martin Luther Univ., D-06097 Halle, Germany), and Petr Urbanek (HP-medica, Bahnhofstrasse 30, D-86150 Augsburg, Germany)

The characterization of the microbubble size distribution in blood is of great practical interest for the design and the use of heart–lung machines and for clinical research. Another application is the continuous estimation of size distribution of ultrasound contrast agents. The measurement system is based on a 2-MHz pulse Doppler device and a special transducer, designed to connect to a 3/8-in. tube. The computer controlled system consists of two simultaneously working channels. The system is self-calibrating. The amplitude of a reference echo from a reflector behind the tube is used for the compensation of different acoustical properties of the tube. The minimal size of detectable gas bubbles and also the resolution of the size distribution is 1 μm in diameter. The detected size distribution will be measured continuously and saved every 5 s. Therefore it is possible to study the time course of the number and the size of gas bubbles, for example, during a heart operation. Showing the effectiveness of an arterial blood filter is done by measurements before and after the filter. Examples of the size distribution of contrast agents in different suspensions versus time after injection will be presented.

Session 4aBVb

Bioresponse to Vibration/Biomedical Ultrasound: Medical UItrasound I—Transduction and Propagation

Junru Wu, Chair

Department of Physics, Box 2-11200, University of Vermont, Burlington, Vermont 05405

Contributed Papers

9:15

4aBVb1. Ultrasonic phased array design for reduced crosstalk. John Dodson and Karl Grosh (Dept. of Mech. Eng. and Appl. Mech., Univ. of Michigan, Ann Arbor, MI 48109-2125)

In recent years, therapeutic ultrasound has received increased attention as a treatment modality for cancer therapy (hyperthermia) and heart arythmia (ablation surgery). Ultrasonic phased arrays offer the benefit of noninvasive treatment with the flexibility of variable focus and beam steering. In designing and building these arrays, electromechanical isolation is extremely important as interelement crosstalk will degrade array performance. However, manufacturability and cost, constraints which may conflict with the goal of reduced crosstalk, must also be a consideration. Efficient numerical methods present a useful design tool for array design and allow the investigation of structural acoustic phenomena in general. In this paper, a two-dimensional finite-element model for an ultrasonic phased array in contact with the human body is used to study array design. The effects of material selection and geometry for matching layer(s) and interelement structural matrix are examined for their impact on radiation pattern, power input to the fluid domain, and deleterious effect on the electromechanical behavior of the transducers. Finally, design guidelines are formulated and, if possible, compared to experimental results.

9:30

4aBVb2. A new method of ultrasonic hydrophone calibration using KZK wave modeling. Hendrik J. Bleeker (ATL Ultrasound, 22100 Bothell Everett Hwy., P.O. Box 3003, Bothell, WA 98041) and Peter A. Lewin (Drexel Univ., Philadelphia, PA 19104)

Most commercially available ultrasonic transducers exhibit finite amplitude distortion in water during hydrophone measurements needed to comply with regulatory requirements. The frequencies observed due to finite amplitude distortion can easily exceed ten times the transducer center frequency and 100 MHz. Typically, hydrophone calibrations are supplied only up to 15 or 20 MHz and do not exhibit a flat response. The frequency response above 15 MHz should be known to accurately represent the acoustic information, especially for high-frequency transducers ranging between 7.5 and 15 MHz. A new hydrophone calibration technique has successfully predicted the frequency response of hydrophones up to 100 MHz. A circular source transducer was first characterized and then modeled using the KZK wave propagation model. This model accounts for diffraction, absorption, and nonlinearity. The transducer frequency response was measured with a hydrophone and compared to the simulation. This difference characterized the frequency response of the hydrophone and was used to estimate the hydrophone calibration. The estimated calibration at 20 MHz was checked and provided good agreement with the manufacturer calibration supplied. Acoustic measurement accuracy will be improved if the hydrophone frequency response is deconvolved from the actual acoustic transducer response.

9:45

4aBVb3. High-frequency backscatter measurements of bovine tissues with unfocused and focused transducers. Subha Maruvada and Kirk K. Shung (Grad. Prog. in Acoust. and Bioengineering Prog., Penn State Univ., State College, PA 16801, sxm104@psu.edu)

In order to improve resolution of ultrasonic imaging, high-frequency scanners which are employed in the range 20–50 MHz are needed. As a result, it is critical to obtain data on ultrasonic scattering and attenuation in this frequency range. At these high frequencies, it is not feasible to make scattering measurements with unfocused transducers, therefore focused transducers are needed. Using the standard substitution method to calculate the backscatter coefficient, as is used with unfocused transducers, yields erroneous results for focused transducers. The assumption that the reflected echo from a perfect reflector in the far field can be calculated as though the transducer acts like a point source does not hold for focused transducers. Therefore a method is presented for focused transducers where the flat reflector is substituted by a particulate reference medium whose backscatter coefficient is well known and documented. Unfocused transducers in the range 10–20 MHz will be used in comparison with focused transducers to measure backscatter. For 15–35 MHz, focused transducers will be used. Results for the backscatter coefficient will be presented for various bovine tissues.

10:00

4aBVb4. The acoustic properties of microcalcifications in the context of breast ultrasound. Martin E. Anderson (Dept. of Biomed. Eng., Duke Univ., Durham, NC 27708), Mary S. C. Soo (Duke Univ. Medical Ctr., Durham, NC 27708), and Gregg E. Trahey (Duke Univ., Durham, NC 27708)

Breast microcalcifications are small crystals of calcium phosphates which are important diagnostic indicators under mammography. Despite continuing advances in the resolution and sensitivity of modern scanners, the role of ultrasound in the breast clinic is limited by the inability of current ultrasound practice to visualize microcalcifications reliably. While mammography is currently the gold standard for their detection, approximately 70% of breast biopsies prompted by the presentation of microcalcifications result in negative biopsy. The improved visualization of microcalcifications under ultrasound could provide additional diagnostic information in these cases, potentially reducing the number of negative biopsies. A review is presented of the theoretical and empirical results to date of this group's studies of the acoustic properties of microcalcifications as characterized through the application of classical theoretical models, including the Faran model, and through clinical studies of suspected *in vivo* microcalcifications and breast tissue backscatter. The similarities found between theoretical and clinical results strongly suggest that microcalcifications have elastic scattering properties. The findings are presented in an analysis of the relative impact of phase aberration, spatial resolution, and tissue backscatter on the visualization of microcalcifications with ultrasound. [Work supported by NIH Grant No. RO1-CA43334.]

10:15

4aBVb5. Acousto-mechanical imaging for cancer detection. Armen Sarvazyan (Artann Labs., 22 Landsdowne Rd., East Brunswick, NJ 08816)

One of the hot areas of ultrasonic medical diagnostics is elasticity imaging (EI). Recently, an alternative method of imaging tissue structures in terms of their elasticity, the method of mechanical imaging (MI), has emerged. The essence of MI is the solution to an inverse problem using the data of stress patterns on the surface of tissue compressed by a force sensor array [A. P. Sarvazyan, Proc. Tenth IEEE Symposium ''Computer based medical systems,'' Maribor, Slovenia, 120–125 (1997)]. A prototype of the device for mechanical imaging of the prostate has been developed and feasibility of MI technology has been proven in recent clinical studies. The present paper describes a modality of medical imaging, acousto-mechanical imaging (AMI) comprising features of both EI and MI. The data on the stress pattern measured by the force sensor array of MI complement the strain data obtained by ultrasonic EI device. An embodiment of the AMI device comprises a force-sensing array made of sound transparent film inserted between an ultrasonic scanner probe and the tissue. Synergy of combining these two complementary methods results in the fact that for certain applications the AMI technology has a diagnostic potential much superior to that of EI and MI separately.

10:30–10:45 Break

10:45

4aBVb6. Shear wave excitation in a rubberlike medium by focused shock pulse. Yury A. Pishchal'nikov, Valery G. Andreev, Oleg V. Rudenko, Oleg A. Sapozhnikov (Dept. of Acoust., Phys. Faculty, Moscow State Univ., Moscow 119899, Russia, yura@naexp.phys.msu.su), and Armen P. Sarvazyan (Artann Labs., East Brunswick, NJ 08816)

Ultrasonic excitation of shear waves is important for elasticity imaging of biological tissue. In the shear wave elasticity imaging shear strain is remotely induced in tissue by the radiation force of focused modulated ultrasound. Previous study has shown that tone bursts of millisecond duration at megahertz frequencies with intensities and exposures typical for diagnostic ultrasound may induce bulk radiation force of the order of 0.01 N/cm^3 and shear displacements of the order of 10 μ. In this work, new nonlinear mechanisms are discussed that provide detectable shear displacements by means of single submicrosecond pulses. Two new possibilities are discussed. Significantly enhanced radiation force and greater shear displacement can be produced in tissue by single acoustic videopulse with nonzero momentum. Another possibility is based on the enhanced nonlinear absorption which occurs after the shock is formed in the temporal profile of the pulse. Nonlinear mathematical models describing the process of generation of shear waves using single pulses are developed. Experimentally, single focused pulses are excited by high-power laser radiation inside an optoacoustic cell. Shear displacements are detected in a sample of gel using optical scattering. Results of modeling and experiments are in good agreement. [Work supported by CRDF and RFBR.]

11:00

4aBVb7. A virtual source method for evaluating ultrasound propagation through tissues mapped from medical images. Jie Sun and Kullervo Hynynen (Div. of MRI, Dept. of Radiology, Brigham and Women's Hospital, Harvard Med. School, 221 Longwood Ave., Boston, MA 02115)

A numerical model which uses digitized layer interfaces was developed for calculating multilayer ultrasound propagation, reflection, and refraction. This model was used to project the acoustic field from a spherical phased array (diameter 10 cm, F number 1, frequency 1 MHz) into multilayer tissues, and was verified by performing phantom measurements. A section on each layer interface in the propagation path was used as a virtual source. Digitized profiles of the tissue layer interfaces of arbitrary shapes can be obtained from MRI, CT, or ultrasound. The phase delay for each element of the phased array was first calculated, and then applied to each element for phase correction. With no phase correction, the focus of the array was shifted and defocused. By using the phased array, the shifted focal point was corrected and the side lobes were reduced. The effects of the array element size (number) were investigated for a deep sonication where the layer thickness information was obtained from a series of MRI scans of a volunteer. For this particular array, it is shown that the element size of 0.39 cm^2 is small enough to produce a near-optimum focus. [Work supported by NCI Grant No. CA 46627.]

11:15

4aBVb8. An A-mode ultrasound technique for tracking the advance of coagulation front in laser irradiated tissue. Zhigang Sun, Hao Ying, Brent Bell, Massoud Motamedi (Biomed. Eng. Ctr., The Univ. of Texas Medical Branch, Galveston, TX 77555-0456), and Jialiang Lu (The Univ. of Houston, Houston, TX 77204)

This communication presents an A-mode ultrasound technique for automatically determining the advance of coagulation damage front in laser irradiated tissue. The basic assumption underlying our technique is that the waveform of the echo signal scattered from a tissue region undergoing coagulation should be changing more rapidly than tissue regions where no coagulation is taking place. In this technique, we first track rf echo signals scattered from many small tissue regions during heating by computing the cross-correlation coefficient between two consecutively acquired echo signals. We then use the resulting coefficients as a measure of waveform change to determine the position of coagulation front via an automatic procedure. To test our technique, we carried out ten *in-vitro* experiments in which pig liver samples were irradiated using an Nd:YAG laser with fluence in the range of 32.4–112.0 W/cm^2 for 290 s. A 10-MHz broadband single-element spherical focused ultrasound transducer was used to detect the thermal coagulation front. The mean-square difference between ultrasonically and histologically determined coagulation depths was 1.0 mm, whereas the mean coagulation depth was 7.2 mm. This good agreement between ultrasonically and histologically determined results shows the potential of our technique for monitoring coagulative tissue damage during thermal therapy.

11:30

4aBVb9. Two-dimensional nonlinear propagation of pulsed ultrasound through a tissue-like material. Ibrahim M. Hallaj, Steven G. Kargl (Appl. Phys. Lab., Univ. of Washington, Seattle, WA 98105), Ronald A. Roy, and Robin O. Cleveland (Boston Univ., Boston, MA 02215)

During a lithotripsy operation or ultrasound surgery, high-intensity acoustic waves propagate through tissue. Damage and thermal dose delivered to the target site and to tissue neighboring the target site is of significant interest in such operations. Results of a study of the propagation of intense ultrasonic pulses through a tissuelike material are presented. Two-dimensional simulations showing the effects of nonlinearity in inhomogeneous absorbing materials are used to calculate the acoustic pressure field, from which input is obtained for a bioheat equation calculation. Results show the mean and peak acoustic pressures within a target volume as a function of time, and the thermal dose delivered to a target volume due to a pulsed source in the range of frequencies used for ultrasound surgery. The degradation of the focusing ability of an ultrasonic array in nonlinear inhomogeneous media is illustrated, and conclusions regarding therapeutic ultrasound are made. [Work supported by ONR and DARPA.]

Session 4aEA

Engineering Acoustics: Sensors for Smart Systems

Robert D. Corsaro, Chair
Naval Research Laboratory, Code 7135, 4555 Overlook Avenue, SW, Washington, DC 20375-5000

Chair's Introduction—9:15

Invited Papers

9:20

4aEA1. Sensor engineering at the microscale. Thomas B. Gabrielson (Appl. Res. Lab., Penn State Univ., P. O. Box 30, State College, PA 16804, tbg3@psu.edu)

Modern techniques for microfabrication have led to visions of significant new sensor applications. However, the translation of devices from centimeter scales to micrometer scales can cause effects that are unimportant at the larger scales to dominate the behavior at the smaller scales. When sensor structures approach material grain sizes or ferroelectric or ferromagnetic domain sizes, design philosophies need to be adjusted; when clearances are very small, viscous and thermal effects can dominate the dynamic response. While microfabrication has made dimensional downscaling of several orders of magnitude possible, intuition and ''engineering practice'' developed on the macroscale may lead to unsound devices on the microscale. Successful miniaturization of devices and structures depends on an understanding of the changing roles of various forces, energies, and interactions on the appropriate scales. Even with this level of understanding, optimum performance may not be achieved by simple shrinking of a macroscale device. Nowhere is this more evident than in biological organisms where markedly different solutions to sensing, searching, navigation, locomotion, and thermal regulation are used by organisms of different sizes.

9:45

4aEA2. Area averaging sensors for vibro-acoustic control. Brian H. Houston and Robert D. Corsaro (Naval Res. Lab., Code 7130, Washington, DC 20375)

NRL has been conducting acoustic control research for underwater and in-air sound and vibration applications. In many of these studies, the success of the control system depends substantially on the ability of the sensors to measure the physical quantities of interest accurately while ignoring extraneous contributions. For example, the sensors developed for use in the Acoustic Boundary Control (ABC) research were required to monitor accurately the normal component of the average acoustic field on the tile surface, while being insensitive to transverse and high wave number components. The extremely low noise figures required in this study forced the development of special-purpose acoustic pressure and velocity sensors which now are finding use in other applications, including vibration isolation mounts.

Contributed Papers

10:10

4aEA3. Sound power emission measurement and control on a 300-t hydraulic press. Fabrizio Bronuzzi, Caterina Cigna, Mario Patrucco (Dip. Georisorse e Territorio, Politecnico di Torino, Corso Duca degli Abruzzi 24, 10129 Torino, Italy, patrucco@vdiget.polito.it), and Maurizio Sassone (''E. Sassone'' s.r.l., Monale (AT) Italy)

The paper summarizes the results of research work carried out to evaluate properly the noise emission characteristics of a 300-t hydraulic press in a complex noise propagation environment. On the basis of the collected data, some control measures were designed, and the results are discussed. As well known, noise is widely affected by the acoustical environment the sound propagates through, and an effective description of the source emission properties is required for the control design. Traditional measurement techniques are not the best for this purpose, because the information on acoustic pressure depends on the environment. To overcome this problem, the emitted acoustic power was then evaluated by means of acoustic intensity measurements; in such a way it is possible to qualify real and complex sound sources, locating the contribution of single parts that participate in sound generation. Resulting maps describing the contribution of every single part, an effective individuation of improvement strategies was possible, since the priorities for intervention were available. A new campaign of sound intensity measurements confirmed the expected control measures' results, and the direct measurements of the operators' exposure confirmed the success of the intervention.

10:25

4aEA4. Active control of sound transmission through an industrial sound encapsulation. Paul Sas, Wouter Dehandschutter, Rene Boonen, and Antonio Vecchio (Dept. of Mech. Eng., Div. PMA, Katholieke Universiteit Leuven, Celestijnenlaan 300B, B-3001 Heverlee, Belgium, wouter.dehandschutter@mech.kuleuven.ac.be)

In this paper, an active control system is developed to enhance the sound transmission loss of a sound encapsulation at low frequencies. The sound encapsulation is used to shield a noise source, such as an air compressor. The control systems comprise two control loudspeakers and two error microphones. The Filtered-X LMS adaptive feedforward control algorithm is used to drive the signals at the error microphones to a minimum. The reference signal for the control algorithm is taken from the noise source, i.e., the tacho signal of the compressor. This study deals with the optimization of the location of control sources and error sensors in view of the achieved control performance. From this perspective it is important to note that minimizing the sound-pressure level at the error microphones, which are constrained to be located inside the sound encapsulation, does not necessarily reduce the acoustic intensity measured in the

far field of the encapsulation (or at least, not to the same degree). Relevant issues which are considered are: the coupling of control sources with the acoustic modes of the enclosure, changes to the interior sound field due to active control, etc.

10:40

4aEA5. Structural vibration mode imaging using photorefractive holography. K. L. Telschow and V. A. Deason (Idaho Natl. Eng. and Environ. Lab., Lockheed Martin Idaho Technologies Co., MS 2209, Idaho Falls, ID 83415-2209, telsch@inel.gov)

Photorefractive processing of optical interference offers a noncontacting optical method for vibration detection that forms an optical ''lock-in'' amplifier. Multiwave mixing with synchronous detection allows measurement of both the vibration amplitude and phase of a vibrating surface directly as a function of the excitation frequency. Narrow bandwidth detection with flat frequency response can be achieved at frequencies above the photorefractive response ($\sim$100 Hz). A minimum detectable displacement amplitude of a few picometers has been demonstrated for a point measurement, with the possibility of further improvement. The method can also be configured to provide full-field imaging of resonant vibrational modes of materials with various microstructures and shapes. Both specularly and diffusely reflecting surfaces can be accommodated. Results will be presented showing the noncontacting imaging capabilities for vibrational spectral analysis of structures and characterization of material properties that these sensors provide for smart systems. [Work suported by the U.S. Department of Energy.]

10:55

4aEA6. Piezoelectric resonant sensor for sound velocity of liquids. Helmut Nowotny (Institut für Theoretische Physik, Vienna Univ. of Technol., Wiedner Hauptstr. 8/136, A-1040 Wien, Austria), Ewald Benes, Branka Devcic-Kuhar, Martin Gröschl, Dagmar Harrer, Rudolf Thalhammer, and Felix Trampler (Institut für Allgemeine Physik, Vienna Univ. of Technol., Wiedner Hauptstr. 8/134, A-1040 Wien, Austria, benes@iap.tuwien.ac.at)

The sensor consists of two circular piezoelectric plates with the liquid in between. Each piezoelectric plate is a bulk resonator with two electrodes driven in thickness extensional modes. The two resonator plates are electrically connected in parallel and the total admittance is measured in the vicinity of two quasiharmonic series resonance frequencies by a PC-controlled electric admittance measurement system. The extensional thickness mode purity was checked directly by corresponding mode pattern measurements using a laser speckle vibration amplitude measurement system. The sensor element is mounted in a thermostated aluminum housing to avoid temperature effects and to obtain the sound velocity of the liquid at a defined temperature. Two quasiharmonic resonance frequencies are used to allow the elimination of the unknown density of the liquid. The sound velocity is calculated out of a rigorous one-dimensional theoretical model of the three-layer sensor arrangement. In addition to the determination of the sound velocity of the liquid, the mass density of the liquid can be obtained from the same primary measurements. However, at present the accuracy of the density value is one order of magnitude lower than that of the velocity (typically 0.1% for water, acetone, glycerin).

11:10

4aEA7. Transient behavior of acoustic gyrometers. Philippa Dupire and Michel Bruneau (Lab. d'Acoustique, IAM—UMR 6613, Univ. du Maine, av. O. Messiaen, 72085 Le Mans Cedex, France, amb@laum.univ-lemans.fr)

Acoustic gyrometry was developed during the past 15 years as a new miniaturized or low-cost technology. The operation of acoustic gyrometers employs acoustic fields inside fluid-filled resonant cavities to determine angular velocities. Until now, research efforts and design methodology have concentrated both on trapezoidal miniaturized gyros (etched on silicon chips) as well as on cylindrical gyros designed using classical techniques. These approaches are restricted to the frequency domain, which involves only the steady Coriolis effect. Nowadays, the need for a time domain solution for inertial effects on acoustic fields clearly arises when dealing with applications involving strong variations of the rotation rates. This paper aims at providing such advances in ''inertial-acoustic'' theory and modeling. The discussion covers cylindrical gyros because the presence of an unsteady rotational velocity gradient (with respect to the radial coordinate) of a gas in a cylindrical cavity adds one of the most important features both in the basic physics underlying ''inertial-acoustic'' transient processes and in the behavior of the gyros. In conclusion, only some new requirements, which have to be taken into account in the design of acoustic gyros, will be addressed as a discussion of the transient behavior in detail is beyond the scope of this paper. [Work supported by DRET-DGA.]

THURSDAY MORNING, 25 JUNE 1998 GRAND CRESCENT (W), 8:00 TO 10:45 A.M.

Session 4aMU

Musical Acoustics: Timbre of Musical Sound I

James W. Beauchamp, Chair
University of Illinois, 2136 Music Building, 1114 West Nevada, Urbana, Illinois 61801

Invited Papers

8:00

4aMU1. Spectral versus harmonic information for timbre: Pilot and experimental results. Einar Mencl (Haskins Labs., 270 Crown St., New Haven, CT 06511, einar@tom.haskins.yale.edu)

This work summarizes results from six experiments on steady-state timbre perception. The primary variable of interest was the physical similarity of tones within each trial: tones were either similar by virtue of their spectral envelope (i.e., they shared resonance characteristics) or their harmonic envelope (i.e., they shared overtone characteristics). These two continua were chosen as candidates for basic perceptual dimensions because (1) they each contain specific types of information about the sound source/event, and (2) a cursory computational analysis suggests that very different processes are needed to extract information about resonances versus overtones from the auditory input. It was reasoned that if different subsystems subserve the processing of timbre along each of these two dimensions, then performance on perceptual tasks would dissociate under appropriate experimental manipulations. These included

(1) attentional focus and response type (timbre similarity judgment, pitch direction judgment, or pitch contour judgment), (2) perceptual priming, and (3) auditory masking. Results demonstrate that in general both dimensions of similarity can support perceptual priming, but also show specific dissociations which inform the understanding of steady-state timbre perception. [Portions of this research were supported by NSF Grant No. DBS-9222358 to Jamshed Bharucha.]

8:25

4aMU2. Macrotimbre: Contribution of attack, steady state, and verbal attributes. Gregory J. Sandell (Parmly Hearing Inst., Loyola Univ., 6525 N. Sheridan, Chicago, IL 60626, sandell@sparky.parmly.luc.edu)

The timbre attributes of a source that are contained across variations in pitch, dynamic, and articulation can be referred to as its ''macrotimbre.'' The timbre of a musical instrument changes across pitch, and thus learning its macrotimbre can be dependent upon the ability to make comparisons among individual pitches. Using 20 pitches shared by two easily confused instruments ($C4$ to $G5$, oboe and English horn), listeners performed better at categorizing novel notes (from a pitch range different from those heard in training) when training facilitated comparison of notes (versus isolated presentations) [Sandell and Chronopoulos, Proc. Third ESCOM Conf., Uppsala, Sweden, 222–227 (1997)]. This indicated that information contained across pitches was useful in generalizing timbre knowledge to previously unheard notes. The present study used (1) multiple versions of the tones (natural, shortened, attack + synthetic steady-state, synthetic steady-state alone), (2) a sorting task under conditions where comparisons were facilitated or hindered, and (3) ratings of verbal attributes (nasal, breathy, muffled, etc.) to isolate the relevant across-pitch timbre properties, and appraise the relative contributions of the attack and steady state.

8:50

4aMU3. Methods for measurement and manipulation of timbral physical correlates. James W. Beauchamp (School of Music and Dept. of Elec. and Comput. Eng., Univ. of Illinois at Urbana-Champaign, Urbana, IL 61801)

Multidimensional scaling calculations based on the results of listening tests comparing musical sounds almost always identify one perceptual dimension which is strongly correlated with a measure of the spectral centroid, whereas other dimensions are less obviously correlated to particular physical parameters. Some parameters which have been suggested as good correlates are attack time, spectral flux, and spectral irregularity [J. Krimphoff *et al.*, J. Phys. IV (C5), 625–628 (1994)]. Also, spectral rolloff, preattack noise, general noisiness, and inharmonicity have been proposed as important factors. Based on time-variant spectral analysis with custom software [J. Beauchamp, Audio Eng. Soc. Preprint No. 3479], these parameters and others can be quantified and manipulated, leading to various applications. For example, synthesis using simplified parameter contours provides stimuli for discrimination testing to reveal listeners' sensitivities to these parameters. Measurement of spectral differences can be correlated to discrimination results. Equalization of parameters common to two instrument sounds allows listeners to focus on those timbral aspects which are different. Finally, hybrid musical sounds created by interchanging parameter contours between instruments can shed light on timbral perception.

9:15–9:30 Break

9:30

4aMU4. Perceptual independence of excitor and resonator properties in percussive instrument sounds. Stephen McAdams (Inst. de Recherche et de Coordination Acoustique/Musique (IRCAM), 1 place Igor Stravinsky, F-75004 Paris, France), Koei Kudo, and Holle Kirchner (IRCAM, F-75004 Paris, France)

The perceptual independence of various sensory dimensions has been studied using a speeded classification paradigm developed by Garner. In this paradigm the speed of classification of items varied along one dimension is compared to the same classification when random variation along a second dimension is present. Perceptual interaction is inferred when performance in the latter is slower or worse compared to the former. Such interactions have been shown for simple dimensions such as pitch, loudness, and a spectral dimension of timbre. The present study tested the independence of complex dimensions with overlapping spectral and temporal characteristics. One dimension consisted of the hardness of a percussion mallet (excitor property) and the other of either xylophone bar density or tympani size (resonator properties). These complex source-related dimensions show evidence of interaction in both performance scores and reaction times. A greater independence is found when unrelated variation is introduced along a third dimension (striking force), although performance scores decrease globally. The results are discussed in terms of previously reported interactions among simple dimensions and interactions among complex source properties.

9:55

4aMU5. The effect of amplitude and centroid trajectories on the timbre of percussive and nonpercussive orchestral instruments. John M. Hajda (Prog. in Systematic Musicology, Dept. of Ethnomusicology, UCLA, Box 951657, Los Angeles, CA 90095-1657)

This study explores the salient acoustical attributes of the associative aspects of Western orchestral instrument timbres. Association refers to a subject's ability to connect or link a stimulus with its intended comparison; this occurs in tasks such as identification, matching, or categorization. Unlike most previous research in instrument association, this study considers both continuant (nonpercussive) and impulse (percussive) classes of instruments. Certain acoustical transformations, such as the partitioning of a signal into attack, steady state, and decay, are not possible for both classes. Findings which have been made regarding the salience of signal segments must be questioned in terms of their application to a more general and inclusive theory of musical timbre. Previous perceptual and timbre synthesis research has demonstrated the importance of global rms amplitude and spectral centroid (the

amplitude-weighted mean frequency). Here, both amplitude and centroid are considered in terms of individual trajectories. The evolutions of and relationships between these trajectories are markedly distinct for continuant and impulse classes of instruments. The perceptual significance of these trajectories is measured through the experimental control of their magnitude and direction. A convergence method is employed to ascertain the necessary and sufficient conditions for instrument association.

10:20

4aMU6. Computer identification of wind instruments using cepstral coefficients. Judith C. Brown (Phys. Dept., Wellesley College, Wellesley, MA 02181, brown@media.mit.edu)

Earlier results on computer recognition of the oboe and saxophone [J. C. Brown, J. Acoust. Soc. Am. **101**, 3167(A) (1997); ''Musical instrument identification using cepstral coefficients as features'' submitted to J. Acoust. Soc. Am.] have been extended to include the clarinet and the flute. Approximately 30 samples of duration 2–10 s of each of the four instrument types comprise the test set. The training set consists of longer segments of approximately 1 min duration. Eighteen mel-based cepstral coefficients were calculated for each of the sounds. The training data were summarized by a cluster analysis with Gaussian probability density functions formed from the mean and variance for each of the clusters. The probability of belonging to each of the four classes was then calculated for the test sounds, and a Bayes decision rule was invoked to assign them to one of the classes. Results indicate that this is a very promising method for automatic recognition of musical instruments.

THURSDAY MORNING, 25 JUNE 1998

CASCADE BALLROOM I, SECTION A (W), 8:40 TO 10:45 A.M.

Session 4aNS

Noise: Acoustical Materials and Propagation

Trevor R. T. Nightingale, Chair
National Research Council, IRC Acoustics, Building M27, Montreal Road, Ottawa, ON K1A 0R6, Canada

Chair's Introduction—8:40

Contributed Papers

8:45

4aNS1. Effect of the flat panel resonance on the absorption characteristics of a microperforated-panel absorber. Ke Liu (Inst. of Acoust., Academia Sinica., P.O. Box 2712, Beijing 100080, PROC), Fenglei Jiao (Beijng Inst. of Light Industry, Beijing 100037, PROC), and Hui Ding (Beijing Municipal Inst. of Labour Protection, Beijing 100054, PROC)

The flat panel resonance can affect the absorption characteristics of a microperforated-panel absorber. This effect was not considered in the existing theory. In this paper, the method of measuring the no-hole flat panel resonance is applied to analyze this effect. The sample is a stainless steel microperforated panel, with the diameter of perforations $d=0.2$ mm, the separation of holes $b=2.5$ mm, the thickness of panel $t=0.2$ mm, and the thickness of cavity behind the microperforated-panel $D=95$ mm. Theoretically, its resonance frequency is 508 Hz, and the maximum absorption coefficient is 0.9. However, the experimental result shows that there are two absorption peaks at 280 and 680 Hz, respectively. Two resonance peaks also appear as a sample of a no-hole flat panel is measured. When the thickness of cavity is changed, the result is still similar. Moreover, when sound intensity is enhanced, two resonance peaks also exist, and move towards low frequency. Therefore, it is possible that the absorption frequency band can be widened, or the absorption coefficient within absorption frequency band can be enlarged, by using the resonance frequency of different materials. [Work supported by the National Natural Science Foundation of China.]

9:00

4aNS2. Prediction system for rotor–stator interaction noise generation, propagation, and optimization of acoustic liners. Yadong Lu (Inst. of Acoust., Acad. of Sci., P.O. Box 2712, Beijing, 100080, PROC), Zongan Hu, and Jiya Cui (Beijing Univ. of Aeronautics and Astronautics, Beijing, 100083, PROC)

Based upon the acoustic mode matches between the source modes and propagating modes, this paper applies the flexible tolerance optimization method to optimize the acoustic parameters (impedance), geometric structure parameters, such as open area ratio, cavity depth, and hole diameter, and operating condition parameters, such as blade passing frequency. The optimum values of the design variables are determined when the in-duct sound suppression approaches the maximum. It can be derived from the optimum results that the emphasis of the engineering optimization design of the perforated plate honeycomb structure should be placed on the optimum choice of the open area ratio and cavity depth. Some other referential criteria for the engineering design of the multi-linings are also provided. Thus, the theoretical prediction system for rotor–stator interaction noise generation and in-duct propagation and optimization of acoustic liners has been developed in this paper. By means of this prediction system, the acoustically multi-sectioned treatments can be theoretically designed for the suppression of rotor–stator interaction noise with discrete frequency, in advance of the beginning of the practical engineering design of acoustic liners.

9:15

4aNS3. The Wiener–Hopf technique and scattering of acoustic waves in ducts. Fredrik Albertson (MWL, Dept. of Vehicle Eng., Royal Inst. of Technol., 100 44 Stockholm, Sweden, fal@fkt.kth.se)

The reactive silencer is the standard product for attenuating noise in ducts with flue gases. A model problem is solved for such a silencer by calculating the scattering of acoustic waves at sharp edges in a two-dimensional waveguide. In this waveguide the walls are considered acoustically hard and to ensure the existence of a unique solution an edge condition is applied. An exact analytical solution is found using the Wiener–Hopf technique, which relies heavily on the Fourier transform with assumptions on regularity of the solution. Control methods are employed to check the theory and the accuracy of the numerical results. One of the control methods is derived from the energy conservation principle. Symmetries, so-called reciprocity relations, are found for the reflection and transmission matrices and are first used to check the analytic calculation and later to reduce the numerical work. A quasistationary model is also used for control purposes. It is valid for low frequencies and checks the phase of the transmission and reflection coefficients, while the energy method only covers the corresponding absolute values. Physical results as the behavior of the reflection and transmission coefficients are also presented. These coefficients are analyzed as functions of both frequency and geometry.

9:30

4aNS4. Acoustic response of a thin poroelastic plate. Kirill V. Horoshenkov (Dept. of Civil and Environ. Eng., Univ. of Bradford, Bradford BD7 1DP, England) and Kimihiro Sakagami (Kobe Univ., Rokko, Nada, Kobe, 657 Japan)

The expression for the viscosity correction function in the Attenborough model for the acoustic properties of rigid frame porous media is modified to account for the viscothermal effects in the oscillatory flow in the vibrating frame. The Helmholtz integral equation formulation is used to produce the solution for the sound field reflected from an infinite, thin, porous, elastic plate. The effect of an air cavity behind the plate is considered. A parametric study is performed to predict the effect of variations in the microscopic parameters of the poroelastic plate.

9:45–10:00 Break

10:00

4aNS5. A nondestructive method for measuring the airflow resistance of locally reacting jet engine nacelle components. T. R. T. Nightingale (Inst. for Res. in Construction, Natl. Res. Council Canada, Ottawa, ON K1A 0R6, Canada) and Brandon Tinianow (Johns Manville Tech. Ctr., Littleton, CO 80127)

As a noise control measure to reduce inlet fan noise of jet engines, the nacelles are often equipped with a locally reacting absorber formed by a thin resistive wire mesh placed over a honeycomb structure. Historically there has been a problem to measure, in nondestructive fashion, the airflow resistance of repaired or refurbished nacelle parts for the purpose of demonstrating OEM compliance. ASTM-C522 is the traditional airflow resistance test method and is destructive, requiring a sample be cut from the specimen and fit into an apparatus where the flow velocity due to a constant pressure drop can be measured. This paper presents the theory and practical considerations of a nondestructive test method to measure the airflow resistance using an ASTM-E1050 impedance tube placed against the wire mesh of the complete specimen. Transfer matrix theory is used to describe the combined impedance of the wire mesh and air cavity. It is shown that at the air cavity resonance frequency the measured system impedance will be that of the wire mesh with the real part being the airflow resistance. Airflow resistance estimates obtained using the presented method and the ASTM-C522 test method are shown to be in excellent agreement.

10:15

4aNS6. A measurement of sound power lever of noise source—Vibration velocity method. Weicheng Yang (Res. Inst. of Household Elec. Appliance of China, No. 6 Yuetan Beixiaojie, Beijing 100037, PROC) and Wuzhi Qiao (Beijing Inst. of Light Industry, Beijing 100037, PROC)

Sometimes it is necessary to measure on-line the noise of some products, such as the household electrical appliances and their parts (motors, compressors, and contacts). Nowadays, several measurements (sound intensity measurement and vibration flow measurement) have been applied, but are not good enough. In this paper, another on-line measurement of sound power lever of noise source vibration, velocity measurement, is proposed. This measurement has two advantages, viz., without the influence of the surrounding area and with less measuring time. After the theoretical analysis and the experiment to the small-type compressor of the refrigerator in a semi-anechoic chamber, the sound-radiating efficiency curve is attained. Based on the vibration velocity measurement, the sound power lever of the compressor is predicated through this curve. The error of the predicated result is less than 2 dB, compared with that of the standard ISO 3745.

10:30

4aNS7. The limit of absorbing characteristics of folded resonators silencers and ways of its realization. Roudolf Starobinski (Lab. Acoustique, Univ. Le Maine, B.P. 535, 72017 Le Mans, France) and Jean Kergomard (Univ. Le Maine, 72017 Le Mans, France)

The folded resonance silencer is a version of 1/4-wave resonator having a very compact shape. They are usually used for suppressing of narrow-band and discrete frequency noise. The sets of resonators tuned to adjacent frequencies can be successfully used for a wide frequency band as well. The general theorems governing the behavior of these resonators are represented in this report. The limits of the muffling characteristics of the resonators and the ways of its realization are investigated on their base. It is shown that the potential muffling ability of a folded linear resonator is 1.2 dB less than for a Helmholtz resonator. However, taking into consideration the possibility of increasing the neck's cross-sectional area, they are more effective at high sound levels because of the higher quality factors. It is shown also that with the help of coupling folded resonators the resonance effects can be created in the frequency band wider than two octaves. These peculiarities make the use of the folded resonators very interesting in noise control systems.

Session 4aPAa

Physical Acoustics and Bioresponse to Vibration/Biomedical Ultrasound: Cavitation Dynamics: In Memoriam Hugh Flynn I

Charles C. Church, Cochair
Acusphere, Inc., 38 Sidney Street, Cambridge, Massachusetts 02139

Ronald A. Roy, Cochair
Department of Aerospace and Mechanical Engineering, Boston University, 110 Cummington Street, Boston, Massachusetts 02215

Chair's Introduction—9:15

Invited Papers

9:20

4aPAa1. Hugh Guthrie Flynn, 1912–1997. David T. Blackstock (Appl. Res. Labs. and Dept. of Mech. Eng., Univ. of Texas, Austin, TX 78712-1063, dtb@mail.utexas.edu)

Hugh Flynn spent most of his first three decades in Columbus, Ohio. During the 1930s he was a newspaper crime reporter and also studied physics at Ohio State University. After graduating from Ohio State in 1939, he took a job with the Navy's Bureau of Ordnance and was a Naval officer during WWII. While working for the NAS Committee on Undersea Warfare after the war, he met F. V. Hunt, who encouraged him to study acoustics at Harvard. There Hugh became interested in cavitation, the beginning of a lifelong love affair with bubbles. After his doctoral degree in 1956, he stayed on as one of Hunt's postdocs. The year 1960 was a propitious one. He married Prudence Turgeon and joined the faculty of the new Electrical Engineering Department at the University of Rochester. He spent the rest of his life in Rochester, where he solidified his international reputation as a respected authority on cavitation and fathered three children, Kitty, Molly, and Nathaniel. Despite retiring as Professor Emeritus in 1978, he stayed active in research for the rest of his life. His projects during retirement included bubble phenomena in medical ultrasonics and nuclear fusion from cavitation.

9:40

4aPAa2. Sonically induced growth of gas bubbles by diffusion. Anthony I. Eller (Sci. Applications Intl. Corp., 1710 Goodridge Dr., McLean, VA 22102, eller@osg.saic.com)

Small gas bubbles that might normally dissolve and disappear may instead grow if set into pulsations of sufficient amplitude by an applied sound field. The mechanism involved is a fundamental asymmetry related to the spherical geometry of the conditions which favors diffusion of gas into the bubble during its expanded state over diffusion out of the bubble during a contracted state. Values of the threshold acoustic pressure for growth were computed by a theory that accounts for spherical bubble dynamics as well as convection terms in the diffusion equation and shows general agreement with measured counterparts. Rates of bubble growth for above-threshold conditions can be unexpectedly high if nonspherically symmetric acoustic streaming occurs. A closely related phenomenon—the onset of nonspherical bubble vibration—occurs within the same general range of acoustic pressures observed for bubble growth and will also be described.

10:00

4aPAa3. Reminiscences on bubble dynamics and cavitation. M. Strasberg (Code 702, David Taylor Model Basin, NSWC, 9500 Macarthur Blvd., West Bethesda, MD 20817-5700)

A review of various topics in cavitation and bubble dynamics discussed with Hugh Flynn, most of them still containing unsolved details. Among these are rectified diffusion of gas into pulsating bubbles and the effect of surface films; thermal rectified diffusion and the associated temperature rise; parametric shape oscillations of bubbles and the failure to observe resonances; the threshold conditions for cavitation, the cause for the persistence of the required nuclei, and the effect of a temporary static pressure increase on the nuclei; the dynamics of an individual cavity growing and collapsing in a compressible liquid and the associated internal pressure and radiated sound; and the cavitation noise radiated by cavitating propellers.

10:20

4aPAa4. Physics of acoustic cavitation in liquids: H. G. Flynn's review 35 years later. A. Prosperetti (Dept. of Mech. Eng., The Johns Hopkins Univ., Baltimore, MD 21218)

In 1964 Hugh G. Flynn published in Vol. I, Part B of the series *Physical Acoustics*, edited by W. P. Mason (Academic Press), a long review article by the title ''Physics of Acoustic Cavitation in Liquids.'' This was the first comprehensive review of the topic and has had a enormous influence on the field, serving as an introduction to cavitation and bubble dynamics for at least two generations of researchers (including the present author). A summary of that work will be presented in the light of subsequent developments and the present understanding of the cavitation phenomena. [Work supported by NASA.]

10:40

4aPAa5. Cavitation terminology. Wesley L. Nyborg (Phys. Dept., Univ. of Vermont, Burlington, VT 05405)

Among the important contributions Flynn made to the subject of acoustic cavitation was his introduction of concepts and language which have greatly helped our thinking and our communications. In 1964, he defined two acceleration functions and used them for classifying the behavior of gas-filled cavities in a sound field. The more violent kind of behavior, involving ''collapse,'' occurs when one of the functions, the inertial function, dominates at a critical time during cavity contraction. This type of cavitation was initially called ''transient,'' partly because the collapse events are of short duration, and partly because the cavities themselves are often short-lived, disappearing through fragmentation and dissolution after collapse events. The other class of cavitation is called ''stable,'' and represented oscillatory motion which might be complicated but which persists indefinitely. This classification has become very helpful in discussing cavitation phenomena and is widely used in the literature. In recent years, however, the term ''inertial'' has come into use, replacing ''transient,'' for cavitation in which Flynn's inertial function dominates. Examples will be discussed in which Flynn's classification is applied to experimental situations.

11:00–11:15 Break

Contributed Papers

11:15

4aPAa6. Influence of nonradial experimental conditions on the acoustic scattering behavior of a single cavitation bubble. Torsten Niederdraenk (Institut für Technische Akustik, RWTH Aachen, 52056 Aachen, Germany, tni@akustik.rwth-aachen.de)

The dynamic behavior of a single cavitation bubble in a sound field can be evaluated using the acoustic scattering technique. A high-frequency image sound is scattered by the bubble, which pulsates in the primary driving sound field. The scattered sound contains information about the bubble oscillation. The shape of the bubble radius can be observed by evaluating the amplitudes of the scattered sound or by determination of frequency shifts caused by the motion of the bubble wall. If the wavelength of the high-frequency image sound and the bubble size are of the same order, or if the boundary and excitation conditions in the experimental setup do not fulfill the demand of a spatially independent primary driving field, the bubble is exposed to nonradial driving conditions. Modes of bubble oscillation may develop that can be decomposed into spherical harmonic components of different order. These modes affect the acoustic scattering behavior, because sound emissions of the nonradial contributions of bubble oscillation are characterized by different directivities. Moreover, the amplitudes of sound emitted by higher-order spherical harmonics depend more strongly on the receivers' distance to the bubble.

11:30

4aPAa7. Susceptibility of silicon wafers to acoustic microcavitation damage. Sameer I. Madanshetty and Jogesh B. Chandran (Mech. and Nuclear Eng., Kansas State Univ., 339 Durland Hall, Manhattan, KS 66506-5106)

Acoustic methods are being increasingly used in semiconductor processing. Given the increase in circuit densities being implemented on semiconductor chips the requirements for particulate contamination tolerance are becoming more stringent. The semiconductor industries roadmapped specifications for 1998, 0.25 μm feature size require control of all particulate contaminants larger than 0.08 μm. Fine cleaning of post CMP wafers is currently being done by using the megasonics process. The cleaning effectiveness depends on the strength of the acoustic fields used. In using high-intensity, high-frequency acoustic fields it is crucial to ensure that the surface of the silicon wafer does not suffer any damage due to cavitation. This paper assesses the damage potential of semiconductor wafers to acoustic microcavitation. Acoustic microcavitation is brought about by low megahertz acoustic fields giving rise to micron size bubbles that live a few microseconds. In exposing a surface to continuous waves one could obtain cavitation effects in an average, overall sense; the details of nucleation, evolution of inertial events, however, get glossed over. Both, pulsed and cw insonification were studied. Experimental measurements of the thresholds for cavitation damage of semiconductor wafers will be presented.

11:45

4aPAa8. Study of the correlations between the cavitation noise power and bubble volume rate in an acoustic cavitation bubble field. Stéphane Labouret, Jacques Frohly, Nathalie Poulain, Roger Torguet, and Francis Haine (IEMN-DOAE, Université de Valenciennes, B.P. 311, Le Mont Houy, 59 300 Valenciennes Cedex, France, labouret@univ-valenciennes.fr)

The effects of ultrasound cavitation depend on both the insonification conditions and the liquid characteristics, by means of their influence on the bubble cloud. For this reason, the study of the cavitation bubble field is of interest. The implemented hyperfrequency method allows the monitoring of the bubble volume rate, called void rate a, during the insonification, versus acoustic pressure or dissolved air concentration in water. On the other hand, the oscillating behavior of bubbles found expression in a characteristic acoustic spectrum. The cavitation noise power, defined as the integral of the cavitation acoustic spectrum, permits the estimation of the importance of the dynamic activity of the bubble field. The aim of this paper is to set the relations between the void rate evolution and the cavitation noise power. The evolutions of the cavitation noise power versus the void rate will be examined in two situations, whether the void rate evolution is due to the acoustic pressure variation or to a change of the dissolved air content in water. A link between the cavitation noise power and the increased speed of the void rate will be set.

Session 4aPAb

Physical Acoustics: Radiation and Diffraction

Andrew A. Piacsek, Chair
Department of Physics, Central Washington University, MS 7422, Ellensburg, Washington 98926

Chair's Introduction—9:15

Contributed Papers

9:20

4aPAb1. Acoustical helicoidal waves and Laguerre–Gaussian beams: Applications to scattering and to angular momentum transport. Brian T. Hefner and Philip L. Marston (Dept. of Phys., Washington State Univ., Pullman, WA 99164-2814)

Traveling waves having helicoidal wavefronts are cylindrically symmetric except for an azimuthal angular dependence of $\exp(im\phi - i\omega t)$, where m is an integer. The amplitude vanishes at the screw-phase dislocation [J. F. Nye and M. V. Berry, Proc. R. Soc. London Ser. A **336**, 165–190 (1974)] on the z propagation axis. Examples of paraxial helicoidal waves are the Laguerre–Gaussian beam solutions of the parabolic wave equation. Some potentially useful properties of acoustical helicoidal waves are analyzed. Just as for linearly polarized optical Laguerre–Gaussian beams [L. Allen *et al.*, Phys. Rev. A **45**, 8185–8189 (1992)], the ratio of the axial angular momentum flux to the acoustical beam power is found to be m/ω so that axial radiation torques are generated when the acoustical energy is absorbed. Helicoidal waves are predicted to backscatter in a way which reveals the axisymmetry of the scatterer when combined with a helicoidal mode-selective detector. Some strategies for acoustical helicoidal wave generation and detection will also be described. [Work supported by the Office of Naval Research.]

9:35

4aPAb2. Vibrations of a film with arrays of point masses. Takato Handa, Andy Piacsek, and Roger Yu (Dept. of Phys., Central Washington Univ., Ellensburg, WA 98926)

As a result of the Bloch theory, the eigenstates of an electron in a strictly periodic potential are extended and the eigenenergy levels form allowed energy bands separated by forbidden gaps. Very recently there has been a great deal of excitement in the physics community concerning the extension of the idea to electromagnetic and acoustic waves. The three-dimensional (3-D) photonic band material with a periodically varying dielectric constant has been found. The frequency band structure of a rectangle thin film with arrays of point masses on it interests us. It is assumed that the film is supported by a frame so that at the boundaries, the vibration vanishes. By taking advantage of this zero-boundary condition, a double sine wave expansion was used to transform the 2-D wave equation into a matrix equation. Up to 12 000 eigenvalues and eigenfunctions can be obtained accurately. A number of point mass configurations (or arrangements) were studied and their frequency structure provides rich insight into the physics of the system. In this report, the calculated frequency band structure of the system and the wave functions will be discussed.

9:50

4aPAb3. Absorbing boundary conditions for acoustic waves and Huygens principle. Gérard Mangiante (Lab. Mécanique d'Acoust., Ctr. Natl. Recherche Sci., 31 Chemin Joseph-Aiguier, 13402 Marseille Cedex 20, France, mangiante@lma.cnrs-mrs.fr) and Sylvestre Charles (Schlumberger Geco-Prakla, Inversion Methods, Houston, TX 77077)

When numerically simulating acoustic wave propagation, artificial boundaries must be introduced into the model to limit the area of computation. To eliminate the spurious reflections produced by these artificial boundaries, absorbing boundary conditions are needed. Such boundary conditions are usually approximated according to two main strategies: paraxial wave equation and the use of viscous media. This paper describes a new approach of the boundary conditions problem using Jessel's formulation of Huygens principle [Jessel (1991), Mangiante (1994), Charles (1996)]. A suitable distribution of Huygens secondary sources was arranged in a transition zone surrounding the model, and adjusted to produce destructive interference to cancel the wave field outside this transition zone. Perfect absorption can be theoretically obtained and can be numerically illustrated. However, due to practical difficulties, only an approximation of this solution has been derived for a finite difference scheme. This approximation consists in splitting the history of the wave field within user defined time windows. The main difficulties are related either to the space and time behavior of the Green functions, or to the solution of the wave equation problem via a finite difference scheme.

10:05

4aPAb4. Diffraction by a hard half-plane: Useful and simple frequency forms to an exact time-domain solution. Djamel Ouis (Eng. Acoust., P.O. Box 118, S-22100, Lund, Sweden, djamel.ouis@kstr.lth.se)

The general case of diffraction of a spherical wave by a hard half-plane is considered. The calculations are based on Biot and Tolstoy's exact and explicit form of the solution to the diffraction problem by a hard wedge. This solution is given in the time domain but its Fourier transform is not available. It is the aim of this paper to present some useful and simple approximations for the early time diffracted field which when transformed into the frequency domain is shown to be satisfactory in comparison to numerical integration. This could give some valuable contributions to calculations on scattering by noise barriers and the like. The results of some experiments are also presented.

10:20–10:35 Break

10:35

4aPAb5. A discussion on nonradiating sources. Ricardo E. Musafir (Acoust. & Vib. Lab/Dept. of Mech. Eng. and Hydraulics Dept., Univ. Federal do Rio de Janeiro-CP 68503, 21945-970, RJ, Brazil, rem@serv.com.ufrj.br)

The fact that a given source sound field can be generated by more than one source distribution is quite well known but is frequently given an overexaggerated importance. This paper discusses the structure of sources of silence, the name given by Doak [Proc. I.O.A., 10-2, 693–700 (1988)] to the reactive part of a source distribution, i.e., to the part with no far field and which is responsible for the ambiguity. It is shown that, since all ambiguities can always be traced back to the impossibility of distinction between the sound fields of a point monopole and of an isotropic point quadrupole, a simple criteria can be established for identifying a general source distribution as active or reactive. The particular case of the velocity field due to a point dipole (i.e., to a point force) in an homogeneous medium at rest is worked out in detail, since it contains a source of silence (associated with the vorticity at the source point) which is frequently mishandled. The concepts presented are applied to aeroacoustic wave equations, to discuss optimization of the choice of dependent variable as well as some case of identification of nonradiating sources found in the literature.

10:50

4aPAb6. To lend an ear to the dimensionality of space. Jean Hardy (Lab. acoustique, Univ. Le Mans BP535 72017 Le Mans CNRS UMR 6613, France)

Since the laws of physics are essentially based on our feeling of the real word, it seems natural to consider the problem of the space dimensionality. The usual conception of the space follows from the vision and leads to the conviction that dimensionality of the real word is 3; a similar approach but using the ears is done in this work. An acoustical source emits an impulse in a homogeneous and isotropic N-dimensional space. The Green's function obtained shows that odd and even dimensionalities have different behaviors. For odd dimensionalities the Green's function is a combination of a step, a delta function and its derivatives; the step function is responsible for reverberation, the other gives the structure of the impulse front which travels at the wave velocity. The even dimensionality uses the half derivatives for expressing propagation, the wave propagation is subject to reverberation which decreases with the time, the wave velocity is not well defined, and the signal transmission and consequently the communication are difficult if not impossible. An explicit demonstration with 1, 2, 3, 4, and 5 dimensions can be done using musical and speech samples.

11:05

4aPAb7. Resolution of the convected Helmholtz's equation by integral equations. M. Beldi (France Acoust., Res. and Development Dept., 11, rue du 8 Mai 1945, 60350 Berneuil sur Aisne, France) and F. S. Monastir (Univ. of Center, 5000 Monastir, Tunisia)

Using the free-field convected Green's function, the integral formulation for the acoustic velocity potential and its normal derivatives associated with Helmholtz's equation in a uniform flow is established. An original development for the calculation of the Cauchy principal value in the singular integrals is presented. As far as the finite part in the hypersingular integral is concerned, a new transformation of the convected double-layer potential normal derivative is proposed. The result of this transformation avoids the explicit calculation of the finite part. Moreover, it generalizes the famous works [M. P. Stallybrass, J. Math. Mech. **16**, 247–1286 (1967)] [A. W. Maue, Zeit. für Phys. **126**, 601 (1949)] on the resolution of the Helmholtz's equation, by integral equations, in a fluid with null velocity. This new variational method by integral equations can easily be coupled with the variational formulation for studying the convection and refraction effects of the acoustic waves in a nonuniform flow [M. Beldi, Third International Conference on Theoretical and Computational Acoustics, Newark (14–18 Aug. 1997)]. The numerical results shows the importance of these effects.

11:20

4aPAb8. Enhanced ray theory. Nikolai E. Maltsev (Instrumar Ltd., P.O. Box 13246, Stn 'A' 39 Pippy Pl., St. John's, NF A1B 4A5, Canada)

The numerical implementation of standard ray theory faces two intrinsic difficulties, the infinite value of the field at caustics and the nonlinear boundary value problem for computation of eigenrays—rays which connect source and receiver. In an earlier work [N. E. Maltsev, J. Math. Phys. **35**, 1387–1389 (1994)], a nontraditional approach was developed to treat the caustics problem. This report describes a procedure of embedding a nonlinear boundary value problem for eigenrays into the space of a higher number of dimensions, where it becomes an initial value problem for a new system of equations, differential equations for eigenrays. These two enhancements of the ray theory produce a regular method of computation of the field anywhere in the media, including shadows, for harmonic and pulse sources with all physical attributes of rays, such as arrival times and angles.

11:35

4aPAb9. Experimental study on the vortex-shedding sound from a yawed circular cylinder. Jong-Soo Choi and Hoon-Bin Hong (Dept. of Aerosp. Eng., Chungnam Nat'l Univ., 220 Kung-Dong, Taejon, Korea)

For a cylinder in a uniform flow stream, sound is generated by the fluctuating pressure on the surface of a cylinder due to the vortex shedding behind the cylinder. It is known that the major parameters in predicting the acoustic pressure at the far field are the flow velocity, the correlation length, and the fluctuating lift coefficient. In this experimental study, the correlation length of the wake is measured with hot-wire sensors and compared to the one obtained with pressure sensors on the cylinder surface. A new measurement technique for the unsteady lift estimation by using a pressure sensor rotating circumferentially is used to show reasonably good results. The surface pressure and the radiated sound are also measured simultaneously for different yaw angles and showed that the reduced normal velocity component to the cylinder axis reduces the unsteady lift fluctuation which results in lowered sound-pressure level. However, experimental result shows that ''the cosine law'' which uses the normal velocity component as a characteristic velocity for noise generation from a yawed cylinder needs to be carefully reviewed.

Session 4aPAc

Physical Acoustics: General Topics (Poster Session)

James P. Chambers, Chair

National Center for Physical Acoustics, University of Mississippi, Coliseum Drive, University, Mississippi 38677

Contributed Papers

All posters will be on display from 9:15 a.m. to 12:00 noon. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 9:15 a.m. to 10:45 a.m. and contributors of even-numbered papers will be at their posters from 10:45 a.m. to 12:00 noon. To allow for extended viewing time, posters will remain on display until 12:00 noon on Friday, 26 June.

4aPAc1. Theory on mass transport generated by ultrasonic Lamb waves. Gonghuan Du, Zhemin Zhu, Xiaoliang Zhao (Inst. of Acoust. and State Key Lab of Modern Acoust., Nanjing Univ., Nanjing 210093, PROC), and Junru Wu (Univ. of Vermont, Burlington, VT 05405)

It was reported that acoustic streaming generated by a Lamb-wave sensor (LWS) can be used to pump fluids and transport solids in a small-scale system [Appl. Phys. Lett. **59** (1991)]. The speed, observed by attracting small polystyrene spheres suspended in water, is proportional to the square of the wave amplitude and is about 100 μm/s for 6.5-nm surface displacement of the plate. In this paper, a fairly practical model, i.e., a thin isotropic plate in contact with a viscous liquid layer on its side, is presented. Based on the rigorous propagation theory of Lamb waves and Nyborg's [Phys. Acoust. B **2**, 265–295] and Bradley's [J. Acoust. Soc. Am. **100**, 1399–1408 (1996)] theories of acoustic streaming, the first-order ultrasonic field in the viscous liquid layer and the second-order mass-transport velocity distribution of the liquid are calculated. Comparison between our theoretical values of the Ao-mode Lamb wave and Moroney *et al.*'s experimental result is given. It shows good agreement. [Work supported by NSF of China.]

4aPAc2. Photoacoustic study of the electron–hole transport in a piezoelectric semiconductor. Weimin Gao, Viltali Gusev, Christ Glorieux, Kris Van Rostyne, Ward Van Looy, Jan Thoen (Lab. Akoestiek en Thermische Fysica, Dept. Natuurkunde, Katholieke Universiteit Leuven, Celestijnenlaan 200D, B-3001 Leuven, Belgium), and Gustaaf Borghs (IMEC, B-3001 Leuven, Belgium)

In a piezoelectric semiconductor acoustic waves can be excited via the inverse piezoelectric effect. This excitation process is the result of a screening of the applied electric field by the expanding space distribution of electrons and holes which are generated by pulsed laser radiation. The time and space characteristics of the electron–hole transport in this process are contained in the corresponding acoustic signals. The piezoelectric direct-gap semiconductors cadmium sulfur selenide and gallium arsenide are investigated experimentally at room temperature. A pulsed nanosecond UV excimer laser is used to excite electrons and holes on these samples. The generated longitudinal acoustic waves are detected by a laser interferometer. The obtained acoustic pulses of cadmium sulfur selenide show that both the expansion of the screened region in space and the electron–hole plasma expansion are supersonic at the time scale of laser action. The observations in this optoacoustic experiment indicate that the fast carrier transport is governed by photon recycling, i.e., the reabsorption of photons released by radiative recombination of electron–hole pairs. Experiments for gallium arsenide are in progress and will also be reported.

4aPAc3. Ultrasonic investigation of glass transition dynamics of polyurethane systems. Pierre-Yves Baillif, J. Mohamed Tabellout, and Jacques Emery (Gr. ultrason, UMR 6515 Chimie et Phys., des Materiaux Polymeres, Univ. du Maine, Av. O. Messiaen, 72035 Le Mans Cedex, France, ultrason@aviion.univ-lemans.fr)

Ultrasonic technique is an interesting tool to probe high-frequency dynamic properties of materials by determination of the mechanical modulus. Associated with other methods (rheology, DLS, etc.), it allows the investigation of molecular dynamics associated with sol–gel and glass transitions in polymers on a wide frequency range. Polypropylene glycol triol crosslinked with diisocyanate is a good model system for this purpose in polymer networks. By varying the amount of diisocyanate, stable systems are formed which correspond to different stages of polyurethane gel formation. Three triols with different molar masses (260, 700, and 6000 g/mole) are used as precursors in order to apprehend the influence of the macromolecular chain size on the systems dynamics. The temperature dependence of the ultrasonic absorption and velocity shows a relaxation process, the so-called alpha-relaxation associated with the glass transition. The influence of the stoichiometric ratio $r=[\mathrm{NCO}]/[\mathrm{OH}]$ on this transition is much less important in a large molecular mass precursor system, indicating that entanglement effects become predominant. The ultrasonic relaxation times associated with those obtained by other techniques are represented in an Arrhenius diagram and follow the same William–Landel–Ferry (WLF) law, showing the different systems dynamics' similarity. The WLF parameters seem, in that case, to have a character of universality.

4aPAc4. Longitudinal structural relaxation of glass-forming epoxy oligomer. Mami Matsukawa (Dept. of Electron., Doshisha Univ., Kyotanabe, Kyoto, 610-03 Japan) and Norikazu Ohtori (Niigata Univ., Niigata, 950-21 Japan)

The glass transition process of a common epoxy oligomer (diglycidyl ether of bisphenol A) has been characterized by Brillouin scattering (GHz range) and ultrasonic pulse spectroscopy (MHz range). As the temperature decreases, longitudinal wave velocity rapidly increased. It showed a large frequency dispersion at temperatures higher than the thermodynamical glass transition temperature (T_g). At T_g, velocity changes showed a clear kink. Wave attenuation also showed a peak of the main (alfa) relaxation at temperatures higher than T_g. The peak temperature shifted to lower temperatures in the MHz range, following the VFT type relaxation behavior, which is very common in the main relaxation. The velocities, however, still showed a clear frequency dispersion at temperatures lower than T_g. Because the main relaxation usually results from a large-scale conformational rearrangement of the polymer chain backbone, the effect of this relaxation seems to be neglected at these temperatures. Taking into ac-

4a THU. AM

count the frequency changes of attenuation peak widths, the effect of other subrelaxations at lower temperatures is discussed. These results are not in contradiction with the acoust-optic dispersion measurements of this oligomer [Kruger *et al.*, Phase Transition (to be published)].

4aPAc5. Hot-spot model of single-bubble sonoluminescence. Kyuichi Yasui (Dept. of Phys., Waseda Univ., 3-4-1 Ohkubo, Shinjuku, Tokyo, Japan)

The hot-spot model of single-bubble sonoluminescence (SBSL) is studied. The temperature is assumed to be spatially uniform inside the bubble except at the thermal boundary layer near the bubble wall even at the strong collapse based on the theoretical results by Kwak *et al.* [Phys. Rev. Lett. **77**, 4454 (1996)]. The effect of the kinetic energy of gases inside the bubble is taken into account, which heats up the whole bubble when gases stop their motions at the end of the strong collapse. A bubble in water containing air is assumed to consist mainly of argon based on the hypothesis by D. Lohse [D. Lohse *et al.*, Phys. Rev. Lett. **78**, 1359 (1997)]. Numerical calculations under an SBSL condition reveal that the kinetic energy of gases heats up the whole bubble considerably. It is also clarified that vapor molecules (H_2O) undergo chemical reactions in the heated interior of the bubble at the collapse and that chemical reactions decrease the temperature inside the bubble considerably. It is suggested that SBSL originates in quasithermal radiation from the whole bubble rather than a local point (the bubble center) heated by a converging spherical shock wave widely suggested in the previous theories of SBSL.

4aPAc6. Nonlinear reflection of acoustic waves at the dissipative isotropic solid–solid interfaces. W. H. Jiang, J. J. Chen, and Y. A. Shui (Inst. of Acoust. and State Key Lab. of Modern Acoust., Nanjing Univ., Nanjing 210093, PROC, whjiang@nju.edu.cn)

The nonlinear reflection at the solid–solid interfaces is investigated under the condition that a SV wave is incident from a nondissipative solid medium to a dissipative solid medium. The amplitude of the freely propagating reflected second harmonic SV wave is calculated. Comparing with previous calculation without consideration of dissipation, it is observed that the peaks of the reflected second harmonic SV wave in the vicinity of critical angles for refractive P and SV waves become smooth and the influence of nonlinear properties of the refractive medium on the amplitude of the reflected second harmonic wave is more notable. A experiment is carried out for glass–copper or rock interface. The SV wave of 8 MHz is incident from glass in which the attenuation of the acoustic wave is negligible upon glass–copper or rock interface. The reflected second harmonic SV wave is detected by a transducer with the working frequency of 16 MHz. The amplitudes of the second harmonic wave at the different incident angles are measured. The resultant angular dependence is in basic agreement with the calculation. [Work supported by NSF of China.]

4aPAc7. Estimation of forward-propagated ultrasonic fields in layered fluid media. K. L. Ha, M. J. Kim (Dept. of Phys., Pukyong Natl. Univ., 599-1 Daeyon-dong, Nam-gu, Pusan 608-737, Korea), W. Y. Zhang, S. G. Ye, and X. F. Gong (Inst. of Acoust., Nanjing Univ., Nanjing 210093, PROC)

The forward propagation of two-dimensional ultrasonic fields through layered fluid media was investigated theoretically by the angular spectrum method combined with the Rayleigh–Sommerfeld diffraction theory and experimentally by measurement of the fields using a precise scanning system with a needle-point hydrophone. Two types of transmitting transducers, one of which is a plane and the other of which is a concave type with the same frequency and the same aperture, were used. In theoretical analysis fields of incidence on a boundary of layer were calculated by the Rayleigh–Sommerfeld diffraction theory and those of transmission throughout the boundary were obtained by applying the angular spectrum method to the fields of incidence and considering the spatial-frequency-dependent transmission coefficients on each boundary. The optimum angular range and the spatial sampling interval could be obtained by supposing the virtual boundaries in the homogeneous medium of water and its appropriateness was confirmed experimentally. It was shown that the forward-propagated fields which could be estimated theoretically agreed reasonably with the measured data in two kinds of simple layered media with one or two boundaries.

4aPAc8. High temporal resolution of the acoustic transients associated with optical cavitation in water using a large area piezoelectric transducer. David C. Emmony and Robin D. Alcock (Dept. of Phys., Loughborough Univ., Loughborough, Leicestershire LE11 3TU, UK, d.c.emmony@lboro.ac.uk)

A PVDF piezoelectric transducer has been used to study the shock wave associated with a laser generated cavitation bubble in water. The measurements depend upon the accurate recording of the voltage developed in a large (3 mm) diameter but thin (9* m) piezoelectric film transducer placed close to a Nd YAG laser-generated cavity in water. The spherical shock transients excite the film over the duration of the passage of the wave through the transducer, and an invertion algorithm of the voltage trace gives the pressure profile along a radius. Pressure profiles due to initial breakdown and first collapse are in excellent agreement with those obtained using full field high-speed optical interferometry of the laser–liquid interaction.

4aPAc9. Particle interactions in coupled phase theory for sound propagation in concentrated emulsions. J. M. Evans and K. Attenborough (Faculty of Technol., The Open Univ., Milton Keynes MK7 6AA, UK, j.m.evans@open.ac.uk)

Theoretical studies of the acoustically induced interphase heat transfer in concentrated emulsions are reviewed. Interactions between the thermal fields around the particles influence the heat transfer and hence the complex wave number of propagating sound. Two modeling approaches are discussed: a hydrodynamic treatment and thermal wave scattering. These methods are used to calculate the interphase heat transfer at high concentrations. Modified heat transfer terms including the effect of particle interactions are obtained, which may be used in coupled phase theory. Predictions of the complex wave number using a revised coupled phase model [J. M. Evans and K. Attenborough, J. Acoust. Soc. Am. **102**, 278–282 (1997)] are compared with experimental data on ultrasound propagation in oil in water emulsions. The comparisons show the nondimensional frequency and volume fraction regions where interactions have a significant effect on the wave number.

4aPAc10. Particle trajectories in a drifting resonance field separation device. Bernhard Handl, Martin Gröschl, Felix Trampler, Ewald Benes (Institut für Allgemeine Physik, Vienna Univ. of Technol., Wiedner Hauptstr. 8/134, A-1040 Wien, Austria, benes@iap.tuwien.ac.at), Steven Woodside, and James Piret (Univ. of British Columbia, Vancouver, BC V6T 1Z3, Canada)

Drifting resonance field (DRF), a new method for the separation of particle/fluid suspensions is introduced. Unlike particle filters using ultrasound-induced coagulation, the DRF method only utilizes the primary acoustic radiation force to achieve the separation effect. This is done by cyclically switching the driving frequency of the DRF resonator between consecutive resonances, causing a directed motion of the particles and thus concentrating the particles in a certain region of the separation volume. Trajectories of polystyrene spheres (approximately 12 microns in diameter) suspended in distilled water have been measured in a prototype DRF resonator, varying the number of the utilized resonance frequencies and the acoustic power input. The measured particle trajectories are compared to computer simulations based on the same process parameters as used in the experimental setup, showing a high correspondence between experi-

mental and simulated data. Both experiments and simulations prove the high potential of the DRF method for separation technology. A promising field of application is the up-scaling of acoustic cell retention systems recently introduced in animal cell culture, potentially yielding higher flow-through rates. The presented results are a valuable basis for design and process parameter optimization of future DRF separation devices.

4aPAc11. Methods of processing laser Doppler anemometry signals to extract sound field information. John S. Cullen, David B. Hann, Clive A. Greated, and D. Murray Campbell (Dept. of Phys. and Astron., Univ. of Edinburgh, The Kings' Bldg., Mayfield Rd., Edinburgh EH9 3JZ, UK)

For several years laser Doppler anemometry (LDA) has been used to measure mean flow rates in gaseous and liquid flows. It is only recently that LDA has been developed as a technique for recording instantaneous particle velocities, enabling nonintrusive measurements to be made in sound fields. The paper describes a typical experimental setup and discusses various methods of digitally processing the LDA photodetector signals to extract the required velocity information. Particular emphasis is placed on the zero counting and Hilbert transformation techniques. The problem of signal dropout is also addressed. LDA measurements of acoustic particle velocity amplitudes are compared with acoustic velocity amplitudes deduced from probe microphone and impedance measurements. It is found that LDA measurements could be made not only in monofrequency sound fields but also in more complex sound fields containing a fundamental component and additional harmonics. Theoretical and practical limitations of the techniques are investigated and used to assess the viability of applying them to studies in the bores and mouthpieces of brass instruments.

4aPAc12. Approach to characterize the sound field of pulse-excited ultrasonic sensors using a Laser-Doppler vibrometer. Bernd Henning, Stefan Prange, Mark Schuart, and Karsten Dierks (Institut für Automation und Kommunikation e.V. Magdeburg, Steinfeldstrasse 3, D-39179 Magdeburg-Barleben, Germany)

A new efficient approach to make visible sound fields in front of acoustic sensors in liquids will be presented. The goal is to investigate the sound field characteristic of real manufactured acoustic transmitters particularly by discontinuous excitation like pulse, burst, and other. Another point is the analysis of the influence of materials and sensor design on the final acoustic behavior. In comparison to the Schlieren technique it is possible to use moderate excitation of transmitters (only a few volts) and more variable signal shapes too. The experimental setup consists of a Laser Doppler vibrometer and a high-resolution three-dimensional positioning system. The control of the positioning system as well as the data acquisition and processing will be fully automatic realized by a personal computer. A software tool archives the data and calculates the complex sound field characteristic in dependence on time or place. The result is an interesting animation representing the real sound field characteristic of acoustic sensors. The advantages and limits of the approach and developed experimental setup will be described. Some examples will show the sound fields in front of cylindrical focusing transmitters. The influence of different excitation signal shapes on the final sound field will be discussed.

4aPAc13. Simulation of ultrasound propagation in a thermally turbulent fluid using Gaussian beam summation and Fourier modes superposition techniques. Christian Lhuillier, David Fiorina, Jean-Luc Berton (Commissariat a l'Energie Atomique, Cadarache, DER/SSAE/LSMR, F-13108 Saint-Paul-lez-Durance cedex, France, ssaelsmr@macadam.cea.fr), and Daniel Juve (UMR CNRS 5509, Ecole Centrale de Lyon, F-69131 Ecully, France)

The CEA (French atomic agency) develops ultrasonic devices for the control and the inspection of Fast Breeder Reactors. The ultrasonic pulses propagate in a liquid which may be turbulent. To predict or interpret the effects of a homogeneous isotropic thermal turbulence, a numerical model has been implemented. Independent realizations of random thermal field are generated by superposition of random Fourier modes. For each realization the complex pressure field is calculated by the Gaussian beam summation method. Statistics over the set of realizations provide an estimation of the variance of time of flight and intensity fluctuations. Simulations have been performed with a point source. When the level of fluctuations is small, the model yields results which are in a good agreement with the analytical solutions given by Chernov and Rytov. When the level of fluctuations grows, it predicts correct evolutions. Multiple eigenrays are well predicted and calculations in time domain with a pulsed source show multiple transmitted signals or modifications in the signal shape.

4aPAc14. Lattice Boltzmann methods in acoustics. James M. Buick, D. Murray Campbell, and Clive A. Greated (Dept. of Phys. and Astron., Univ. of Edinburgh, Rm. 4201 J.C.M.B., The Kings' Bldg., Mayfield Rd., Edinburgh EH9 3JZ, UK)

The lattice Boltzmann model is a recently developed technique for the numerical simulation of fluid motion. It has been successfully applied to a variety of incompressible fluid phenomena and has been shown to be an efficient simulation tool which is ideally suited for implementation on massively parallel computers. Here the application of the lattice Boltzmann model is considered to simulate acoustical processes in which the fluid can, to a first approximation, be considered to be incompressible. The validity and limitations of this approach are considered and the results of simulations performed on the Cray T3D at Edinburgh University are presented. These include sound propagation in tubes, modeling of air flow in the mouthpiece of brass instruments, and the simulation of acoustic streaming in tubes and around cylinders. In each case the fluid velocity, density, and pressure are found at each grid point and each time-step of the simulation, allowing flow patterns and density changes to be observed. The results of the simulations are compared to the predictions of theory and/or experimental observations and suggest that the lattice Boltzmann method can be used as a numerical tool to give insight into complex acoustical situations.

4aPAc15. Acoustical characteristics of the noise radiated from supersonic multijets. Yoshikuni Umeda and Ryuji Ishii (Div. of Aeronautical and Astronautics, Dept. of Eng. Sci., Kyoto Univ., Yoshida Hon-Machi, Sakyo-ku, Kyoto, 606 Japan)

The acoustical characteristics of the noise radiated from combined jets exhausted from one main nozzle and 1, 2, 4, or 8 subnozzles which were placed asymmetrically or symmetrically to the main nozzle were investigated experimentally. The diameter d of the main nozzle was 5 mm and that of subnozzle a/d was 0.2 or 0.6, where a is the diameter of the subnozzle. The center-to-center spacing b of the main and subnozzles was fixed to 5 mm ($b/d=1.0$). In this experiment, the pressure ratio R of the jets was varied from 2.00 to 6.33. From the frequency characteristics of the screech tone, it is found that in almost all cases, two oscillation modes appear, but when eight subjets with $a/d=0.6$ were used, the screech tone disappeared completely. Although the total cross-sectional area of the multiple nozzles was larger than that of a single jet, it is found that the sound-pressure level radiated from multijets becomes the same or lower than that from the single jet. [Work supported by the Ministry of Education and Culture of Japan, Grant No. C2-09650187.]

4aPAc16. Acoustic parameters of high functional plastics. I. Measurement of acoustic velocities (wave propagation velocities). Masahide Gakumazawa and Minoru Akiyama (Dept. of Systems Eng., Shibaura Inst. of Technol., 307 Fukasaku, Ohmiya-shi, Saitama, 330 Japan)

Recently, high functional plastics (so-called engineering plastics) which improved the weak point of conventional plastics have been used in the industry. The acoustic parameters of these engineering plastics in the high frequency have been sparsely reported in the literature using a variety of experimental techniques. In this study the acoustic velocities (wave propagation velocities) of engineering plastics are measured using the scanning acoustic microscopy in the range from 50 to 200 MH, and the velocities were compared with the velocities reported in the literature in the lower frequencies.

THURSDAY MORNING, 25 JUNE 1998 FIFTH AVENUE ROOM (W), 8:00 TO 10:45 A.M.

Session 4aPP

Psychological and Physiological Acoustics: Plenary Preview—Issues Related to Speech Perception and Hearing Impairment

Jacek Smurzynski, Chair
Department of Otorhinolaryngology, University of Basel, Kantonsspital, Petersgraben 4, CH-4031 Basel, Switzerland

Contributed Papers

8:00

4aPP1. Effects of stimulus level and background noise on vowel representations in the auditory brain stem of cats. Bradford J. May (Dept. of Otolaryngol.—HNS, Johns Hopkins Univ., Baltimore, MD 21205) and Murray B. Sachs (Johns Hopkins Univ., Baltimore, MD 21205)

Effects of stimulus level and background noise on vowel representations in the ventral cochlear nucleus (VCN) were investigated by recording single-unit discharge rates in anesthetized cats. The quality of vowel encoding by populations of neurons can be evaluated by plotting vowel-driven rates as a function of each unit's best frequency (BF, the most sensitive frequency); the resulting rate profiles are assumed to show good representations of formant structure if there are high driven rates at BFs near formant frequencies and low rates at BFs near spectral troughs. Among the VCN unit types recorded in the present study, chopper units exhibited the most detectable formant-to-trough rate differences across vowel levels and in background noise. By contrast, primarylike units produced vowel representations that reflected the limited dynamic range of auditory-nerve fibers [May *et al.*, Aud. Neurosci. **3**, 135–162 (1996)]. Primarylike units with low spontaneous rates (SR$<$18 sp/s) showed better representations than high SR primarylike units, but failed to respond at low vowel levels. These results suggest that peripheral representations of speech sounds are sharpened by patterns of auditory nerve convergence in the cochlear nucleus. [Work supported by NIDCD Grant No. DC00109.]

8:15

4aPP2. The role of the auditory periphery in the categorization of stop consonants. Robert I. Damper (Dept. ECS, Univ. of Southampton, Southampton SO17 1BJ, UK, rid@ecs.soton,.ac.uk)

A proper understanding of speech perception requires an understanding of the staged restructuring of information that occurs as an acoustic stimulus is transformed by the auditory system into a phonetic percept. It is now firmly established that a physiologically faithful computer simulation of peripheral processing feeding into a trainable artificial neural network (ANN) is capable of reproducing the important aspects of the categorization of initial stop consonants into voiced and unvoiced classes. Correct behavior is found to be insensitive to the ANN architecture and a precise training scheme used, suggesting that such categorization is very basic to the repertoire of auditory processing strategies. Unlike a human or animal listener, a computational model can be easily manipulated and interrogated to discover the underlying basis of category formation. In particular, the physiological-based simulation of the auditory periphery can be dramatically simplified and the effect on categorization observed. It is found that proper modeling of frequency scaling and neural adaptation are essential to correct simulation of the well-known shift of phoneme boundary with place of articulation for stop consonants. The role of the auditory periphery in this case seems to be to emphasize the region of first formant information around the time of voicing onset.

8:30

4aPP3. Integrating monaural and binaural spectral information. Michael A. Akeroyd, Quentin Summerfield, and John R. Foster (MRC Inst. of Hearing Res., University Park, Nottingham NG7 2RD, UK, michael@ihr.mrc.ac.uk)

When speech is presented at a different lateral position to an interfering noise, analyses of both intensity and interaural differences can recover parts of the speech spectrum. It would thus be advantageous if listeners could integrate these two types of information. Their ability to do so using two-formant vowel-like stimuli was tested. A diotic white noise was modified by either increasing the intensity of two 1-ERB-wide bands by 6 dB (''monaural'' condition), by interaurally decorrelating those two bands (''binaural'' condition), or by increasing one band's intensity but interaurally decorrelating the other (''mixed'' condition). Four center frequencies were used—250, 650, 950, and 1850 Hz—and combined in pairs to create four vowels: 250–950, 250–1850, 650–950, and 650–1850. Both bands must be detected to distinguish these stimuli. Identification performance was significantly above chance in all conditions but best in the monaural condition. It was as good in the mixed condition as in the binaural condition. Performance in the mixed condition was better than predicted form performance when only a single band was presented, and thus it could not be due to listening to the bands individually. These results indicate that listeners can integrate monaural and binaural evidence of speech spectra.

8:45

4aPP4. Gap detection threshold in ears with and without spontaneous otoacoustic emissions (SOAEs). Jacek Smurzynski and Rudolf Probst (Dept. of Otorhinolaryngology, Univ. of Basel, Kantonsspital, CH-4031 Basel, Switzerland, smurzynski@ubaclu.unibas.ch)

Gap detection was measured for two groups of normally hearing young adults using broadband (0.1–12 kHz) noise stimuli presented monaurally through an insert earphone. Group I consisted of subjects who exhibited both strong SOAEs and click-evoked otoacoustic emissions (CEOAEs). Group II included individuals with no SOAEs in either ear and CEOAE levels that were <50th percentile of a laboratory normative database. An adaptive 2IFC procedure was used to determine the hearing threshold of the noise stimulus for each ear tested. Next, an adaptive 2IFC gap detection task was performed for the stimuli presented at either 10-, 20-, 30-, or 50-dB sensation level (SL). At the stimulus level of 10 dB SL, the ears in group I exhibited greater intersubject variability and higher mean gap detection thresholds than those in group II. The results at higher SLs did not show any differences between the two groups and were consistent with the data in the literature. A short gap in the broadband stimulus presented near the hearing threshold is masked by SOAEs and strong CEOAEs that are evoked by the stimulus. [Work supported by Swiss National Science Foundation.]

9:00

4aPP5. Modulation masking in a speech recognition task for normal-hearing and hearing-impaired subjects. Rene van der Horst (Dept. of Clinical and Experimental Audiol., Univ. of Amsterdam, Meibergdreef 9, Amsterdam, 1105 AZ, The Netherlands, r.vanderhorst@amc.uva.nl)

The effect of masking in the modulation domain was determined in a consonant recognition task. It is known from psychoacoustical experiments [T. Houtgast, J. Acoust. Soc. Am. **85**, 1676–1680 (1989)] that modulation detection can be masked by another modulation. An experiment was designed that tries to verify whether detection of the modulations in a speech signal is essential for speech recognition. The goal of this experiment was to determine whether stochastic modulations can mask the modulations in a speech signal and disturb speech recognition as effectively as filtering out speech modulations [R. Drullman *et al.*, J. Acoust. Soc. Am. **95**, 1053–1064 (1994)]. The masker used was a stochastic narrow-band signal with minimal inherent fluctuations. The masker signal was, in every critical band, multiplied with the speech modulations. Several masking conditions were created with different bandwidths and center frequencies. The center frequencies were all lying in the region of modulation frequencies which are important for speech (4–20 Hz). Normal-hearing subjects and subjects with sensorineural hearing loss participated in the experiment. The confusion matrices were analyzed by means of multidimensional scaling techniques. The results of the two groups of subjects will be compared with the results obtained with a psychoacoustic model.

9:15–9:30 Break

9:30

4aPP6. A mathematical model of consonant perception by cochlear implant users with the SPEAK strategy. Mario A. Svirsky and Ted A. Meyer (Dept. of Otolaryngol., Indiana Univ. School of Med., 702 Barnhill Dr., RR-044, Indianapolis, IN 46202)

The traditional approach to determine which psychophysical variables are important for speech perception by cochlear implant (CI) users has been to perform correlational analyses. However, the finding of a statistically significant correlation between psychophysical and speech perception variables does not explain the actual mechanisms CI users employ to identify speech sounds. A mathematical modeling approach is proposed that does provide a detailed description of how CI users identify speech sounds based on the acoustic information they receive. To use the proposed approach, it must be hypothesized what the relevant psychophysical dimensions are for a given stimulation strategy. In this study, it is hypothesized that users of the SPEAK stimulation strategy use three types of cues to identify consonants: first, the average cochlear locations stimulated in response to the first three formants; second, the silent gap duration (if present); and third, the ratio between total energy above and below 1000 Hz. Based on a subject's performance along these psychophysical dimensions, the model outputs a predicted consonant confusion matrix. Preliminary comparisons between model predictions and actual data indicate that the model was able to predict which pairs of consonant sounds would be most frequently confused by CI users. [Work supported by NIH.]

9:45

4aPP7. Categorical loudness perception in normal and hearing impaired subjects. Vishakha W. Rawool (Commun. Disord. and Special Education, Bloomsburg Univ., Bloomsburg, PA 17815)

This study compared categorical loudness perception in three subject groups: young (22–24 years) normal, older (43–60 years) normal, and older (60–80 years) hearing impaired (thresholds between 50 and 70 dB SPL at the test frequencies). Eleven subjects were included in each group. The subjects were asked to judge the loudness of warbled tones (2 and 4 kHz) presented at various sound-pressure levels in the following categories: not audible, very soft, soft, comfortable, loud, and very loud. The sound-pressure levels were calibrated for each individual ear. The multivariate analyses of variance was performed on the very soft, soft, comfortable, and loud variable. Average SPLs obtained in each category were used for the analyses. The results showed significant differences in the young normal and the hearing impaired groups in loudness perception across the four categories. No significant differences were apparent in the young normal and older normal groups. The older normal and the older hearing impaired groups differed across all the loudness categories except for the ''loud'' category where no significant differences were apparent. [Work supported by a Bloomsburg University grant for Research and Disciplinary Projects.]

10:00

4aPP8. Across-channel sensitivity to temporal asynchrony in cochlear implantees. Robert P. Carlyon (MRC Appl. Psych. Unit, 15 Chaucer Rd., Cambridge CB2 2EF, UK), Luc Geurts, and Jan Wouters (Univ. of Leuven, Leuven, Belgium)

Five post-lingually deafened users of the LAURAflex cochlear implant detected temporal asynchronies between two 400-ms, 100-Hz trains of 40-μs biphasic pulses, applied to two ''target'' electrode pairs. The pulses in the signal stimulus of each 2IFC trial were delayed by between 0.3 and 4.8 ms on either the basal or apical target, relative to those on the other target. The pulses in the standard stimuli were nearly synchronous, containing a 0.1 ms-delay to the electrode whose pulses were not delayed in the signal. All listeners performed substantially above chance over a wide range of delays. Four were more sensitive to basal delays than to apical delays; one showed the opposite trend. Presenting a ''masking'' 1000-Hz pulse train to electrode(s) between the two targets did not substantially affect the results. This finding, combined with the wide (6–10 mm) separation between the targets suggests that performance was not mediated by neurons responding to both target electrodes. Rather, it is argued that they represent a genuine sensitivity to differences in timing between discrete auditory channels.

10:15

4aPP9. The effect of fast-acting compression on psychoacoustic jnd's. Brent W. Edwards (ReSound Corp., 220 Saginaw Dr., Redwood City, CA 94063)

Fast-acting compression is used in hearing aids to compensate for loudness recruitment in patients with sensorineural hearing loss. Intensity difference limens are independent of the slope of the loudness function and it has been argued that compression normalizes loudness perception in hearing-impaired individuals at the expense of their intensity discrimination performance. Similar arguments have been made with regard to the effect of fast-acting compression on envelopes: Compression reduces the magnitude of envelope fluctuations and thus impairs the near-normal temporal processing capabilities of hearing-impaired individuals. The exact consequences of compression on these perceptual abilities are not clear, however, due to the level- and depth-dependent thresholds of intensity and modulation discrimination tasks, respectively. The effect of fast-acting compression will be quantified in terms of these psychophysical tasks and the results will be related to the speech recognition performance of hearing-impaired individuals.

10:30

4aPP10. Spectro-temporal modulation transfer and speech intelligibility for multiband amplitude compression. Joost M. Festen (Dept. of Otolaryngol., Univ. Hospital VU, P.O. Box 7057, 1007 MB Amsterdam, The Netherlands, jm.festen@azvu.nl)

For ears with sensorineural hearing loss the growth of loudness with stimulus level is often larger than in normal ears. This phenomenon is called recruitment. To restore normal loudness perception and to fit speech in the limited dynamic range of an impaired ear many modern hearing aids apply some form of multiband amplitude compression. However, within the cochlea nonlinearity and frequency selectivity are closely linked and therefore recruitment often goes together with reduced frequency selectivity. While compression may correct the recruitment, it does not improve the frequency selectivity. The opposite is even more correct; multichannel compression reduces the spectro-temporal variations in speech. Therefore in terms of speech intelligibility, the net effect of compression is difficult to predict. In an experiment speech intelligibility after multiband compression in quiet and in noise for 16 hearing-impaired listeners was measured. For each individual listener the speech spectrum was shaped and presented monaurally halfway in the ear's dynamic range in all conditions. Results show a decrease in intelligibility with both increasing compression ratio and increasing number of channels. These results can be fitted with a single curve when intelligibility is plotted as a function of the transfer of phase-locked modulations in time and frequenc.

THURSDAY MORNING, 25 JUNE 1998 GRAND BALLROOM A (S), 9:15 TO 11:20 A.M.

Session 4aSA

Structural Acoustics and Vibration: Statistical Energy and Fuzzy Structure Analyses—Similarities and Differences

Joseph W. Dickey, Chair
CNDE, Johns Hopkins University, 3400 North Charles Street, Baltimore, Maryland 21218-2689

Chair's Introduction—9:15

Invited Papers

9:20

4aSA1. Resonators attached to a structure—a SEA canon. R. H. Lyon (RH Lyon Corp, 691 Concord Ave., Cambridge, MA 02138-0799, rhlyon@lyoncorp.com)

One of the earliest problems analyzed using SEA ideas was a plate structure connected to a single resonator. Then structure-to-structure problems replaced the single resonator with a multiplicity of resonators as a model for the second structure. Because of the desire to relate the problem to those involving real structural connections—to beams, plates, and sound fields—the resonators became models for the natural modes of the structure and the resonators lost their identity. Fuzzy structure models have emphasized the resonators themselves, with passing attention to actual structural arrangements that they apply to. Of particular interest to SEA, however, is the relationship between modal density and ''mass density,'' and the relative strengths of internal and coupling damping, particularly as they relate to the transient response of the system. This presentation will discuss these matters.

9:55

4aSA2. Rudimentary statistical energy analysis and structural fuzzies. G. Maidanik (David Taylor Res. Ctr., Bethesda, MD 20084) and J. Dickey (Johns Hopkins Univ., Baltimore, MD 21218)

Structural fuzzies (Sfs) is a dynamic complex that is attached to a primary structure in order to provide a higher degree of dissipation. The dissipation is with respect to the power that is injected into the driven primary structure. It is argued that care needs to be exercised in the definition of the loss factor that is associated with this dissipation. This definition is examined and discussed; then, the role of the Sfs is analyzed in terms of a rudimentary statistical energy analysis (SEA). It is shown that the effectiveness of the dissipation is enhanced if the coupling of the Sfs to the primary structure is strong enough to approximate equipartitioning of modal energies, if the modal density in the Sfs either approximates or exceeds that in the primary structure and, finally, if the inherent loss factor in the Sfs is sufficiently higher than that in the primary structure. The definitions of strong enough coupling, modal densities, and inherent loss factors will be given. Examples of Sfs will be cited and discussed.

Contributed Papers

10:30

4aSA3. Spatial distribution of energy in SEA acoustic volumes. Evan B. Davis (Boeing Commercial Airplane Group, P.O. Box 3707 M/S 67-ML, Seattle, WA 98124-2207, evan.b.davis@boeing.com)

Statistical energy analysis (SEA) methods predict the space average energy in acoustic volumes. The predicted SEA acoustic energy levels are converted to SPL levels using the assumption that the sound field is diffuse. The SEA prediction methods estimate a single, space averaged value for the acoustic energy in the enclosure, whereas the theory of diffuse sound fields in enclosures predicts a nonuniform energy distribution in the enclosure. The energy levels and SPLs increase near the boundaries of the enclosure. The theory of diffuse sound fields can therefore be applied to generate an estimate of the spatial distribution of sound energy from the predicted spatial average. The sound field distributions are based on the Waterhouse corrections for reverberation rooms [Waterhouse, J. Acoust. Soc. Am. **27**, 247–258 (1955)]. The use of the Waterhouse correction in the interpretation of SEA predictions is most useful in acoustic volumes which are relatively small in relation to the acoustic wavelengths of interest. Enclosures which have dimensions on the same order as the acoustic wavelengths are commonly encountered in transportation systems such as automobiles and airplanes.

10:55

4aSA4. Fuzzy structure, initial problem. Poincare theorem. Samuil A. Rybak (N. N. Andreyev Acoust. Inst., Shvernik Str. 4, Moscow 117036, Russia)

A system consisting of a master structure—an oscillator, coupled with a number of oscillators, distributed in some frequency range—is considered. The master oscillator is initiated by a delta-function pulse, and damping is neglected. The energy flows from the master structure to the fuzzy system, the amplitude of its oscillations falls exponentially, and the amplitudes of fuzzy system oscillators are exponentially growing. This means that asymptotically the master structure becomes silent and the fuzzy oscillators have the maximum of their amplitudes. After the time T, depending on the full number of degrees of freedom, the picture almost fully returns back. But the time T becomes infinite when the number of oscillators is infinite (when their distribution is continual, for example). If at an arbitrary time one changes the signs of the velocities of the full system of oscillators, the phase trajectory turns back strongly the same way, because the system is conservative. The path tracing of the pole, which is originating in the integral describing the influence of the fuzzy system of oscillators on the master oscillator, is the same as usual, but the sign of the index of the exponential attenuation changes and the system goes to the initial state, with amplitude growing exponentially. [Work was partially supported by the Russian Foundation of Basic Research.]

THURSDAY MORNING, 25 JUNE 1998 GRAND BALLROOM I (W), 7:45 TO 10:45 A.M.

Session 4aSC

Speech Communication: Speech Perception Theories and Models (Poster Session)

Paul Iverson, Chair
House Ear Institute, 2100 West Third Street, Los Angeles, California 90057

Contributed Papers

All posters will be on display from 7:45 a.m. to 10:45 a.m. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 7:45 a.m. to 9:15 a.m. and contributors of even-numbered papers will be at their posters from 9:15 a.m. to 10:45 a.m. To allow for extended viewing time, posters will remain on display until 7:30 p.m. A cash bar will be set up in the poster area from 5:00 p.m. to 7:30 p.m. preceding the banquet.

4aSC1. Testing two information taxonomies in Spanish. Guillermo Andrés Toledo (Dépt. de langues et linguistique, Faculté des lettres, Univ. Laval, Cité Univ., Québec, QC G1K 7P4, Canada, ucineacad@sminter.cam.ar)

Two information taxonomies, i.e., in the first experiment, New/Given distinction and, in the second experiment, three cognitive categories of the mental representation of referents: activation, semiactivation, inactivation differences [W. Chafe, in Tomlin *Coherence and Grounding in Discourse* (John Benjamins, Amsterdam, 1987), pp. 21–51 and K. Lambrecht, *Information Structure and Sentence Form* (Cambridge U.P., Great Britain, 1994), pp. 1–1161 have been explored in Argentinian Spanish. In the first experiment, semispontaneous materials were recorded from communicative interactions between three male speakers, one female speaker, and the listener. Brand-New plus New (Inferrable) versus Given (Textually Evoked) items were selected for acoustic analysis. In the second experiment, activation, semiactivation, and inactivation states encoded in semispontaneous discourses were taken from interactions between three male speakers and the listener. In both experiments, tonal prominences were studied through intonational contours on pairs of contrasted items: natural values and data transformed through logarithmic z-score normalization. Results in the first experiment indicated a relevant tonal encoding in new items. Findings in the second experiment have shown higher prominences correlated with inactive states, lower prominences with active and semiactive states.

4aSC2. Perception of speech gestures. René Carré (ENST, Dept. Signal, 46 rue Barrault, 75634 Paris Cedex 13, France) and Pierre L. Divenyi (Exp. Audiol. Res., V. A. Med. Ctr., Martinez, CA 94553)

Perception of a synthetic French /aya/ sequence, generated by coproduction of two simultaneous gestures (tongue constriction and lip rounding), was tested in a series of experiments. The tokens consisted of eight different degrees of gesture reduction in which the target /y/ was not reached. The /ay/ trajectory in the plane $F1$–$F2$ was not rectilinear (it pointed first to /i/ before reaching /y/), having to do with the fact that the acoustic effect caused by lip rounding (lowering $F2$) becomes manifest later than the effect due to constriction (lowering $F1$ and raising $F2$). The /aya/ transition is, therefore, an appropriate stimulus to study the role of the linearity of trajectories in speech perception. Results show that unless /y/ is present as a steady-state vowel for at least 30 ms, the lip rounding gesture is ignored and the percept is /aia/. Thus the results strongly suggest

that formant trajectories are perceived as time averages through a mechanism that calculates the direction of transition toward an (unreached) target from a running average of the direction of the formant trajectory. These findings point to the importance of the trajectory direction in vowel reduction and of the time dimension in phonetic distinctions. [Work supported by NATO.]

4aSC3. Coarticulation in CV syllables: A locus equation and EPG perspective. Marija Tabain (Speech, Hearing and Lang. Res. Ctr., Dept. of Linguist., Macquarie Univ., Sydney, Australia 2109)

Stop, nasal, and fricative consonants of Australian English were paired with a wide variety of vowel monophthongs to form CV syllables which were placed in a carrier phrase. The EPG and acoustic data were obtained from four female speakers producing these phrases. Locus equations [Lindblom, Acoust. Soc. Am. **35**, 1773–1781 (1963)] were generated from the acoustic data, and coarticulation indices [Farnetani, *Speech Production and Speech Modelling*, edited by Marchal and Hardcastle (1990), pp. 93–130] were calculated from the EPG data. The coarticulation indices were compared with the slope values from the locus equations [Krull, PERILUS **10**, 87–108 (1989)] in order to determine the degree of coarticulatory resistance for a given consonant phoneme as measured in the articulatory and acoustic domains. Results suggest that the greatest degree of coarticulation is shown in the labial and velar consonants, and less in the apico-alveolar consonants. The least degree of coarticulation is shown by the lamino-postalveolar consonants /S Z/. These results agree with data from Australian Aboriginal languages and with results presented by Recasens for Catalan [J. Phon. **12**, 61–73 (1984); **19**, 177–192 (1991)].

4aSC4. A model for dependences in phonetic categorization. Roel Smits (Dept. of Phonet. and Linguist., Univ. College London, 4 Stephenson Way, London NW1 2HE, UK)

A quantitative model for human categorization behavior is proposed which can be applied to four-alternative forced-choice categorization data involving two binary classifications. A number of processing dependences between the two classifications are explicitly formulated, such as the dependence of the location, orientation, and steepness of the class boundary for one classification on the outcome of the other classification. The significance of various types of dependences can be tested statistically. Analyses of two data sets from the literature [Repp *et al.*, J. Exp. Psych: Human Percept. Perform. **4**, 621–637 (1978); Whalen, Percept. Psychophys. **46**, 284–292 (1989)] show that interesting dependences in human speech recognition can be uncovered using the model.

4aSC5. Familiarity and pronounceability of nouns and names. Aimée M. Surprenant (Dept. of Psychol. Sci., Purdue Univ., West Lafayette, IN 47907-1364, aimee@psych.purdue.edu), Susan L. Hura (Lucent Technologies, Holmdel, NJ 07733), Mary P. Harper, Leah H. Jamieson, Stephen A. Hockema, Tsung-Hsiang Hsueh, Michael T. Johnson, Pramila N. Srinivasan, Scott M. Thede, Christopher M. White, Glenis Long, and Ayasakanta Rout (Purdue Univ., West Lafayette, IN 47907-1353)

Proper names have several properties that create problems for speech recognition systems: the number of names is large and ever changing, names can be borrowed directly from other languages and may not conform to usual pronunciation rules, and the variety of pronunciations for names can be high. Because the set of proper names is so dynamic and machines are notoriously poor at phoneme recognition, a promising approach to designing a name recognition system is to incorporate statistical aspects of proper names (e.g., frequency, familiarity). Unfortunately, there exists relatively little data on the distribution of names. Ratings of familiarity and pronounceability were obtained for a randomly chosen sample of 199 surnames (from 80 987 entries in the Purdue phonebook) and 199 nouns (from Kucera–Francis). The ratings for nouns versus names are substantially different: nouns were rated as more familiar and easier to pronounce than surnames. Frequency and familiarity were more closely related in the proper name pool than the word pool, although the correlations were modest. Ratings of familiarity and pronounceability were highly related for both groups. The value of using frequency and the ratings of familiarity and pronounceability for predicting variations in actual pronunciations of words and names will be discussed.

4aSC6. Ranking the pitches of concurrent vowels. D. Dwayne Paschall (Texas Tech Univ. Health Sci. Ctr., Box 42073, Lubbock, TX 79409) and Peter F. Assmann (Univ. of Texas at Dallas, Richardson, TX 75083-0688)

When two vowels are presented simultaneously, listeners can label them more accurately if their fundamental frequencies differ, and under some conditions they can report which vowel has the higher pitch. In this study the interaction of fundamental frequency and binaural cues on judgments of the relative pitches of concurrent vowel pairs is examined. On each trial listeners were told which vowels were presented and were asked to indicate which vowel was on the higher pitch. The first vowel had an F_0 of 140 Hz and the second vowel was either 1, 2, 4, or 8 semitones lower or higher than the first. When the fundamental frequency difference was small (1 semitone), the ability to rank the pitches of the two vowels was near chance level, and dichotic presentation provided little advantage over monaural presentation. With monaural presentation performance improved gradually with increasing F_0 difference, reaching a maximum near 80% correct at $+8$ semitones. When the stimuli were presented dichotically there was a dramatic improvement between 1 and 2 semitones and performance was near 100% with larger F_0 differences of 4 and 8 semitones. The interaction of F_0 differences and binaural cues places important constraints on models of concurrent vowel segregation.

4aSC7. Selection of a tonotopic scale for vowels. Terrance M. Nearey (Dept. of Linguist., Univ. of Alberta, Edmonton, AB T6G 2E7, Canada)

Various distinct tonotopic scales (e.g., log frequency, ERB, Bark) have been used for the representation of fundamental and formant frequency data for vowels. In previous work in our laboratories [T. Nearey, Proc. ICSLP92, 583–586 (1992)], generalized linear modeling was used to assess the relative appropriateness of these scales on both production measurement and vowel categorization data on each scale using several alternative vowel normalization schemes. That work focused on F_0 and F_1 and used the classic data of Peterson and Barney [G. Peterson and H. Barney, J. Acoust. Soc. Am. **24**, 175–184 (1952)]. Analysis of that data suggested that the log scale as provided somewhat better fits to the data with less evidence of violations of modeling assumptions. However, results in modeling perceptual data (from Canadian English listeners) were less clear. The present study will extend the scope of the earlier work by examining more data and expanding the statistical modeling framework. The production analyses will be augmented using an additional large production data set for another dialect of American English [J. Hillenbrand *et al.*, J. Acoust. Soc. Am. **97**, 3099–3111 (1995)]. The perceptual analyses will be augmented using additional perceptual data sets collected in our laboratories.

4aSC8. An exemplar-based account of emergent phonetic categories. Francisco Lacerda (Stockholm Univ., Inst. of Linguist., 106 91 Stockholm, Sweden)

This paper describes how an exemplar-based model can be used to study the early stages of language acquisition and phonetic category formation. The paper discusses in particular how a (conceptually) very simple model, inspired by Edelmans notion of Neural Darwinism, may account for the emergence of phonetic categories arising from the interaction between ambient language input and limited memory-representation resources. It presents an explicit attempt to demonstrate how phonetic categorization may emerge ''spontaneously'' from exposure to natural infant-directed speech data. The results suggest that phonetic categories may emerge if the auditory input is statistically correlated with one (or several) other sensory inputs. The perceptual-magnet effect is also discussed in the light of the present model. According to the model, the perceptual-magnet effect can be seen as a consequence of distributed memory representations. [Work supported by The Bank of Sweden Tercentenary Foundation, grant 94-0435.]

4aSC9. Effect of spectral distance on vowel height perception. Patrick C. M. Wong and Randy L. Diehl (Dept. of Psych., Univ. of Texas at Austin, Austin, TX 78712, pcmwong@ccwf.cc.utexas.edu)

Perceived vowel height has been reported to vary inversely with the distance (in Bark) between the first formant frequency ($F1$) and the fundamental frequency ($F0$) [H. Traunmueller, J. Acoust. Soc. Am. **69**, 1465–1475 (1981)]. However, in a study using back vowels, Fahey *et al.* [J. Acoust. Soc. Am. **99**, 2350–2357 (1996)] found that phonetic quality was not only determined by the $F1-F0$ distance, but also by the tonotopic distances between *any* adjacent spectral peaks (e.g., $F3-F2$, $F2-F1$, and $F1-F0$), with greater weight accorded to smaller distances. The present study further tested this possibility by using front vowels. Listeners identified three sets of synthetic vowels varying orthogonally in $F1$ and $F0$ and ranging from /i/–/ɪ/, /ɪ/–/ɛ/, or /ɛ/–/æ/. The results allowed this possibility to be rejected. Furthermore, they supported the study of Hoemeke and Diehl [J. Acoust. Soc. Am. **96**, 661–674 (1994)] that $F1-F0$ was the best predictor of perceived vowel height for the phonological distinction [+/−high] (i.e., the /ɪ/–/ɛ/ distinction), while $F1$ alone was the best predictor for the two other vowel set distinctions. [Work supported by NIDCD.]

4aSC10. Syllable as the segmentation unit in perceiving spoken Chinese. Chin-Hsing Tseng (Dept. of Special Education, Natl. Kaohsiung Normal Univ., Kaohsiung, Taiwan)

How spoken Chinese is segmented by native speakers has been addressed by the author in recent years. Evidence has accumulated to support the view that the syllable is the level at which speakers segment their spoken language. Earlier studies found that Mandarin–Chinese phonemes were detected faster in real syllables than in nonexistent syllables, that they were faster for level-tone and falling-tone syllables than for rising-tone or dipping-tone syllables, and that structural complexity had no effect on phoneme detection. These effects have preempted phoneme and word as the segmentation unit in real-time processing. This study was designed to investigate the role of stress foot in segmentation, to discover the locus of the tone effect, and to replicate the tone and complexity effects with Taiwanese speech materials. Stress assignment and phoneme detection tasks were employed as the experimental paradigms. The results indicated that stress assignment in Mandarin is not consistent enough to warrant a stress-foot segmentation procedure, that the tone effect originates from syllable duration variations, and that earlier effects were replicated with Taiwanese.

4aSC11. Phonological boundaries and the spectral center of gravity. Michelle R. Molis, Randy L. Diehl, and Adam Jacks (Dept. of Psychol., Univ. of Texas, Austin, TX 78712, molis@mail.utexas.edu)

A critical limit of 3–3.5 Bark has been reported for the ''spectral center of gravity'' effect between the second and third formants ($F2$ and $F3$) and is assumed to correspond to a perceptually natural boundary [A. K. Syrdal and H. S. Gopal, J. Acoust. Soc. Am. **79**, 1086–1100 (1986)]. This hypothesis was tested for both the [+/−back] and [+/−coronal] distinctions in English vowels. Subjects identified two sets of three-formant synthetic vowels which varied orthogonally in $F2$ and $F3$ and ranged between /ʊ/–/ɪ/ ([+/−back]) or /ɝ/–/ʊ/ ([+/−coronal]). For the /ʊ/–/ɪ/ distinction, a boundary shift was observed solely as a function of $F2$, ruling out an invariant $F3-F2$ boundary. For the /ɝ/–/ʊ/ series, both $F2$ and $F3$ influenced boundary location. There was a relatively stable $F3-F2$ boundary for the mean identification responses in this case, but it occurred at less than the predicted 3–3.5 Bark difference. Follow-up results will also be presented. [Work supported by NIDCD.]

4aSC12. An exemplar-based model of silent-center syllable perception. Stephen Winters and Keith Johnson (Dept. of Linguist., Ohio State Univ., 222 Oxley, 1712 Neil Ave., Columbus, OH 43202)

Rakerd and Verbrugge [J. Memory Lang. **26**, 558–563 (1987)] suggested that the perception of silent-center syllables—CVC syllables with the middle 60% of the vowel edited out and replaced with silence—had to involve some form of speaker normalization. Crucial to their argument was their subjects' ability to perceive cross-gender hybrids of such syllables. Rakerd and Verbrugge reasoned that even modest accuracy in perceiving these hybrids entailed some sort of normalization of the disparate acoustic qualities of male and female voices in order to derive one perceptual whole. Rakerd and Verbrugge's study has been replicated in order to test the ability of a computationally implemented exemplar model to perceive silent-center syllables. Exemplar-based modeling utilizes no ''speaker normalization'' as such, but, nonetheless, the model in this study approximated the performance of human subjects in perceiving silent-center syllables correctly. The model's performance on the important ''hybrid'' case was almost identical to subjects' responses. These results suggest that the human ability to perceive silent-center syllables does not necessarily entail some process of speaker normalization in speech recognition. [This material is based upon work supported under a National Science Foundation Graduate Fellowship and NIDCD Grant No. R29-DC01645-06.]

4aSC13. The time course of learning: Neurophysiologic changes during speech training. Kelly L. Tremblay, Nina Kraus, Therese McGee, and Steve Zecker (Dept. of Commun. Sci. and Disord., Northwestern Univ., 2299 North Campus Dr., Evanston, IL 80208)

The ability of normal-hearing adults to identify and discriminate novel speech sounds improves with training. Neurophysiological changes associated with this improvement include increased duration and area, as well as shortened onset latency, of the mismatch negativity (MMN) [N. Kraus *et al.*, J. Cog. Neurosci. **7**, 25–32 (1995); K. Tremblay *et al.*, J. Acoust. Soc. Am. **102**, 1–12 (1997)]. The MMN is a passively elicited, preattentive, auditory event-related potential that reflects auditory discrimination and echoic memory [R. Naatanen *et al.*, Acta Psychol. **42**, 313–329 (1978)]. To date, little is known about the time course of neurophysiological change and the manifestation of learning. The present study examined the time course of perceptual learning and its relation to changes in the MMN with the expectation that cortically evoked potentials could provide insight into the biological processes underlying perceptual learning and serve as a clinical tool for evaluating change during efforts of (re)habilitation in the communicatively impaired. Specifically, it was found that neurophysiological change is apparent in the MMN preceding behavioral improvement. This suggests that auditory training alters the neural activity that provides the necessary coding of the speech stimuli and those changes are later integrated into functional behavior.

4aSC14. Context effects in the auditory identification of Spanish fricatives /f/ and /θ/: Hyper and Hypospeech. Sergio Feijóo, Santiago Fernández, and Ramón Balsa (Dpto. Física Aplicada, Fac. de Física, 15706, Santiago, Spain, fasergio@uscmail.usc.es)

Twenty-eight subjects heard 40 Spanish words in which the initial fricative was /f/ or /θ/, combined with vowels /e/ and /u/. Ten words were used for each particular combination (2 fricatives$\times$2 vowels $\times$10 words). Two forms of speech (Hypo and Hyperspeech) and four conditions were considered: (1) Isolated fricative segment; (2) fricative segment + 51.2 ms of the following vowel; (3) fricative + whole following vowel; (4) whole word. The statistical analysis showed that, despite their differences in production and acoustic characteristics, isolated fricative segments were equally recognized in Hypo and Hyperspeech (cond. 1). Including the vowel (conditions 2 and 3) significantly improved recognition of both fricatives for both forms of speech, except for the combination /f/+/e/: While fricative identification improves slightly in Hyperspeech, in Hypospeech, recognition decreases with respect to cond. (1). For this particular combination, an acceptable recognition rate is only achieved in the whole word condition for both forms of speech. The results indicate that: (a) Clear speech is not necessarily more intelligible than hypospeech; and (b) The role of vocalic context in the recognition of /f/ and /θ/ is similar for both forms of speech.

4aSC15. Evidence of independent verbal processors for the same stimulus: Insights from dichotic verbal transformations. James A. Bashford, Jr. and Richard M. Warren (Dept. of Psych., Univ. of Wisconsin, Milwaukee, WI 53201, bashford@csd.uwm.edu)

A clearly enunciated word heard repeating without change undergoes illusory verbal transformations (VTs) to different words or syllables. Surprisingly, when a repeating word is delivered dichotically with an interaural asynchrony of half its duration, the separate lateralized images of the word are heard to undergo independent VTs (e.g., ‘‘commence’’ heard at one ear while ‘‘tress’’ is heard at the other) [see Warren and Ackroff, Nature **259**, 475–477 (1976)]. The present study eliminates a peripheral explanation attributing such independent changes to lateral asymmetries in repetition effects upon the lower auditory pathways, which might provide conflicting acoustic-phonetic information to a single verbal processor. A dichotic alternation paradigm, in which successive statements of a repeating word were presented to opposite ears (e.g., ‘‘flame’’ presented left, then ‘‘flame’’ presented right . . .) was also found to produce independent VTs. However, when higher amplitude noise was presented to the ear opposite the alternating speech signal, contralateral induction [Warren and Bashford, Percept. Psychophys. **20**, 380–386 (1976)] produced diffuse medial localization of the otherwise lateralized signals, and despite monaural verbal stimulation, VTs were linked across ears. This and other evidence to be discussed demonstrate independent preattentive verbal processors for spatially distinct sources. [Work supported by NIH.]

4aSC16. Inducing a ‘‘perceptual magnet’’-like effect in a nonspeech modality. Fatima Husain and Frank H. Guenther (Dept. of Cognit. and Neural Systems, Boston Univ., 677 Beacon St., Boston, MA 02215, fhusain@cns.bu.edu)

Kuhl and colleagues have described a ‘‘perceptual magnet effect’’ in which subjects show a decreased ability to discriminate between vowel-like stimuli when these stimuli fall near a prototypical vowel from their native language. Guenther and Gjaja [J. Acoust. Soc. Am. **100**, 1111–1121 (1996)] proposed that the effect results from experience-based map formation in auditory brain regions. Because this model attributes the effect to neural mechanisms that have been identified in cortical areas subserving different sensory modalities, it predicts that it should be possible to induce magnet effects in nonspeech modalities. To test this, adult subjects' discrimination sensitivities to narrow-band noise stimuli were tested before and after training sessions meant to approximate an infant's exposure to speech sounds. Training consisted of a categorization task that preferentially exposed subjects to stimuli from a small frequency range meant to correspond to a prototypical vowel. Sensitivity tests after training revealed that subjects had become worse at discriminating sounds within this ‘‘prototype’’ range as compared to a ‘‘nonprototype’’ range that was absent during training. This suggests that the magnet effect may not depend on phonetic processing and may reflect neural mechanisms common to different sensory modalities. [Supported by NIDCD and the Sloan Foundation.]

4aSC17. Interarticulator phasing and locus equations. Anders Lofqvist (Haskins Labs., 270 Crown St., New Haven, CT 06511, lofquist@haskins.yale.edu)

A locus equation plots the frequency of the second formant at vowel onset against the target frequency of the same formant for the vowel in a consonant–vowel sequence. The slope of the equation has been assumed to reflect the degree of coarticulation between the consonant and the vowel, with higher slopes associated with more coarticulation. This study examined the articulatory basis for this assumption, using VCV sequences where the consonant was a bilabial stop /p, b/ and the vowels one of /i, a, u/. Articulatory movements were recorded using a magnetometer system. Four subjects participated and produced ten repetitions of each sequence. One articulatory measure was the temporal phasing between the onset of the lip closing movement and the onset of the tongue body movement from the first to the second vowel. Another was the magnitude of the tongue movement from the onset of the second vowel to the tongue position for the vowel, averaged across four receivers placed on the tongue. When compared with the corresponding locus equation slopes, neither measure showed support for the assumption that the slope serves as an index of the degree of coarticulation between the consonant and the vowel. [Work supported by NIH.]

4aSC18. Latency of MEG M100 response indexes first formant frequency. Krishna K. Govindarajan (MIT, Cambridge, MA 02139), Colin Phillips (Univ. of Delaware, Newark, DE 19716), David Poeppel, Timothy P. L. Roberts (UCSF, San Francisco, CA 94143), and Alec Marantz (MIT, Cambridge, MA 02139)

Magnetoencephalography recordings from the auditory cortex of subjects listening to synthetic vowels show a close correlation between the timing of the evoked M100 response and the first formant frequency ($F1$). These results are consistent with evoked magnetic field latencies elicited by tone stimuli, which show 100 to 300-Hz tones associated with latencies up to 30 ms longer than 500 to 3000-Hz tones. In experiment 1, three-formant vowels /u,a,i/ were presented at two fundamental frequencies ($F0=100$ Hz, 200 Hz). The M100 latency was a function of the vowel identity and not $F0$: M100 was significantly shorter for /a/ than /u/, consistent with the spectral center of gravity being at a higher frequency for /a/. In experiment 2, single-formant vowels (/u/:$F1=330$ Hz, /a/:$F1=720$ Hz) were covaried with two $F0$ values. M100 latencies were shorter for /a/ (high $F1$) than for /u/ (low $F1$), at both fundamentals. In experiment 3, subjects listened to pure-tone complexes with frequencies and amplitudes matching the $F0$ and $F1$ energy peaks of the stimuli in experiment 2. M100 latencies showed the same pattern: latency covaried with the energy

peak corresponding to $F1$, suggesting that the sensitivity to the energy in the $F1$ range is not speech-specific. Finally, an experiment involving two-formant, amplitude-varying vowels will be presented.

4aSC19. Perceptual magnet effect in corner vowels. Robert Allen Fox and Lynn Carahaly (Dept. of Speech and Hearing Sci., Ohio State Univ., Columbus, OH 43210)

In previous studies of speech perception, Kuhl and colleagues have provided evidence that the category goodness of a vowel influences the perception of that particular vowel—the perceptual magnet effect. These data indicate that listeners demonstrate poor discrimination of vowel tokens when they are close to the prototype vowel but good discrimination when the vowel is close to a nonprototypic vowel. However, data will be provided from experiments conducted in this laboratory using an MDS paradigm on the /i/–/ɪ/ distinction that indicate this effect is eliminated or severely curtailed when all tokens are good exemplars of the /i/ category ($>75\%$) and the nonprototypic exemplars are not near a phoneme boundary. In addition, given the almost exclusive reliance in the literature on the /i/–/ɪ/ vowel pair, additional data will be provided on the magnet effect in the /u/–/ʊ/ and /ʌ/–/ɑ/. Similar to /i/, extreme tokens of both /ɑ/ and /u/ that are distant from the prototype, but not in the direction of another vowel category (and thus near a phoneme boundary). The results will be discussed in terms of whether the effect is linguistic (phonetic) or auditory in nature.

4aSC20. Surveying auditory space using vowel formant data. Matthew J. Makashay and Keith Johnson (Dept. of Linguist., Ohio State Univ., 222 Oxley Hall, 1712 Neil Ave., Columbus, OH 43210, makashay@ling.ohio-state.edu)

Recent research has suggested that vowel category learning and the perceptual magnet effect are the natural consequences of auditory neural map formation [F. H. Guenther and M. N. Gjaja, J. Acoust. Soc. Am. **100**, 1111–1121 (1996)]. As these effects are language specific, the organization of the perceptual map crucially is dependent on the input received during learning. Guenther and Gjaja implemented a working model of a self-organizing neural map to show that, given input from Gaussian distributions around vowel category centers, unsupervised training leads to a warping of the perceptual space toward the category centers. This work replicates Guenther and Gjaja's results and provides a more realistic test of the model by using real vowel formant data instead of idealized distributions. Single and multiple male talker vowel data were used to train auditory neural maps. Single-talker simulations resulted in map organization similar to Guenther and Gjaja's results. However, no clear vowel category clusters emerged in the multiple-talker simulations. [Work supported by NSF Graduate Fellowship and NIH FIRST Award R29-DC01645-06.]

4aSC21. Relation between discrimination and identification of English vowels. Diane Kewley-Port and Amy T. Neel (Dept. of Speech and Hearing Sci., Indiana Univ., Bloomington, IN 47405)

Although resolution for formant frequency is considerably better than needed for vowel identification in English, adjacent vowels in the $F1$-by-$F2$ plane may not be equally discriminable. Kuhl has reported better discrimination for poor vowel exemplars than good exemplars (''perceptual magnet effect'') for a few vowels in high stimulus uncertainty tasks. The purpose of this experiment was to determine the relation between discrimination thresholds and the judgment of vowels as good, confusable, or non-English. A large $F1$-by-$F2$ vowel space (190 tokens) encompassing five English front vowels was synthesized in equal bark steps. Based on an identification task with goodness ratings, two vowels representing each goodness category were selected for the discrimination task. Eight listeners participated in a minimal uncertainty discrimination task to estimate thresholds for $F1$ and $F2$. Analysis of variance of the 12 thresholds (converted to Δ Barks) found that thresholds for non-English vowels were significantly poorer than the other two categories, whereas thresholds for good and confusable vowels were similar. These results are not consistent with the perceptual magnet effect. Vowels within areas of the $F1$-by-$F2$ space judged as good exemplars are not discriminated more poorly than those in confusable areas. [Work supported by NIH-NIDCD-02229.]

4aSC22. The effect of vowel prototypicality and extremity on discrimination sensitivity. Satsuki Nakai (Univ. of Edinburgh, Edinburgh EH8 9LL, UK, satsuki@ling.ed.ac.uk)

The present study examines the effect of vowel extremity and prototypicality on discrimination sensitivity. Kuhl [1991] has shown that listeners are not as capable of distinguishing other variations of /i/ from a prototypical /i/ (the stimulus with the highest goodness rating, henceforth P) as from a nonprototypical /i/ (a stimulus with a low goodness rating). However, considering evidence suggesting that discrimination sensitivity grows poorer towards the exterior of the vowel space [Schouten and van Hessen, 1992], it is unclear whether it is vowel prototypicality or extremity that correlates with low discrimination sensitivity, for P is the most extreme among the stimuli used in the discrimination task in Kuhl [1991]. In order to investigate the above issue, discrimination curves were obtained from native speakers of Japanese and Greek in the corner of the vowel space where Japanese /u/ and Greek /u/ are located. As Japanese /u/ is not cardinal, the effect of vowel prototypicality and extremity on discrimination sensitivity could be observed separately on Japanese subjects' discrimination curves. The results indicate that both vowel prototypicality and extremity contribute to low discrimination sensitivity, and that the extremity of P correlates with the magnitude of its effect on discrimination sensitivity.

4aSC23. Phonetic categories: Internal category structure and processing speed. Joanne L. Miller (Dept. of Psych., Northeastern Univ., Boston, MA 02115), Peter D. Elimas (Brown Univ., Providence, RI 02912), and Ethan Cox (Univ. of Arizona, Tucson, AZ 85721)

Phonetic categories have a graded internal structure, with some members perceived as better exemplars than others. In a project focusing on voiceless stop consonants, the relation between this structure and processing speed has been examined. Speech continua were constructed in which the voice-onset-time (VOT) value of a word-initial consonant systematically changed from very short to very long. Unspeeded identification tests confirmed that stimuli with very short VOT values are perceived as voiced, whereas those with longer VOT values are perceived as voiceless. Category-goodness rating tests further confirmed that only stimuli within a limit range along each continuum are perceived as good exemplars of the voiceless category; stimuli with shorter or longer VOT values are perceived as poorer exemplars. Initial results from an ongoing experiment involving speeded identification indicate that (1) compared to the best exemplars of the voiceless category, identification of poor exemplars in the voiced–voiceless boundary region is relatively slow, but (2) across a wide range of values, poor exemplars with VOT values longer than the best exemplars are identified as quickly as the best exemplars themselves. These initial findings indicate that internal category structure is not necessarily reflected in processing speed. [Work supported by NIH.]

4aSC24. The effect of speaking style alterations on locus equation stability. Harvey M. Sussman, Eileen Dalston, and Sam Gumpert (Dept. of Commun. Sci. and Disord., Univ. of Texas, Austin, TX 78712, sussman@mail.utexas.edu)

Locus equations, as a phonetic descriptor of stop place of production, have been systematically investigated across speakers, gender, languages, perturbation conditions, syllable position, and manner class. As locus equation slope is thought to directly reflect the extent of anticipatory coarticulation of the vowel on the prevocalic consonant, changes in coarticulation as brought about by altered speaking styles can effectively alter locus equation descriptors of stop place. This study attempted to assess the stability of locus equations within each speaker as citation-style hyperarticulation was compared relative to spontaneous, hypoarticulated output. Twenty speakers, ten male and ten female served as subjects. Traditional citation-style locus equations were generated from productions of CV /t/ words printed on lists with initial /bdg/ followed by ten medial vowel contexts. Locus equations were also derived from spontaneous speech output which was generated by rapid reading of discourse-like passages constructed to resemble normal conversational speech, but with numerous lexical items beginning with /bdg/ in many vowel contexts. Spectral analysis was performed with the Kay Elemetrics CSL system. Comparisons of slopes, y intercepts, R^2, and standard error of estimates across the two speaking conditions will be presented and discussed.

4aSC25. Modeling the perception of features in the identification of English consonants. Tobey L. Doeleman (Dept. of Linguist., Cornell Phonet. Lab., Cornell Univ., Ithaca, NY 14853)

The perception of voicing, place, and manner was investigated in a series of gating experiments in which subjects identified eight English consonants in six different duration conditions [T. L. Doeleman, J. Acoust. Soc. Am. **101**, 3111(A) (1997)]. Subjects heard either all eight stimuli or two mutually exclusive subsets such that certain feature information was given (e.g., manner given in sets [p, t, b, d] and [f, s, v, z]). Although all subjects listened to the same 48 stimuli, identification significantly improved for subjects given place information (i.e., sets of labials and alveolars). The results are modeled using a spreading-activation connectionist model, the Integration-Competition model [Spivey-Knowlton (1996)]. In this model, the phoneme and feature representations compete with one another such that featural distinctions correspond to different weights assigned to the connection between feature and phoneme layers. The target phoneme has high activation from all featural arrays, resulting in the highest activation in the phoneme array. The increase in identification scores across duration conditions is modeled by the increase in activation of the target phoneme with successive time steps. The model also predicts observed consonant confusion patterns in that distracter phonemes sharing more features with the target phoneme will have higher activation levels.

4aSC26. Additive effects of phonetic distinctions in word learning. Joseph V. Pater (Dept. of Linguist., Univ. of Alberta, Edmonton, AB T6G 2E7, Canada, joe.pater@ualberta.ca), Christine L. Stager, and Janet F. Werker (Univ. of British Columbia, Vancouver, BC V6T 1Z4, Canada)

While infants have been demonstrated to be sensitive to a wide variety of phonetic contrasts when tested in speech discrimination tasks [Eimas *et al.* (1971) et seq.], recent work [Stager and Werker (1997)] has shown that following habituation to a word–object pairing, infants of 14 months fail to notice when the place of articulation of the initial consonant is switched [b/d]. Using the same procedure, the present study has found that infants do not respond to a change in voicing [b/p]. They do, however, notice a switch between dissimilar words [lɪf/nim]. One interpretation of these findings is that 14-month-olds do not encode either place or voice distinctions in lexical representations, so that words differing in only these features are treated as identical. To test this hypothesis, the effect of combining featural contrasts is currently being investigated by examining whether infants do respond to a change in both place and voice [d/p]. If there is such an additive effect, the contrasts must be represented. This would entail that an explanation for the failure to distinguish words differing in only a single feature should invoke processing factors, rather than representational ones.

4aSC27. Vowel perception by children and adults: Based on steady states or transitions? Joan E. Sussman, Jason Steinberg, and Julie Fenwick (Dept. of Communicative Disord. and Sci., State Univ. of New York at Buffalo, 122 Park Hall, Buffalo, NY 14260, jsussman@acsu.buffalo.edu)

The current investigation studied whether children, aged 4–5 years, and adults identified vowels on the basis of steady-state or transitional formant frequencies. Four types of synthetic tokens, created with a female voice, served as stimuli: (1) steady-state centers—220-ms formant frequencies for the vowels [i] and [ae]. (2) vowelless tokens with transitions appropriate for [bib] and [baeb], (3) ''congruent'' tokens that combined the first two types of stimuli into [bib] and [baeb], and (4) ''conflicting'' tokens that combined the transitions from [bib] with the vowel in [baeb] and vice versa. Results showed that children identified vowels most accurately in steady-state centers and congruent stimuli (96%) and used steady-state cues in 87% of conflicting tokens for vowel identification. Adults were equally accurate for vowelless, steady-state, and congruent tokens (99%) and used steady-state and transition cues for vowel identification. Findings do not support Nittrouer's Developmental Weighting Shift Hypothesis, but findings for adults do lend some support to the Dynamic Specification theory of Strange and colleagues. [Work supported by a Research Development Fund award from the Research Foundation at the University at Buffalo and by the Center for Hearing and Deafness at UB.]

4aSC28. Lexical competition in spoken word recognition by younger and older adults: A comparison of the rime cognate, neighborhood, and cohort. Mitchell S. Sommers (Dept. of Psych., Washington Univ., Campus Box 1125, St. Louis, MO 63130) and Shigeaki Amano (NTT Basic Res. Labs., 3-1 Morinosato-Wakamiya, Atsugi, Kanagawa, 243-01 Japan)

A central issue in research on spoken word recognition has been to specify the lexical candidates that are initially activated and compete during lexical discrimination. Amano [J. Mem. Lang. (submitted)] has recently proposed a new candidate set, the rime cognate (RC), which suggests that the set of potential lexical candidates consists of all items containing the same phoneme sequence as the rime (vowel nucleus plus coda) of that word. The present study was designed to compare the ability of the RC and two other sets, the neighborhood and cohort, to account for spoken word recognition in both younger and older adults. Density measures derived from the neighborhood were significantly correlated with identification scores for younger and older subjects. In contrast, RC density metrics were significantly correlated with identification scores only for older adults. Both neighborhood and RC density measures accounted for a greater percentage of the variance in spoken word recognition for older than for younger listeners. Metrics derived from the cohort were not significantly correlated with identification scores for either younger or older adults. The implications of these findings for several current models of spoken word recognition are discussed. [Work supported by the Brookdale foundation.]

4aSC29. Integration of auditory and visual speech information. Michael D. Hall, Paula M. T. Smeele, and Patricia K. Kuhl (Dept. of Speech and Hearing Sci., Univ. of Washington, Seattle, WA 98195, mdhall@u.washington.edu)

The influence of visual information on auditory speech perception can be observed under conditions where the two sources of information are discrepant [McGurk and MacDonald (1976)]. Recent studies from our laboratory [Hall *et al.* (1996)] focused on one such condition, where a face producing /b/ is combined with a dubbed auditory /g/ token, resulting in the frequent auditory perception of /bg/. These studies used auditory tokens that varied in quality, and have suggested that the probability of auditory consonant identification as /g/ was sufficient to predict the probability of /bg/ responses. The current study extended this approach to the converse conditions, where an auditory /b/ was dubbed on a visual /g/ production, which typically results in the frequent auditory perception of /d/ or /th/. Additionally, similarity ratings were collected for sets of auditory /g/ and /b/ stimuli, as well as for both types of auditory-visual conditions (/g/+/b/ and /b/+/g/). Multidimensional scaling (MDS) of these ratings was used to determine whether auditory category structure predicts MDS solutions for combined auditory and visual information. Implications of the results for the nature of auditory-visual speech integration will be discussed. [Work supported by NICHD.]

4aSC30. Speech-sound perception in normal and learning-disabled children: Effect of lengthened CV transition duration. Ann R. Bradlow, Nina Kraus, Trent Nicol, Therese McGee, Jenna Cunningham (Auditory Neurosci. Lab., Northwestern Univ., 2299 N. Campus Dr., Evanston, IL 60208, abradlow@nwu.edu), and Thomas D. Carrell (Univ. of Nebraska–Lincoln, Lincoln, NE 68538)

Previous research has established that many children with learning problems exhibit perceptual deficits in response to certain auditory stimuli, such as stop consonants in CV context. In order to investigate the precise acoustic features of stop consonants that pose perceptual difficulties for these children, discrimination thresholds were compared for normal and learning-disabled children along two separate synthetic /da/–/ga/ continua that differed only in the duration of the formant transitions. Results showed that simply lengthening the formant transition duration from 40 to 80 ms did not result in improved discrimination thresholds for the learning-disabled children. Similarly, an electrophysiologic response that is known to reflect the brain's preconscious ability to detect a change from one auditory stimulus to another indicated diminished responses in these children to /da/ versus /ga/, regardless of transition duration. This finding suggests that the brevity of the CV formant transitions is not the sole acoustic basis for the observed perceptual deficits. Furthermore, signal processing techniques aimed at enhancing perception of stop consonants likely need to consider other acoustic features, such as relative amplitude and frequency characteristics, instead of (or in addition to) CV transition duration. [Work supported by NIH R01 DC 01510.]

4aSC31. Listener variability and multiple perception processes. Stephanie Lindemann (Program in Linguist., Univ. of Michigan, 213 W. Mosley St., Apt. 6, Ann Arbor, MI 48103, lindeman@umich.edu)

The relationship between the perception of individual phonemes and the perception of words is investigated to determine if these are similar tasks for listeners. Since listeners may be able to use top-down processes in full words, it is argued that, at least for some, the task of identifying isolated phonemes may be very different from the task of identifying a full word. Because greater variability in performance is expected from non-native speakers, 15 native speakers of Sudanese Arabic (as well as a control group of 12 native speakers of American English) were tested on identification of American English phonemes in word context and the same instance of these phonemes extracted from the words. The overall agreement between corresponding items on the two tests was significantly better than chance, but scores of individual subjects varied substantially, with phoneme-word test agreement falling below chance for many subjects. Additionally, not all non-native subjects performed better on the full word task, suggesting that they do not all use higher-level information. It is therefore argued that studies of phoneme-level perception are more relevant to some listeners' word perception than to others'.

4aSC32. Effects of affective tone of voice on spoken word recognition. Lynne C. Nygaard, Jennifer S. Queen, and S. Alexandra Burt (Dept. of Psych., Emory Univ., Atlanta, GA 30322)

Traditionally, the study of emotional tone of voice has been considered separately from the study of formal linguistic aspects of spoken language. Research has typically focused either on how listeners detect emotion in an individual's voice, or on how listeners extract abstract linguistic content. The present study was designed to investigate how these two sources of information interact. In two experiments, listeners were presented with homophones that had one affective (happy or sad) and one neutral meaning and with nonhomophones that had either a happy, sad, or neutral meaning. Within each experiment, words were spoken in a tone of voice that was either congruent, incongruent, or neutral with respect to affective meaning. In the first experiment, naming latencies were collected to determine if tone of voice affects the speed of lexical processing. In the second experiment, transcription accuracy was assessed to determine if tone of voice affects the selection of word meaning. The results suggest that emotional tone of voice was used to resolve lexical ambiguity, but did not appear to affect specifically the time course of lexical access. The implications of these results for current theories of spoken word recognition will be discussed. [Work supported by NIH.]

4aSC33. On the perception of qualitative and phonetic similarities among voices. Robert E. Remez, Jennifer L. Van Dyk (Dept. of Psych., Barnard College, 3009 Broadway, New York, NY 10027-6598), Jennifer M. Fellowes (Columbia Univ. College of Physicians and Surgeons, New York, NY 10032), and Philip E. Rubin (Haskins Labs., New Haven, CT 06511)

A perceiver who learns to recognize an individual talker becomes familiar with attributes of the talker's voice that are present in any utterance regardless of the linguistic message. Customary accounts of individual identification by ear presume that such durable personal aspects of an individual's speech are graded qualities (e.g., vocal pitch and pitch range; melodious, breathy, or raspy timbre; etc.) independent of the acoustic properties that evoke segmental phonetic contrasts. Alternatively, some classic and recent studies alike suggest that familiarity includes attention to attributes of dialect and idiolect conveyed in the articulation of consonants and vowels. These linguistic phonetic properties of speech are effective for recognizing a talker when voice quality is eliminated as a source of information. The present investigation sought direct evidence of attention to qualitative and phonetic attributes of speech. Natural samples and sine wave replicas of sentences spoken by five male and five female talkers were used in a similarity tournament with adult listeners. The results establish the differential perceptual resolution of qualitative and phonetic attributes in the perception and recognition of talkers. [Work supported by NIDCD and NICHD.]

4aSC34. Perception of American English /r/ and /l/ by Mandarin speakers: Influences of phonetic identification and category goodness. Feng-Ming Tsao, Michael Hall, Richard Eyraud, and Patricia K. Kuhl (Dept. of Speech and Hearing Sci., Univ. of Washington, Seattle, WA 98195)

Studies in speech perception show a perceptual magnet effect for listeners which is characterized by shrunken perceptual distances near excellent exemplars of phonetic categories and stretched distances near poor exemplars. This study extended this work by examining the influence of Mandarin speakers' phonetic identification and category goodness on their perception of American English /r/ and /l/. Eighteen /ra/ and /la/ tokens were used that varied $F2$ and $F3$ frequencies. Listeners identified stimuli as Mandarin /u/, /m/, or /l/, provided goodness ratings for them, and rated the similarity of all stimulus pairs. Multidimensional scaling analyses of similarity ratings exhibited that the perceptual ''map'' for Mandarin speakers differs substantially from that of American speakers [Iverson and Kuhl, J. Acoust. Soc. Am. (1996)] listening to the same stimuli. The results indicate that (1) Mandarin speakers perceived /u/ and /l/ in the space, (2) perceptual distance was shrunk near the best exemplars of phonetic categories and stretched near the category boundary, (3) variations of $F2$ were more important than $F3$, and (4) individual differences in /l/ identification corresponded to a degree of shrinking near the best exemplars of the /l/ category. Results indicate that listeners use their native-language representations in the perception of a non-native language.

4aSC35. Non-native listeners' representations of within-word structure. Takashi Otake (Dokkyo Univ. and Max-Planck Inst. for Psycholinguistics, Wundtlaan 1, 6525 XD, Nijmegen, The Netherlands, otake@dokkyo.ac.jp), Kiyoko Yoneyama (The Ohio State Univ., Columbus, OH 43210-1298), and Hideki Maki (Salem-Teikyo Univ., Salem, WV 25426-0500)

Human listeners may form conscious representations of potential within-word structure in which lexicon is represented by some phonological units. An earlier study examining monolingual speakers of Japanese and English with native inputs suggests that levels of representation by Japanese speakers may be involved with richer knowledge of word-internal structure, while English speakers are sensitive to syllables [Otake *et al.*, Proceedings of EUROSPEECH 95 **3**, 1703–1706 (1995)]. The present study investigated how monolingual speakers of English learning Japanese could form conscious representations of potential within-word structure in Japanese. Three groups of subjects (N: 36, 33, and 40 for three different levels) were presented with 150 Japanese spoken words and asked to mark on a written transcript of each word the second natural division point from the onset in the word. The statistical analysis showed that all groups exploited syllables to represent Japanese words irrespective of Japanese proficiency. These results suggest that the exploitation of phonological unit to represent within-word structure in foreign inputs may be the same as the one in the native input as proposed by our recent pilot work (Otake and Yamamoto, 134th Meeting of the ASA **101**). [Work supported by Fulbright senior research grant and TAF.]

4aSC36. The discovery of natural classes by non-native listeners. John Kingston and Elliott Moreton (Linguist. Dept., South College 226, Univ. of Massachusetts, Amherst, MA 01003)

Using natural disyllabic stimuli, English speakers were trained to sort one, two, or three pairs of German nonlow rounded vowels into two categories, then a new pair was added and their ability to sort it into the same categories was tested. Each pair contrasted in [high], [tense], and [back]. Category membership was determined by only one of these features. Three-pair training provided enough information to tell which one; two-pair narrowed it down to two possibilities; one-pair left it open. Listeners in three experiments using different pairs of test vowels sorted the test vowels significantly better as the number of training pairs increased. This suggests they were inducing feature-based natural classes. The rival exemplar model of Nosofsky [J. Exp. Psych.: Gen. **115**, 39–57 (1986)] would say that while more pairs bring more phonetic variety, they also exemplify more robustly the dimensions along which the categories differ phonetically, and thereby shift attentional weight to those dimensions. Similar experiments will pit these two hypotheses against one another with training pairs which contrast only in the feature that defines the sort categories. If listeners are using feature-based natural classes, their performance should not improve with more training pairs; if they are using exemplars, it should. [Work supported by NIH.]

Session 4aSP

Signal Processing in Acoustics, Engineering Acoustics, Architectural Acoustics and Education in Acoustics: Acoustics in Multimedia—Systems Issues I

Daniel R. Raichel, Chair

Department of Mechanical Engineering, Steinman Hall, The City College of The City University of New York, New York, New York 10031

Invited Papers

8:00

4aSP1. *The Journal of the Acoustical Society of America* and other technical journals of the future—a new multimedia format?
Daniel R. Raichel (Dept. of Mech. Eng., Steinman Hall, The City College of The City Univ. of New York, New York, NY 10031)

The *Journal of the Acoustical Society of America (JASA)* is now available in two formats—the traditional printed issue and the CD-ROM. However, CD-ROMs are in the early process of being displaced by the newer, more powerful DVDs which feature a capacity of approximately 2.8 GB per disk side rather than the 650 MB standard for each CD-ROM. CD-ROMs, however, will not immediately be rendered obsolete by the use of a current DVD player since the latter is backward compatible for CD-ROM playback. However, even in the CD-ROM format, a technical journal can take on new aspects of multimedia, which were hitherto unavailable or impracticable in printed media. The greatly enhanced capacity of the DVD disk provides even greater opportunities for extended expositions. Animation scenes, photographs, extended databases which would not have otherwise been included in a printed article, dynamic plots, sound effects, virtual reality scenes which permit one to ''visit'' laboratory sites or ''witness'' the topical phenomena, and even computer programs and entire reproductions of cited references can be incorporated in an article slated for publication on the new multimedia disks or on future storage devices.

8:25

4aSP2. Mediacoustics, a software package on CD for teaching acoustics using multimedia techniques. Michael Wahlrab (01dB, Inc., P.O. Box 796, Skaneateles, NY 13152)

Mediacoustics uses multimedia techniques on CD for teaching acoustics. It is divided into four study areas: basic physics, noise and man, room acoustics, and noise control. Sound clips, text, pictures, photos, and video animation illustrate each module. The index and chapters may be edited. Individual text passages may be annotated and an included acoustical dictionary provides definitions of common terms and formulas. Any contemporary multimedia computer equipped with a sound card and amplified loudspeaker can be used to access Mediacoustics and its hundreds of video clips.

8:50

4aSP3. Animations created in *Mathematica* for acoustics education. Victor W. Sparrow (Grad. Prog. in Acoust., Penn State Univ., University Park, PA 16802) and Daniel A. Russell (Kettering Univ., Flint, MI 48504)

One of the unifying assumptions in acoustics and vibrations is that waves and structures move and vibrate. In teaching these topics it makes sense to show students exactly HOW things move. For the last several years the authors have been using the symbolic manipulation program *Mathematica* [Wolfram Research, Inc., Champaign, IL] to produce brief animations for teaching acoustics. World Wide Web sites have been created for easy access to the animations by all students. The present paper will explain how the animations are created and will demonstrate several including one and two degrees of freedom oscillators, a piston in a tube, reflected waves, doppler effect, elastic waves, circular membranes, and circular plates. The current animations are relatively simple constructions to keep the file sizes to a minimum. However, improved animations containing substantially enhanced graphics will be possible with Internet 2.

Contributed Papers

9:15

4aSP4. A software teacher for acoustical measurements. Ila Tokola, Matti Karjalainen, and Martti Rahkila (Helsinki Univ. of Technol., Lab. of Acoust. and Audio Signal Processing, P.O. Box 3000, FIN-02015, Helsinki, Finland)

The huge development in computer technology has made it tempting to use computers in acoustics education. QuickSig, an object-oriented Common Lisp based signal processing environment has been used for acoustical measurements. It provides an excellent means to present graphics and handle signals, and it is easy to program for educational purposes as well. A Simple Loudspeaker Measurement Program is a Computer Based Education (CBE) application for self-studying of acoustical measurements. The application deals with free-field measurements of loudspeakers and related theory. The measurements are actually made with the program and genuine results are achieved each time. The software teacher interactively helps the student to obtain results that make sense and monitors the progress with multiple-choice questions. The results can also be

saved for post-processing. The main difference between this application and on-line help systems in measurement packages is that the teacher emphasizes learning of acoustical measurements, not the measurement software itself! On the other hand, with this application, the students do real measurements, including the loudspeaker and microphone setup in an anechoic chamber, unlike most self-study CBE applications. Furthermore, the application can be used also by engineers, technicians, etc. for assisted measurements.

9:35

4aSP5. Design and development of PC-IMAT. John W. Schuler (Naval Personnel Res. and Development Ctr., Armstrong Lab., Brooks AFB, TX 78235) and Murray S. Korman (U.S. Naval Acad., Annapolis, MD 21402)

NPRDC has been tasked to provide empirical evidence of effective instructional strategies for the acquisition of conceptual knowledge under the project name PC-IMAT (Personalized Curriculum for Interactive Multisensor Analysis Training). The domain to be investigated and demonstrated includes those concepts required for the successful planning and execution of antisubmarine warfare (ASW), specifically, the conceptual knowledge underlying the prediction of sound transmission paths and detection ranges. Navy-standard models which are used in fleet SONAR prediction systems are available to support the learners' conceptual understanding of the elements of the oceanographic environment which affect acoustic propagation. To date, these models are employed in a microcomputer based, stand-alone delivery architecture in both a linear interactive courseware (ICW) format and in modules suitable for independent exploratory learning. Midshipmen taking SP411 (Underwater Acoustics and Sonar) are currently using PC-IMAT to help investigate what are the effective instructional strategies which convey understanding of a complex multivariate domain (like ray trace or propagation loss models). Research on ''scientific visualization'' (to enhance comprehension and retention) and student feedback will also be used to help develop and evaluate other training materials including beamforming, reverberation, target motion analysis, and scattering. [Work supported by ONR.]

9:55

4aSP6. Construction of a HATS and its HRTF measurement for 3-D sound. Kyeong Ok Kang, Dong-Gyu Kang, Min-Soo Hahn (Elec. and Telecom. Res. Inst., P.O. Box 106, Yusong Post Office, Taejon, 305-600, Korea, kokang@audio.etri.re.kr), Mun-Jae Jho (Korea Res. Inst. of Standards and Sci., Taejon, Korea), and Dae-Kwon Jeong (Korea Aviation Univ., Koyang, Korea)

Based on the anthropometric data on the head and torso dimensions in Korean male adults, a head and torso simulator (HATS) was constructed. Data, which have no counterparts in the Korean standards, were used with ANSI S3.36-1985. Measuring microphones can be positioned at the eardrum or at the ear canal entrance according to measuring purposes. Head related transfer functions (HRTFs) for the HATS, i.e., binaural impulse responses, were measured at 710 points on a spherical surface of radius 1.55 m using a burst maximum length sequence (MLS) signal of 65 535 samples in an anechoic chamber. The measurement system consists of one part to generate the MLS signal and to drive a Boss 101A loudspeaker and another part to record the output signal of a microphone in the HATS and consequently to measure the impulse response. Also measured were the impulse responses of the driving loudspeaker and some headphones for 3-D sound reproduction in order to get the exact HRTF of the HATS-alone. The impulse-version HRTFs at the sampling frequency of 44.1 kHz, which have filter lengths of 512 points with minimum phase characteristics and can be used for 3-D sound, were finally obtained through a post-processing procedure.

10:15

4aSP7. Measuring and modeling the effect of source distance in head-related transfer functions. Jyri Huopaniemi and Klaus A. J. Riederer (Helsinki Univ. of Technol., Lab. of Acoust. and Audio Signal Processing, P.O. Box 3000, FIN-02015, Helsinki, Finland)

Efficient modeling of human spatial hearing by digital filter approximations of head-related transfer functions (HRTFs) is the key technology in 3-D sound processing. It is well known that the HRTF bears the major static localization cues, the interaural time difference (ITD), and the interaural level difference (ILD) that are functions of frequency and the incident angle of arrival. The effect of source distance has, however, often been neglected in HRTF models. In this paper, a method for efficient distance-dependent HRTF modeling is presented, which is based on both theoretical and empirical data. HRTF measurements on eight human subjects and one dummy head were carried out at two source distances, 2 and 0.65 m. It has been argued in the literature that the distance changes mainly affect the ILD, whereas the ITD remains approximately constant. Based on this finding, which was also supported by the measurements performed in this study, a filter structure that models the ILD change as a function of distance was derived. The results of this study are applicable to many near-field listening applications.

Session 4aUW

Underwater Acoustics: 3-D Propagation Effects I: Where are We Today in Models and Measurements?

Michael B. Porter, Cochair
Scripps Institute of Oceanography, Marine Physical Laboratory, MC 0205, 8602 La Jolla Shores Drive, La Jolla, California 92037-0205

Alexandra I. Tolstoy, Cochair
4224 Waialae Avenue, Suite 5-260, Honolulu, Hawaii 96816

Chair's Introduction—9:15

Invited Papers

9:20

4aUW1. The penetrable wedge as a three-dimensional benchmark. Grant B. Deane (Scripps Inst. of Oceanogr., La Jolla, CA 92003)

In 1989, members of the Acoustical Society of America developed a suite of range-dependent benchmark geometries, allowing the underwater acoustics community to compare the performance of analytical and numerical solutions to two-dimensional propagation problems. These benchmarks were successful in providing the acoustics community with a set of standards, and continue to be useful as reference geometries. Over the past decade, there have been significant advances in analytical and numerical solutions to three-dimensional propagation problems, and establishing three-dimensional benchmarks is a timely development. A useful benchmark must eliminate the undesired complications of a real-word environment but retain the essential physics of the problem while being tractable to a variety of modelers. A geometry which satisfies these constraints is the ocean wedge with a penetrable basement. The three-dimensional Green's function for the penetrable wedge has already been studied by a number of investigators, and measured with scale-model tank experiments. The state of modeling and measurement for the wedge will be discussed in the context of its usefulness as a three-dimensional benchmark.

9:40

4aUW2. Three-dimensional propagation effects: Modeling, observations, and suggested benchmark cases. Kevin B. Smith (Dept. of Phys., Naval Postgrad. School, Monterey, CA 93943, kevin@physics.nps.navy.mil)

Over the past several years, many acoustic propagation models have been adapted to compute the influence of azimuthal coupling. These so-called three-dimensional (3-D) models should provide more accurate predictions of the acoustic field in 3-D variable environments than previous 2-D×N models. However, such advanced 3-D propagation models are generally slower than their 2-D ×N predecessors. It is, therefore, important to understand the significance of the difference between 3-D and 2-D×N and when such effects should be considered. Furthermore, as of this date, no formal set of benchmark cases and solutions has been defined to test the accuracy of the various 3-D models currently being used. This paper will provide a general overview of 3-D propagation models and how they can be used to assess the significance of azimuthal coupling in ocean acoustics. Specific examples of such influences will be provided from a 3-D parabolic equation model and compared to results from the corresponding 2-D×N version. Of particular interest will be the pseudo-3-D effects predicted by 2-D×N calculations and the ability to distinguish true 3-D effects in experimental data. Finally, several environmental scenarios will be suggested as possible benchmark cases for future studies. [Work sponsored by ONR Code 321OA.]

10:00

4aUW3. Experimental measurements of three-dimensional underwater sound propagation over a variable bathymetry. Stewart A. L. Glegg, Joseph M. Riley, and Antony LaVigne (Ctr. for Acoust. and Vib., Dept. of Ocean Eng., Florida Atlantic Univ., Boca Raton, FL 33431)

There appear to be very few experimental measurements of the three-dimensional features of sound fields in the ocean, especially in shallow water where bottom interactions can have a dominant effect. One of the reasons for this lack of experimental data can be attributed to the difficulty in designing an experiment to identify three-dimensional propagation effects. Bathymetric refraction occurs, for the most part, on a very large scale and so to differentiate between local variability of the propagation and refraction by a bathymetric feature, measurements have to be carried out simultaneously over a large area. This is easier to achieve in the laboratory environment where repeatability of the experimental results is not an issue. This paper will describe a series of laboratory scale experiments which were carried out to identify the major features of underwater sound propagation over a realistic three-dimensional bathymetry. The model used is based on the bathymetry of the Santa Lucia Escarpment, but is not unlike bathymetry found in other coastal regions in different water depths if the correct scaling is used. A number of three-dimensional sound maps will be shown of the field from a point source. [Work supported by ONR.]

10:20

4aUW4. Three-dimensional sound propagation modeling of the Santa Lucia escarpment data. Michael B. Porter (Marine Physical Lab., Code 0205, 8602 La Jolla Shores Dr., La Jolla, CA 92037, porter@mpl.ucsd.edu)

The KRAKEN normal-mode program includes options for both $N\times2$-D and full 3-D modeling. The latter is implemented using Gaussian beams to treat the horizontal refraction of modes. Since the model allows for mixed acoustoelastic problems, 3-D sound propagation can be simulated over ocean bottoms supporting shear waves. To study its accuracy, the model is applied to several 3-D scenarios, including both a simple wedge and a topographically complicated case based on the Santa Lucia escarpment off the coast of California. The results are compared to the laboratory scale data presented by Glegg *et al.* in this session.

10:35–10:40 Break

10:40

4aUW5. Benchmarking two three-dimensional parabolic equation methods. Frederic B. Sturm (Lab. M.N.C., I.S.I.T.V, Université de Toulon, France), John A. Fawcett, and Finn B. Jensen (SACLANT Undersea Res. Ctr., 19138 La Spezia, Italy)

Results of acoustic propagation modeling using two different three-dimensional parabolic equation methods are presented. One model [J. Fawcett, J. Acoust. Soc. Am. **93**, 2627–2632 (1993)] uses standard splitting and operator approximations while the other method [F. Sturm, M. Pélissier, and D. Fattaccioli, *Proc. of the Third European Conference on Underwater Acoustics*, 243–248 (1996)] does not split the azimuthal operator and maps the bathymetry into an equivalent flat bottom problem. Examples of propagation over wedge and corrugated bathymetries are considered for different frequencies. Both point-source and modal initial fields are used. The three-dimensional solutions exhibit interesting three-dimensional refractive effects. The convergence of the solutions with respect to azimuthal discretization is also discussed.

10:55

4aUW6. Parabolic equation modeling with the split-step Fourier algorithm in four dimensions. F. D. Tappert (Appl. Marine Phys., Univ. of Miami, RSMAS, Miami, FL 33149, ftappert@rsmas.miami.edu)

Fully range-dependent 3-D (with coupled azimuths) single-frequency PE/SSF models have existed for many years, and 2-D (vertical plane at fixed bearing) wide-angle broadband PE/SSF models have existed for even longer. By combining these, a 4-D (three space dimensions plus time) PE/SSF model has been developed that runs on a desktop computer. This model is exactly reciprocal if ocean currents are omitted, and is exactly energy conserving if absorption is omitted. Examples of model predictions, projected onto two dimensions, are displayed for shallow water propagation. In a situation where ray chaos due to variable bathymetry is strong, plots of intensity as a function of travel time and bearing at fixed depth and range are especially interesting. [Work supported by ONR.]

11:10

4aUW7. Three-dimensional ocean acoustics: Techniques and examples. Michael D. Collins, Joseph F. Lingevitch, and Gregory J. Orris (Naval Res. Lab., Washington, DC 20375, collins@ram.nrl.navy.mil)

The method of matched asymptotics can be used to solve scattering problems involving a compact object in a waveguide [M. D. Collins and M. F. Werby, J. Acoust. Soc. Am. **85**, 1895–1902 (1989)]. This approach involves different asymptotic limits in different regions. In the inner region near the object, the Green's function can be approximated by either the free-space or half-space Green's function depending on the vertical location of the object. The inner problem is therefore the free-space or half-space scattering problem. The outer solution satisfies the three-dimensional parabolic equation. When horizontal variations in the waveguide are sufficiently gradual, the solution is asymptotic to a specular point source for $ka=O(1)$. This approximation can be improved by including a vertical dipole correction. The three-dimensional parabolic equation has been applied to free-space scattering problems by solving the exterior problem for the scattered field directly [M. F. Levy and A. A. Zaporozhets, J. Acoust. Soc. Am. **103**, 735–741 (1998)]. This approach is also useful for some waveguide scattering problems. [Work supported by ONR.]

11:25

4aUW8. Limitations on adiabatic normal modes in range-varying environments. Peter C. Mignerey (Acoust. Div. 7120, Naval Res. Lab., Washington, DC 20375-5350, mignerey@nrl.navy.mil)

Curvilinear coordinates are used to extend adiabatic normal modes to weak range-dependent environments including those with sloping bottoms. Reciprocal vertical wave numbers of a reference mode provide the coordinate scale factors. This approach is valid for negligible horizontal derivatives of intermodal vertical-wave number ratios. Large gradients in the bottom impedance make these ratios large, implying significant mode coupling in oceanic regions spanning both soft and hard terrains. Additionally, a large number of eigenvalues are needed to compute adiabatic normal-mode solutions for high-resolution acoustic fields. The direct solution of eigenvalue problems on a fine grid is often prohibitive. Alternatively, the eigenvalues may be interpolated to the fine grid. This approach usually requires extrapolation toward edges across which the horizontal wave number becomes imaginary at mode cutoff. One choice for an interpolation function is the integrated reciprocal vertical wave number. This function, proportional to water depth, is smooth and real throughout the cutoff region, while causing the horizontal wave number to vanish along the cutoff line. [Work supported by the Office of Naval Research and partially by DOD High Performance Computing facilities at NRL.]

11:40

4aUW9. Three-dimensional acoustic effects due to ocean currents. Oleg A. Godin (School of Earth and Ocean Sci., Univ. of Victoria, P.O. Box 1700, Victoria, BC V8W 2Y2, Canada, ogodin@uvic.ca)

In the presence of currents, vertically stratified fluids become acoustically anisotropic in the horizontal plane and the group and phase velocities of the acoustic normal mode are generally not parallel. In this paper, the resulting 3-D effects and their implications for acoustic monitoring of ocean currents are discussed in terms of the normal modes. The sensitivity of various acoustic quantities to the vertical and in-plane and out-of-plane horizontal components of the flow velocity is estimated. Despite anisotropy, there is no azimuth coupling at long-range sound propagation in a stratified moving medium. When a waveguide varies in the horizontal plane, the 3-D effects due to current-induced anisotropy and horizontal refraction are superimposed. The relative importance of these two phenomena in shallow- and deep-water scenarios is analyzed by considering horizontal (modal) rays in the presence of currents. For the purposes of numerical simulations, a parabolic approximation is considered for sound in a moving fluid. A new parabolic equation is proposed for efficient numerical modeling of the 3-D acoustic effects in the ocean with currents. [Work supported by NSERC.]

11:55

4aUW10. Three-dimensional solutions to range-dependent problems in shallow water by a pseudo-spectral method. Altan Turgut and Stephen N. Wolf (Naval Res. Lab., Acoust. Div., Washington, DC 20375)

A pseudo-spectral method is used to solve the acoustic and elastic wave equations in the time domain. The application of such a computation and memory-intensive method to shallow-water propagation and scattering problems is discussed. Three-dimensional solutions are given for two

different shallow-water related problems: pulse propagation through solitary internal waves and pulse scattering from bottom inhomogeneities. The size of a typical computational domain for both problems is about (30 wavelength)^3. However, for the propagation problem, distances up to several hundred wavelengths have been achieved by moving the computational domain with the wavefront. The accuracy of the pseudo-spectral method has been checked with analytical solutions to simpler problems. Two-dimensional results of moving-domain approach are compared to those of single-domain and good agreement is observed. The existing method is expected to confirm the accuracy of the approximate solutions such as PE in a complicated range-dependent environment. [Work supported by ONR.]

THURSDAY MORNING, 25 JUNE 1998 GRAND BALLROOM III (W), 11:00 A.M. TO 12:00 NOON

Session 4aPLb

Plenary Lecture

Donna L. Neff, Chair

Boys Town National Research Hospital, 55 North 30th Street, Omaha, Nebraska 68131

Chair's Introduction—11:00

Invited Paper

11:05

4aPLb1. Psychoacoustics of cochlear hearing impairment and the design of hearing aids. Brian C. J. Moore (Dept. of Experimental Psych., Univ. of Cambridge, Downing St., Cambridge CB2 3EB, England, bcjm@cus.cam.ac.uk)

Cochlear hearing impairment is usually associated with damage to the hair cells within the cochlea. When the damage is restricted to the outer hair cells (OHCs), the main consequence is disruption of the ''active'' mechanism which normally enhances the response of the basilar membrane to weak sounds and which sharpens the tuning (frequency selectivity) of the basilar membrane. Psychoacoustically, damage to OHCs results in loss of sensitivity (elevated absolute thresholds), loudness recruitment, and reduced frequency selectivity. Damage to the inner hair cells (IHCs) causes basilar membrane vibrations to be transduced less effectively, so absolute thresholds are elevated, but does not result in altered frequency selectivity or loudness recruitment. Sometimes, IHCs and/or neurons may be completely inoperative at certain places within the cochlea, giving rise to ''dead regions.'' Such regions can strongly influence the perception of pitch and loudness. Current hearing aids can partially compensate for the effects of loudness recruitment by using compression amplification, but there is much controversy about the ''best'' form of compression. The deleterious effects of reduced frequency selectivity on speech intelligibility in noise can be alleviated by various methods for improving the speech-to-noise ratio, although so far only directional microphones have given clear benefits.

Meeting of the Standards Committee Plenary Group

ORGANIZATION OF STANDARDS COMMITTEE PLENARY GROUP MEETING

S1 Acoustics—U.S. Technical Advisory Group (TAG) for IEC/TC 29 Electroacoustics and ISO/TC 43 Acoustics

S3 Bioacoustics—U.S. Technical Advisory Group (TAG) for ISO/TC 43 Acoustics, IEC/TC 29 Electroacoustics, and ISO/TC 108/SC4 Human Exposure to Mechanical Vibration and Shock

S12 Noise—U.S. Technical Advisory Group (TAG) for ISO/TC 43/SC1 Noise and ISO/TC 94/SC12 Hearing Protection

The meeting of the Standards Committee Plenary Group will precede the meetings of the Accredited Standards Committees S1, S3, and S12, to take place in the following sequence on the same day: S12—9:30 a.m. to 11:30 a.m., St. Helens Room; S3—2:00 p.m. to 3:15 p.m., St. Helens Room; S1—3:30 p.m. to 4:45 p.m., St. Helens Room.

Discussion at the Standards Committee Plenary Group meeting will consist of national items relevant to all S Committees, plus a review of the international standardization (U.S. TAG) activities including reports on recent meetings and planning for forthcoming meetings.

Members of S2 on Mechanical Vibration and Shock (and U.S. TAG for ISO/TC 108 and five of its Subcommittees, SC1, SC2, SC3, SC5, and SC6) are also encouraged to attend the Standards Committee Plenary Group meeting, even though the S2 meeting will take place one day earlier, on Wednesday, 24 June 1998 at 9:00 a.m.

The U.S. Technical Advisory Group (TAG) Chairs for the various international Technical Committees and Subcommittees under ISO and IEC, which are parallel to S1, S2, S3, and S12 are as follows:

U.S. TAG Chair/Vice Chair	TC or SC	U.S. TAG
ISO		
P. D. Schomer, Chair H. E. von Gierke, Vice Chair	ISO/TC 43 Acoustics	S1 and S3
P. D. Schomer, Chair H. E. von Gierke, Vice Chair	ISO/TC 43/SC1 Noise	S12
E. H. Berger, Chair	ISO/TC 94/SC12 Hearing Protection	S12
J. Erdreich, Chair H. E. von Gierke, Vice Chair	ISO/TC 108/SC4 Human Exposure to Mechanical Vibration and Shock	S3
D. J. Evans, Chair	ISO/TC 108 Mechanical Vibration and Shock	S2
R. H. Mehta, Chair K. Won, Vice Chair	ISO/TC 108/SC1 Balancing, including Balancing Machines	S2
A. F. Kilcullen, Chair	ISO/TC 108/SC2 Measurement and Evaluation of Mechanical Vibration and Shock as Applied to Machines, Vehicles and Structures	S2
B. E. Douglas, Chair	ISO/TC 108/SC3 Use and Calibration of Vibration and Shock Measuring Instruments	S2
D. J. Vendittis, Chair	ISO/TC 108/SC5 Condition Monitoring and Diagnostics of Machines	S2
G. Booth, Chair	ISO/TC 108/SC6 Vibration and Shock Generating Equipment	S2
IEC		
V. Nedzelnitsky, U. S. TA	IEC/TC 29 Electroacoustics	S1 and S3

Meeting of Accredited Standards Committee S12 on Noise

P. D. Schomer, Chair S12, and Chair U. S. Technical Advisory Group (TAG) for ISO/TC 43/SC1, Noise
U. S. CERL, P.O. Box 9005, Champaign, Illinois 61826-9005

B. M. Brooks, Vice Chair, S12
Brooks Acoustics Corporation, P.O. Box 3322, Vernon, Connecticut 06066

H. E. von Gierke, Vice Chair, U.S. Technical Advisory Group (TAG) for ISO/TC 43/SC1, Noise
1325 Meadow Lane, Yellow Springs, Ohio 45387

E. H. Berger, Chair, U. S. Technical Advisory Group (TAG) for ISO/TC 94/SC12, Hearing Protection
E-A-R/Aearo Company, 7911 Zionsville Road, Indianapolis, Indiana 46268-1657

Accredited Standards Committee S12 on Noise. Working group chairs will report on their progress for the production of noise standards.

SCOPE OF S12: Standards, specifications and terminology in the field of acoustical noise pertaining to methods of measurement, evaluation and control; including biological safety, tolerance and comfort and physical acoustics as related to environmental and occupational noise.

THURSDAY AFTERNOON, 25 JUNE 1998 CASCADE BALLROOM II (W), 1:00 TO 3:55 P.M.

Session 4pAAa

Architectural Acoustics: Case Study on the New Tokyo International Forum

J. Christopher Jaffe, Chair
Jaffe Holden Scarbrough Acoustics, Inc., 114A Washington Street, Norwalk, Connecticut 06854

Chair's Introduction—1:00

Invited Papers

1:15

4pAAa1. Role of the acoustic advisors in the design of the Tokyo International Forum. Teruji Yamamoto (Toho Univ., Funabashi, Japan), Hideki Tachibana (Univ. of Tokyo, Japan), and Christopher Jaffe (Jaffe Holden Scarbrough Acoust., Inc., 114A Washington St., Norwalk, CT 06854)

The Tokyo Municipal Government appointed two acoustical advisors to review the acoustic design criteria developed for the Tokyo International Forum, Dr. Teruji Yamamoto and Dr. Hideki Tachibana. This paper describes the role of acoustic advisors on such a large project and show how they were able to assist the design team throughout the design and construction of the complex. Items to be covered include: process of design competition, acoustic requirements, process of acoustical design, outline of construction; and acoustical check during and after construction.

1:45

4pAAa2. Room acoustic designs for the Tokyo International Forum. Christopher Jaffe, Mark Holden, Russell Cooper (Jaffe Holden Scarbrough Acoust., Inc., 114 A Washington St., Norwalk, CT 06854, dcoppola@jhsacoustics.com), T. Kobayashi, and F. Kawakami (Yamaha Acoust. Res. Labs., Hamamatsu, Japan)

The room acoustic design of the Tokyo International Forum centered around the four performance halls: the convention center, the atrium, and the large multi-use meeting/dining halls. This paper will describe the program use of each space and the acoustic criteria developed to meet these requirements. Major emphasis will be placed on Hall A, a 5000-seat congress hall that was to be used to unamplified symphonic concerts and operas as well as highly amplified Broadway musicals and rock concerts, and Hall B, a 1500-seat classical music hall and piccolo opera house that also would be used for conferences and popular entertainment. A unique architectural feature of Hall B is the first design and installation of a Concert Hall Shaper, a device that enables a proscenium theatre with a stagehouse to be converted to a single acoustic volume for events such as symphony concerts requiring a longer reverberation time. Hall C, an extremely flexible space for fashion shows, summit meetings, and large receptions, and Hall D, another versatile space for theater, recitals, and live television productions, will also be described.

2:15–2:25 Break

2:25

4pAAa3. Sound and vibration control for the Tokyo International Forum. Fukushi Kawakami, Tetsu Kobayashi (Yamaha Acoust. Res. Labs., Hamamatsu, Japan), Christopher Jaffe, Mark Holden, and Russell Cooper (Jaffe Holden Scarbrough Acoust., Inc., Norwalk, CT 06854)

The acousticians for the Tokyo International Forum were faced with a number of very severe noise and vibration problems related to the siting of the complex in the center of downtown Tokyo, the placement of the performance halls next to one another, and the location of a portion of the mechanical equipment on the roof. In addition, subway lines ran parallel to three of the boundaries of the structure, another line ran under the building, and the main tracks of Japan Rail ran above ground alongside the major atrium. This paper will discuss the noise criteria set for the performance rooms and other noise sensitive spaces, the noise and vibration measurements taken to determine the severity of the train-related conditions, predictions of other noise and vibration sources, and the solutions developed by the design team to meet the criteria required by the program. Among the solutions are a secant wall surrounding the foundation, box in box construction for the four performance halls, blocking mass installations at key locations throughout the steel skeleton of the complex, and isolation joints and sound locks between the performance rooms.

2:55

4pAAa4. Design of sound reinforcement and acoustic field control systems for the Tokyo International Forum. Christopher Jaffe, Mark Holden, Paul Scarbrough, David Robb (Jaffe Holden Scarbrough Acoust., Inc., 114A Washington St., Norwalk, CT 06854, dcoppola@jhsacoustics.com), Yasushi Shimizu, Shinjirou Yamashita, and Fukushi Kawakami (Yamaha Acoust. Res. Labs., Hamamatsu, Japan)

The extremely varied program use of the four performance halls at the Tokyo International Forum provided the electroacoustic designers with a number of extremely challenging problems. This paper describes the events that were to take place in all of the rooms, the designs selected to accommodate these programs, and the coordination of the design of the electroacoustic systems with the room acousticians, the architects, and the theater consultants. Of special interest is the design of the sound systems for Hall A, the 5000 seat symphony/congress hall which was to be used for opera and symphony performances without employing close-in microphone amplification techniques, and Hall D, which was to be used for live theater and recitals as well as television productions. The final portion of the paper will discuss the sound contractor's role in completing such an immense project and in particular the relationship that contractors have with designers and owners in Japan.

3:25–3:55 Discussion

Session 4pAAb

Architectural Acoustics: Performance Spaces (Poster Session)

Michael R. Yantis, Cochair
Michael R. Yantis Associates, Inc., 1809 7th Avenue, Suite 1609, Seattle, Washington 98101

Basel Jurdy, Cochair
Michael R. Yantis Associates, Inc., 1809 7th Avenue, Suite 1609, Seattle, Washington 98101

Elizabeth Bozomolov, Cochair
Michael R. Yantis Associates, Inc., 1809 7th Avenue, Suite 1609, Seattle, Washington 98101

Contributed Papers

All posters will be on display from 1:00 p.m. to 5:30 p.m. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 1:00 p.m. to 3:15 p.m. and contributors of even-numbered papers will be at their posters from 3:15 p.m. to 5:30 p.m.

4pAAb1. Early sound distribution in auditorium. Jiqing Wang and Guorong Jiang (Inst. Acoust., Tongji Univ., 1239 Sipin Rd., Shanghai 200092, PROC)

It is known that the strength of the audience area of a hall is largely dependent on the early sound, i.e., the direct sound and the first reflections. They are varied not only with reverberation time, but also with the geometry of the room, the absorption arrangement of the room, and the location of the source and(or) the receiver. Results from digital simulation and field measurements in halls show that the attenuation rate of early sound with the distance between the source and the receiver is much larger than ordinary predictions. It is more rational in complying with the subjective judgment. Therefore it should not be ignored in the acoustical design of a hall.

4pAAb2. The late reverberation time as a new criteria for the evaluation of hall acoustics. V. J. Stauskis (Vilnius Gediminas Tech. Univ., Saulėtekio al. 11, 2040 LT Vilnius, Lithuania)

Investigations show that the field decay after-35 dB has its reverberation values, which differ considerably from the values of the standard reverberation time. The experiment was carried out in a rectangular hall measuring $13.6\times10.7\times7.0$ m. The sound absorption coefficients of all planes in the hall were small (0.02–0.05). One hundred seventy semi-upholstered chairs were used for the experiment. When the chairs were placed in the hall, the decrease in the field of a nonfiltered signal was sudden and pronounced, while in the interval from 1500–4000 ms it was slow. The experiments with the filtered signal showed that, at any frequencies, there existed both the initial sudden and the late slow decrease in the sound field energy. The higher the frequency, the longer the interval of the slow decrease in the sound field. When approximating the decrease from -35 to -40 dB, the reverberation time was 15 s at 250 Hz and 4000 Hz. When approximating the decrease from -45 to -50 dB, the reverberation time was as long as 20–21 s within the frequency range of 500 to -2000 Hz.

4pAAb3. Study on acoustic index variations due to small changes in an observation point. Katsuaki Sekiguchi (College of Sci. and Technol., Nihon Univ., Suragadai, Kanda, Chiyoda-ku, Tokyo, 101 Japan, sekiguch@arch.cst.nihon-u.ac.jp) and Toshiki Hanyu (Junior College of Nihon Univ., Narashinodai, Funabashi, Chiba, 274 Japan)

Acoustic index variations due to small changes in an observation point in a concert hall were studied. Three kinds of methods were used in order to measure the acoustic indexes: (1) one point measurement using a single channel microphone; (2) binaural measurement using a dummy head microphone; and (3) the regular tetrahedron peak point method using a 4-channel microphone system. The acoustic index variations in each of the methods were analyzed. In addition, subjective experiments were performed in order to evaluate the effects of the variations. Compared to these measurements, problems with the acoustic indexes which were analyzed by each of the methods were found. One point measurement may incorrectly evaluate a concert hall.

4pAAb4. Effects of the spatial information of a sound field on listener envelopment. Toshiki Hanyu (Junior College of Nihon Univ., Narashinodai, Funabashi, Chiba, 274 Japan, hanyu@arch.jcn.nihon-u.ac.jp) and Sho Kimura (Nihon Univ., Surugadai, Kanda, Chiyoda-ku, Tokyo, 101 Japan)

The spatial information of sound fields in various kinds of concert halls as virtual sound-source distributions and short-term directional diffusion factors were measured. It is clear that the spatial information in a concert hall is characterized by how the reflections surround a listener. Accordingly, the relationship between the spatial information and listener envelopment was studied. First, subjective experiments were performed using a simulated sound field in an anechoic room in order to study the effects of lateral energy ratios on listener envelopment. As a result, the lateral energy ratios were not sufficient to evaluate listener envelopment. The research demonstrates that listener envelopment is not only related to

4p THU. PM

late arriving reflections but also to early reflections. In addition, listener envelopment is related to the spatial distributions of the reflections that surround a listener. Furthermore, the implications for concert hall design were discussed.

4pAAb5. An evaluation method of concert hall acoustics. Y. Hirasawa (Faith, Inc., Form-Gokomachi 583, Gokomachi-Ebisugawa, Kyoto, 604 Japan) and Z. Maekawa (Environ. Acoust. Lab., Oyodonaka, 4-13-11, Osaka, 531 Japan, nd7j-mekw@asahi-net.or.jp)

A lot of objective physical parameters for acoustic evaluation of a concert hall have been proposed. However, no one can believe that they provide a perfect method for designing a concert hall. There is a problem with the measuring method of physical parameters, i.e., almost all acoustical measurements in a hall use an omni-directional loudspeaker, which is not similar to most musical instruments. Also, no parameter specifies tone-color or sonority of music performance. These are important factors in evaluating concert hall acoustics. The remaining problems must be considered as follows: (1) The position of sources on a stage changes tone-color and generating sound power. (2) The directivity of sources gives different effects on room acoustics at the same position on the same stage. (3) Each musical instrument has its own important frequency range. What instrument shall be selected properly and what execution of the instrument shall be adopted for evaluation? (4) What is a suitable program of music for evaluation of hall acoustics? Some examples of the problems upon sources and its performance with using real musical instruments, and the remaining problems for evaluations of the concert hall will be presented.

4pAAb6. Acoustical behavior of churches: Gothic-Mudejar churches. Juan J. Sendra, Teófilo Zamarreño, and J. Navarro (School of Architecture of Seville, Avda Reina Mercedes s/n, 41012 Sevilla, Spain)

Traditionally, a particularly good acoustic behavior has been assigned to Christian churches. But this statement, which is not true at all, has led to important errors in church rehabilitation works, more so in those that implicate changes of use, such as auditoriums or theatrical halls, which is the case of many churches in Spain. This research group has worked in acoustical analysis for more than ten years, and has participated in many cases of church rehabilitation, among which is acoustical behavior. We are talking about a type of church very common in the south of Spain, Gothic-Mudejar churches. They are churches with these common characteristics: relatively little volume, plants with three naves, and wooden ceilings. With this work, significant conclusions drawn from the acoustical analysis of ten churches of Seville that respond to this type are presented. Their characteristics are quite homogeneous, so it is possible to get conclusions about the acoustical conditions of this ecclesiastical type.

4pAAb7. Acoustical design of Sumida Triphony Hall. Toshiko Fukuchi and Hideo Nakamura (Nagata Acoust., Inc., Minami-Shinjuku-Hoshino Bldg., 8F, 5-23-13 Sendagaya Shibuya-ku, Tokyo, 151-0051 Japan, fukuchi@nagata.co.jp)

The Sumida Triphony Hall was opened near Kinshityo JR station, located in the eastern part of Tokyo, in October 1997. The building contains two concert halls, the Main Hall, with 1801 seats, as the home of the New Japan Philharmonic, and the Recital Hall, with 252 seats. The Main Hall has a rectangular shape with the sloped ceiling rising at 12.5 degrees toward the back of the hall in parallel to the main floor and with two balconies around the main audience area. The reflecting panels were hung over the stage to help the ensemble for players. The shape was basically checked with the technique of computer simulation and the detail of the ceiling and the wall was studied by use of a 1/10th-scale model. An organ with 66 stops was installed at the front of the hall. The Recital Hall is a typical ''shoebox'' hall. In order to attenuate the vibration caused by railroad tracks, the vibration isolating barriers were embedded around a part of the outer walls of the building, and also the vibration insulating structure was adopted at the Main Hall partly and the Recital Hall entirely. The outline of acoustical design will be reported.

4pAAb8. Subjective assessment of the uneven distribution of reflections within a concert hall. Eugenio Collados (Dept. of Phys., Univ. of Santiago, Santiago, Chile, ecollado@lauca.usach.cl)

Most physical variables other than RT are known to vary significantly within a concert hall. Namely, early to late energy ratio, lateral energy fraction, and sound strength. Less well known are the subjective correlates to the above variations. A series of experiments were carried out in order to assess the number and meaning of subjective dimensions throughout a single hall. Testing environments were both an anechoic chamber array and an actual hall with artificial sources, where subjects were asked to judge classical music samples. In both cases the standard spatial distribution was modified by the addition of discrete reflections while keeping level and RT constant. At least four subjective dimensions were found, as many as found in hall-to-hall comparisons. Up to 59% of variance was related to the spatial image of the source. After comparing results between the anechoic chamber and the actual hall, the visual impression seems to partially compensate for the variations due to distance. These results raise some doubts on the validity of listening tests performed without a visual stimulus. It is also suggested that these effects should be considered in designing halls with active systems.

4pAAb9. A neural network analysis of concert hall acoustics. Fergus R. Fricke and Young G. Han (Architectural and Design Sci. Dept., Univ. of Sydney, Sydney, NSW 2006, Australia)

A neural network analysis using data from 51 concert halls was undertaken. The analysis related the acoustic quality of halls, as judged by musicians, to ten hall parameters: volume, surface area, number of seats, length, width, height, mean rake angle of seats, a surface diffusion index, stage height, and extent of stage shell/enclosure. The surface diffusion index and the extent of the stage shell were determined by a group of architects who made judgments on the basis of photographs and drawings of the halls. The results of the analysis are tentative and difficult to generalize, as there are so many inputs and so many possible combinations of parameters, and the effect of a particular factor is, in some cases, neither linear nor monotonic. The results are applied to a particular hall, the concert hall of the Sydney Opera House, where changes are being contemplated. There are some unexpected results which, at this stage, give food for thought rather than the basis for action, as there are obvious limitations to this work, such as extraneous factors influencing judgments of acoustic quality, surface diffusion estimation, and extent of the stage shell and the limited number inputs used to describe the halls.

4pAAb10. Directional dependence of the change of auditory source width by very short time-delay reflections. Masayuki Morimoto and Mariko Watanabe (Environ. Acoust. Lab., Faculty of Eng., Kobe Univ., Nada, Kobe, 657 Japan, mrmt@kobe-u.ac.jp)

Barron [J. Sound Vib. **15**, 475–494 (1971)] showed that spatial impression (SI) corresponding to auditory source width (ASW) decreases when the time delay of reflections relative to the direct sound is shorter than 10 ms. This paper investigates the effect of the direction of lateral reflections on the decrease of ASW. The music motif used in this investigation was a 6.5-s section of the 4th movement of Mozart's Jupiter Symphony (No. 41), which was the same motif as Barron used. The sound fields used as stimuli consisted of a direct sound and two discrete reflections with a 1-ms interval between them. The time-delay of the first reflection relative to the direct sound was 2, 4, or 8 ms. The directions of the

reflections were $\pm 18°$, $\pm 45°$, or $\pm 72°$. The experimental results demonstrate that the decrease of ASW depends on the direction of the reflections. As the reflections arrive from a more frontal direction, ASW decreases more sharply relative to the delay time.

4pAAb11. Reverberation time in Serbian Orthodox worship spaces. Miomir Mijic (Faculty of Elec. Eng., Bul. Revolvcije 73, Belgrade, Serbia)

Seven years ago, at the ASA Meeting in November 1990, this author presented some preliminary results of acoustical research realized in the Serbian Orthodox worship spaces. At that time the results for only seven churches were included in the paper. This research has been continued until now and nearly 60 churches were analyzed, different in volume, age, and the architectural style. Such a large number of measurements allow some more general conclusions about acoustical characteristics of orthodox worship spaces. Reverberation times obtained in realized measurements show the important influence of several specific factors such as strong orthodox rules in church interior design and volume shapes. These factors are influenced by traditional architectural styles accepted as unalterable even in the contemporary architectural design. Some differences in the frequency characteristic of the reverberation time according to different historical periods were also discovered.

4pAAb12. An experiment to identify preference groups among concert hall listeners. M. Miklin Halstead, Johan L. Nielsen, and A. Harold Marshall (Acoust. Res. Ctr., The Univ. of Auckland, Private Bag 92019, Auckland, New Zealand, mm.halstead@auckland.ac.nz)

The task of specifying the acoustical behavior of a concert hall depends upon knowing the preferred subjective acoustical characteristics for the intended program material. Several studies have suggested that the listening population might be divided into groups with respect to their preferred acoustic, with different weightings applied to different subjective factors in each. Our project endeavors to identify and describe these groups, and to make their composition an integral part of hall design. This paper describes a subjective experiment designed to identify and measure individual listening preferences in a simulated concert hall setting, and presents the experiments first results. Following from these results, an attempt is made to place tested individuals into preference groups. The applicability of the experiment is considered in terms of the quality of hall simulation and of the impact of the simulator and experimental design on the listeners discrimination.

4pAAb13. On spatial variability of room acoustics measures. Johan L. Nielsen, M. Miklin Halstead, and A. Harold Marshall (Acoust. Res. Ctr., The Univ. of Auckland, New Zealand, j.nielsen@auckland.ac.nz)

The use of high-energy measurement signals combined with digital multichannel recording and postprocessing leads to a highly efficient collection of impulse responses in concert halls. During measurements in two similar concert halls, the Christchurch Town Hall, New Zealand, and the Hong Kong Cultural Centre (both described in companion papers), responses were collected in all seats in some areas and in different positions in some single seats, in addition to the usual sampling of all areas. An omnidirectional loudspeaker source was used, and the measurements were repeated in several source positions. The distribution of various parameters in these areas are compared to the results for the whole hall. Based on this comparison the use of mean and standard deviation as measures of location and spread for the parameters is discussed, and the variation of parameters with varying source locations is demonstrated. Approaches toward the reduction of parameter sensitivity to insignificant details in the early sound field are briefly discussed and illustrated using the measured data.

4pAAb14. Acoustical design of the Queensland Conservatorium of Music. Keiji Oguchi, Yasuhisa Toyota, and Minoru Nagata (Nagata Acoust., Inc., Minami-Shinjuku-Hoshino Bldg., 8F, 5-23-13 Sendagaya Shibuya-ku, Tokyo, 151-0051 Japan, oguchi@nagata.co.jp)

The auditorium of the Queensland Conservatorium of Music was planned for both classical music concerts and lyric performances in the newly constructed conservatorium in the Southbank of Brisbane, Australia. The retractable orchestra shell is moved forward to the proscenium on the stage, and forms a shoebox style concert space with 643 seats. Small doors are installed on the orchestra shell for acoustical fine tuning in the stage space. Absorptive materials appear when the doors are opened. In lyric performances, the orchestra shell is retracted backward. The frontal area of the stage floor is sunk as an orchestra pit. The reverberation time becomes short, from 2.0 to 1.6 s, with retracting the orchestra shell, and varies by 0.3 s with acoustic curtains on the audience side walls for both concert and lyric use. The acoustical design and characteristics of the auditorium and players' preference on the acoustical tuning elements are reported in this paper.

4pAAb15. Estimation and analysis of acoustic parameters of ancient churches for concert performances. Tiziana Ottonello, Enrico Dassori (IATU—Univ. of Genoa, Via Opera Pia, 16145 Genova, Italy), and Andrea Trucco (DIBE—Univ. of Genoa, Via Opera Pia 11A, 16145 Genova, Italy)

This paper presents a study on the propagation of musical sound, where occasionally concert performances are held. The five selected churches were built in Genoa between the eleventh and the seventeenth century. The analysis was aimed at the computing of acoustic parameters (e.g., reverberation time, early decay time, initial decay gap, speech trasmission index, definition index), based on structural modeling and computer simulation (RAMSETE software) as well as experimental texts. Analysis of results showed that romanic churches generally provide better acoustic characteristics. Furthermore, acoustic not standing modifications (e.g., absorbing panels, curtains) were designed in order to make the observed acoustic parameters comparable with those observed in a typical concert hall. This work was developed in the framework of a project aimed at creating new music halls using the existing old buildings, generally dedicated to other aims.

4pAAb16. Acoustic behavior of a tense-structure. Alessandro Cocchi and Lamberto Tronchin (DIENCA–CIARM Viale Risorgimento, 2, 40136 Bologna, Italy, tronchin@ciarm.ing.unibo.it)

During the design of a concert hall, the acoustical behavior of the sound field is often investigated by using numerical models. In the case of a restoration of the hall, or during the design of the orchestra chamber, acoustic measurements of impulse responses in the hall give precious information on the sound fields, e.g., RT, clarity, and so on. By using numerical simulation, it is then possible to verify the efficiency of the acoustic solution. Much commercial software is available, all of them following the geometric assumptions, as the Eikonal equation, while the diffraction and diffusion of sound are not usually taken into account, and all of them require the absorption coefficient of the walls. The tense-structure, on the other hand, is not so easy to model, owing to the peculiar characteristics, i.e., modal behavior and impossibility to measure the absorption of the membrane. In this paper, the design of an acoustic intervention in a tense-structure is shown. A procedure to find out the ''equivalent absorption'' of the membrane is shown. The results of such a model are compared with the experimental measures.

4pAAb17. Variations in the surface textures of choral reflectors. Elizabeth J. Lee and Campbell J. Yule (Acoust. Res. Ctr., Univ. of Auckland, New Zealand, elee@ccul.auckland.ac.nz)

The commission to design a portable choral reflector seemed a simple process of finding a balance between acoustical and structural requirements. Initial inquiries, however, found a lack of information on the acoustical effect of different surface textures normally used. An experiment was developed to assess both the subjective and objective performance of varying degrees of surface roughness. Eight possible surfaces were chosen and applied to standardized panels. The impulse response of each panel was measured in an anechoic room using a maximum length sequence signal. Early results indicate significant differences in the reflected energy and frequency response of the respective panels. An attempt to obtain a subjective evaluation of the reflections by convolving choral and solo voice recordings with the impulse response of the panels will be reported. A prototype choral reflector based on this work will be described. [The support of the Acoustics Research Centre for this project is acknowledged.]

4pAAb18. Acoustical design of Harmony Hall Fukui. Akira Ono, Suzuyo Yokose, and Katsuji Naniwa (Nagata Acoust., Inc., Minami-Shinjuku-Hoshino Bldg., 8F, 5-23-13 Sendagaya Shibuya-ku, Tokyo, 151-0051 Japan, ono@nagata.co.jp)

Construction of Harmony Hall Fukui, which was sponsored by the Fukui prefecture government, was completed on September 3, 1997. The facility contained a large hall, a small hall, and administrative offices. In the central area of the facility is an administrative room, rehearsal rooms, and machinery space. The two halls are situated at the east and west ends of the facility. This architectural zoning brings advantages and savings for effective sound insulation and noise control of the ventilation system. The large hall seats 1456 people and is a concert hall in the shoebox style. The room shape was studied by computer simulation techniques during the design stage, and the details of the wall and ceiling were studied by use of 1/10th scale model experiments before the construction work started. The small hall seats 610 people in an extremely intimate setting. The seating surrounds the stage, and even the height of the balcony seating is staggered to create what may be termed a ''mini-arena'' style. The unique layout heightens the audience's feeling of proximity to the stage. Acoustical design and acoustical characteristics will be presented at the meeting.

4pAAb19. Effects of the modulated delay-time interval of the single reflection on subjective preference. Junko Atagi, Yoichi Ando (Grad. School of Sci. and Technol., Kobe Univ., Rokkodai, Nada, Kobe, 657 Japan, 962t001n@kobe-u.ac.jp), and Yasutaka Ueda (Hazama Corp., Karima, Tsukuba, 305 Japan)

In the past, experimental results of the sound-pressure level (SPL) were obtained which showed that the indoor sound field is regarded as a time-variant system due to an air current. Using the time-variant model represented by the direct sound and the varying delay time of single reflection, an experiment was performed on an effect on subjective preference. This experiment was focused on the effect of the modulation interval (Δ) of the delay time of the single reflection. The delay time of reflection (Δt_1) was adjusted to the most preferred delay time of reflection which was obtained by the experiments for the time-invariant sound field. In the case of fast tempo music, where the minimum value of the effective duration of the autocorrelation function is $\tau_{e_{\min}}=43$ ms, the scale value of subjective preference is higher in a time-variant sound field than a time-invariant one. On the other hand, in the case of slow tempo music, $\tau_{e_{\min}}=127$ ms, such an effect could not be found. It is suggested that a time-variant sound field might have a positive effect on subjective preference in the performance of a fast tempo music.

4pAAb20. Individual differences of subjective preference for sound fields with a different preferred delay time of reflection. Souichiro Kuroki, Ippei Yamamoto (Faculty of Eng., Kagoshima Univ., 1-21-40, Korimoto, Kagoshima, 890-0065 Japan), Hiroyuki Sakai, and Yoichi Ando (Kobe Univ., Rokkodai, Nada, Kobe, 657-8501 Japan)

The purpose of this study is to clarify the relationship between the individual preferred delay time of reflection and the effective duration of the autocorrelation function (ACF) of sound source. Subjective preference tests were conducted by varying the delay time of reflection (Δt_1) which is one of the four objective parameters related to subjective preference of sound fields. In this paper, five dry sources of instrumental music with different effective duration are used. In addition to the global preference with a number of subjects, individual-preference-scale values also were obtained by the method of paired-comparison tests. An attempt is made here whether or not individual differences in subjective preference may be explained by the categorical classification of musical experience, sex, and age of subjects.

Session 4pAAc

Architectural Acoustics: Speech Intelligibility

Dennis Noson, Chair
107 North 77th Street, Seattle, Washington 98103

Chair's Introduction—4:00

Contributed Papers

4:05

4pAAc1. Modeling of acoustic parameters and speech intelligibility in long enclosures. Lening Yang and Bridget M. Shield (School of Eng. Systems and Design, South Bank Univ., London SE1 0AA, UK, shieldbm@sbu.ac.uk)

Ray tracing computer models have been developed for the prediction of the sound field and speech intelligibility in long enclosures of any cross-sectional shape. The models for enclosures of circular and rectangular cross section are particularly suitable for the prediction of speech intelligibility in underground railway stations. The model for use in long enclosures of rectangular cross section has been validated using data measured in a real underground station in Hong Kong. Measurements of acoustics parameters including sound propagation, early decay time, clarity index C50, and center time D50 were made along the length of the platform, with a single source towards one end of the platform. The model correctly predicts the acoustic parameters, the predictions being particularly accurate at midfrequencies. In addition the speech transmission index is accurately predicted at all frequencies, especially in the far field of the source.

4:25

4pAAc2. Relationship between speech transmission index and easiness of speech perception in reverberatory fields. Hiroshi Sato, Hiroshi Yoshino (Dept. of Architecture, Tohoku Univ., Aramaki Aza Aoba, Aoba-ku, Sendai, 980-77 Japan), and Muneshige Nagatomo (Kajima Tech. Res. Inst., 19-1 Tobitakyu 2-Chome, Chofu-shi, Tokyo, 182 Japan)

When evaluating speech intelligibility in rooms, the Tri-syllable articulation test is used as one of the standard indicators in Japan. Sometimes there are differences between auditory impression and the score of the articulation test. So the psychological scale values of easiness of speech perception (ESP) were measured by experiments, which indicated differences between ESP and articulation scores and a good correlation ($R=0.9$) between ESP and speech transmission index (STI) in reverberation. The purpose of this paper is to point out relationships between the condition of sound field and ESP. The first of two experiments in this study was done in reverberatory fields. Sound-pressure level (SPL) of direct sound, direct sound to first reflection ratio (D/R), initial delay time, frequency characteristics of reverberation, and reverberation time (RT) were varied. The other experiment was done in simulated fields of actual auditoriums. The result of the first experiment shows D/R and RT has alarge effect on ESP and while varying D/R, the correlation between ESP and STI is not good ($R=0.6$). The second experiment indicates that when the sound source is on the center of the stage, the correlation between STI and ESP is 0.8; and a different tendency is found while using the sound reinforcement system.

4:45

4pAAc3. Investigation of the speech intelligibility in the dome space. Satoshi Inoue, Masahiro Katoh, Kiyoshi Sugino (Inst. of Technol., Tokyu Construction Co., Ltd., 3062-1 Tana, Sagamihara, Kanagawa, 229-11 Japan, ino@etd.tokyu-cnst.co.jp), and Hiroyuki Imaizumi (Natl. Inst. for Resources and Environment, 16-3 Onogawa, Tsukuba, Ibaraki, 305 Japan)

In order to investigate the acoustical characteristics of an underground dome (2 320 m^3), objective and subjective properties were tested and the mutual relationship of them was examined. Then to improve the speech transmission quality there, some simple absorption treatments were given. A series of experiments, carried out under different acoustical conditions, has confirmed the effects of the treatment. The main results are as follows: (1) In the dome space without absorption treatment, the differences of some objective measures and subjective responses (speech intelligibility) were great in places where observation was made, e.g., RASTI value was distributed over a range of 0.17–0.59. (2) It is indicated that the speech transmission quality was high at the observation points where the energy, arriving in the first 0.06 s after the arrival of the direct sound, is predominant, and others are almost low. (3) Among the physical measures, EDT, early/late-arriving sound energy ratio, ts (at 1 kHz octave-band), and RASTI were correlated well with speech intelligibility. (4) Rubber tiles on the floor, 12-m cotton canvas hung from the ceiling, and glass wool board were found to be effective in decreasing reverberation time and diffusing the sound. Also, this treatment was useful in improving the speech intelligibility remarkably.

5:05

4pAAc4. Calculation of speech intelligibility using four orthogonal factors involved in autocorrelation functions. Tetsuichi Shoda and Yoichi Ando (Grad. School of Sci. and Technol., Kobe Univ., Rokkodai, Nada, Kobe, 657 Japan)

In this paper, a method of calculating speech intelligibility for the sound field with the direct sound and the first reflection is proposed. This method is based on the model of an auditory-brain system consisting of the autocorrelation mechanisms which is equivalent to the power spectrum. The autocorrelation function (ACF) may play an important role on the speech information processing. Four factors may be extracted from ACF, such that (1) τ_e : the effective duration of ACF (the ten percentile delay) meaning the repetitive feature of signals; (2) τ_1 : the delay time of the first peak of ACF signifying the pitch of signals; (3) ϕ_1 : the amplitude of the first peak indicating the strength of the pitch; and (4) $\Phi(0)$: the power of the signal frame. The distance between the syllable in sound fields and the syllable of the direct sound were calculated by the use of these four factors. Using these distances and its linear combination, speech intelligibility of the syllables in the sound fields may be calculated. Results of speech intelligibility tests as a function of the delay time of the reflection are in good agreement with the calculated values.

Session 4pAO

Acoustical Oceanography and Animal Bioacoustics: Acoustics of Fisheries and Plankton III

Orest I. Diachok, Chair
Naval Research Laboratory, Washington, D.C. 20375

Contributed Papers

2:00

4pAO1. Variation in acoustically measured abundance from repeated surveys of an isolated herring population. I. Huse and M. Ostrowski (Inst. of Marine Res., P.O. Box 1870, N-5024 Bergen, Norway)

An isolated population of herring was surveyed acoustically a total of 18 times while wintering in a fjord in northern Norway. Total abundance and variograms were computed for each survey. Three-dimensional distribution graphs are presented. The results are discussed in terms of total variation of acoustic survey estimates.

2:15

4pAO2. Acoustic estimates of zooplankton and micronekton biomass using an ADCP. Patrick H. Ressler, Douglas C. Biggs, and John H. Wormuth (Dept. of Oceanogr., Texas A&M Univ., College Station, TX 77843-3146, pressler@ocean.tamu.edu)

A calibrated 153-kHz narrow-band ADCP (Acoustic Doppler Current Profiler) has been used to collect acoustic backscatter intensity (Sv) data during several cruises in the Gulf of Mexico. Data have been gathered both while on station and while underway along transects through different hydrographic regimes, enabling an examination of both temporal and spatial trends in backscatter. In addition, zooplankton and micronekton stocks were intensively sampled with a 1 $\hat{m2}$ MOCNESS (Multiple Opening-Closing Net Environmental Sensing System). Empirical correlations between spatial and temporal variations in Sv and in standing stocks of zooplankton/micronekton will be presented. Sv measured with an ADCP may be useful as an index of zooplankton and micronekton biomass. The location and abundance of these food stocks are hypothesized to be important in determining the distribution and abundance of marine cetaceans in the Gulf of Mexico. In addition, Sv-derived estimates of such stocks may allow inferences about secondary biological productivity in the Gulf, and the role that mesoscale circulation features might play in driving large-scale patterns. [This ongoing research is supported by the USGS Biological Resources Division and the US Minerals Management Service under USGS BRD Contract No. 1445-C109-96-004. For more information, see http://www.tamug.tamu.edu/gulfcet.]

2:30

4pAO3. Monitoring the growth and mortality rates of pelagic fish with absorption spectroscopy measurements. Orest Diachok (Naval Res. Lab., Washington, DC 20375) and Paul Smith (Southwest Fisheries Sci. Ctr., La Jolla, California, 02038)

The experimental geometry for absorption spectroscopy measurements, which was employed during Modal Lion, can be adapted for monitoring the growth and mortality rate of pelagic fish. During the juvenile stage, in which sardines grow from 3.5 to 13.5 cm over a period of about 1 year and mortality rates are relatively high, absorption spectroscopy measurements (together with complimentary biological and oceanographic measurements) conducted for a few days approximately once per month would permit investigation of the environmental determinants of these parameters. Measurement sites would have to follow the migration route of juvenile sardines over a period of about 1 year. To minimize cost and simplify logistics, a broadband omnidirectional source (instead of a parametric source) would be suspended from a moored ship or buoy. A relatively low source level ($\sim$170 dB) would be adequate. Simulations will be presented of the temporal evolution of the resonance frequency, absorption coefficient, and transmission loss due to sardines between 0.6 and 6.0 kHz. Simulations will illustrate the sensitivity of transmission loss to source and receiver depth, and the average depth and thickness of absorbing layers. In addition, calculations that illustrate the potential of estimating the depth and thickness of absorbing layers by varying the depth of an omnidirectional source will be presented. [This work was supported in part by the Office of Naval Research.]

2:45

4pAO4. Measurements of snapping shrimp colonies using a wideband mobile passive sonar. Marc Olivieri, Stewart A. L. Glegg, and Robert K. Coulson (Ctr. for Acoust. and Vib., Dept. of Ocean Eng., Florida Atlantic Univ., 777 Glades Rd., Boca Raton, FL 33431)

A wideband passive sonar has been developed to study ambient noise levels in coastal regions. During the past two years a large number of measurements have been made with this system in shallow water off Boca Raton, Florida. The results have demonstrated that the noise from snapping shrimp dominates in this area during low sea states. Localization of the colonies has been possible by moving the sonar to different locations over the period of a few hours. Typically the shrimp are clustered close to man-made structures at or near the shore line, with additional smaller colonies on nearby coral reefs. To speed up the data collection the system has been mounted to an autonomous underwater vehicle and on-board intelligence is being developed to optimize the survey locations in real time. Results from a demonstration experiment will be presented. [Work supported by ONR.]

3:00

4pAO5. Using sound to map fish spawning: Determining the seasonality and location of spawning for weakfish and red drum (Family Sciaenidae) within Pamlico Sound, NC. Joseph J. Luczkovich (Dept. of Biol. and Inst. for Coastal and Marine Resources, East Carolina Univ., Greenville, NC 27858 luczkovichj@mail.ecu.edu), Stephen E. Johnson, and Mark W. Sprague (East Carolina Univ., Greenville, NC 27858)

Fish produce species-specific sounds during courtship, during aggressive encounters with other fish of the same species, and as a response to threats from predators. Particular sounds are species-specific and have been recorded from fish during spawning in captivity. Such acoustical data may be used to identify and locate fish that are reproductive in the sea. In this paper, nocturnal sound production and planktonic egg production are correlated for weakfish (*Cynoscion regalis*) and red drum (*Sciaenops ocellatus*) (Family Sciaenidae), two species for which there are concerns over declining stocks. Recordings of underwater sounds and ichthyoplankton net tows were made after sunset at ten sites in Pamlico Sound, NC. Sites were relocated with a differentially corrected Global Positioning System receiver on a biweekly basis from May through October 1997. Based on laboratory studies, weakfish ''purring'' sounds and red drum ''knocking'' sounds only occur during spawning. ''Purring'' was recorded along with sciaenid eggs at stations near Ocracoke and Hatteras Inlets during May through July. ''Knocking'' was recorded in September and October at sites away from the inlets, also occurring with sciaenid eggs in one case. Mapping of sciaenid spawning areas may be possible using acoustical data alone, but may overestimate egg production.

3:15

4pAO6. Using fish sounds to identify spawning activity of weakfish (*Cynoscion regalis*) and red drum (*Sciaenops ocellatus*) in nature.
Mark W. Sprague (Dept. of Phys., East Carolina Univ., Greenville, NC 27858, spraguem@mail.ecu.edu), Joseph J. Luczkovich (Inst. for Coastal and Marine Resources, Greenville, NC 27858), and Stephen E. Johnson (East Carolina Univ., Greenville, NC 27858)

In the coastal systems of the southeastern United States, populations of weakfish (*Cynoscion regalis*) and red drum (*Sciaenops ocellatus*) are in decline, and knowledge of their spawning areas is important for the formulation of fisheries management plans. Weakfish and red drum, both members of Family *Sciaenidae*, use their swim bladders to produce species-specific sounds associated with spawning activity. Large spawning aggregations of these fish can produce sound levels as high as 145 dB (re:1 μPa), and these sounds can be used to identify spawning areas. Recordings and spectrograms of spawning sounds will be presented for each species. Water depth, bottom type and contour, sound-speed gradient, and water current gradient all affect the propagation of the fish sounds through the water. Measurements of these factors and the sound level of the fish calls are used to obtain an approximate range to the spawning aggregation. [Work supported by North Carolina Division of Marine Fisheries and U.S. Fish and Wildlife Service.]

THURSDAY AFTERNOON, 25 JUNE 1998 EAST BALLROOM B (S), 12:50 TO 3:45 P.M.

Session 4pBV

Bioresponse to Vibration/Biomedical Ultrasound: Applications of Microbubble Based Echo Contrast Agents II

Nico De Jong, Cochair
Erasmus University, Room Ee2302, P.O. 1728, 3000DR, Rotterdam, The Netherlands

Inder Raj S. Makin, Cochair
Ethicon Endo-Surgery, 4545 Creek Road, Cincinnati, Ohio 45242

Invited Papers

12:50

4pBV1. Quantification of myocardial blood flow using ultrasound-induced destruction of microbubbles administered as a constant venous infusion. Sanjiv Kaul (Univ. of Virginia, Charlottesville, Virginia)

It was hypothesized that if microbubbles are administered as a continuous infusion, then destroying them in the myocardium and measuring their myocardial reappearance rate at steady state will provide a measure of mean myocardial microbubble velocity. Conversely, measuring their myocardial concentration will provide an assessment of microvascular cross-sectional area (CSA). Myocardial blood flow (MBF) can then be calculated from the product of the two. Accordingly, experiments were performed in 21 dogs where MBF was altered mechanically or pharmacologically in the absence or presence of coronary stenosis. Microbubbles were delivered as a constant infusion, and two-dimensional ultrasound was performed using different PI. The myocardial video intensity (VI) versus PI plots were fitted to an exponential function: $y = A(1 - e^{-\beta\tau})$, where A is the plateau VI reflecting the microvascular CSA, and β reflects the rate of rise of VI and hence, microbubble velocity. Excellent correlations were found between flow and β, as well as the product of A and β. It is concluded that MBF can be quantified by destroying microbubbles with ultrasound during a venous infusion.

1:10

4pBV2. Gas-filled liposomes as ultrasound contrast agents for blood pool, thrombus-specific and therapeutic applications.
Evan C. Unger (Dept. of Radiology, The Univ. of Arizona, Tucson, AZ 85724-5067, eunger@vms.ccit.arizona.edu), Thomas McCreery, Dekang Shen, GuanLi Wu, Robert Sweitzer, and Qui Wu (ImaRx Pharmaceutical Corp., Tucson, AZ)

Our group has developed technology for stabilizing microbubbles with phospholipid coatings (Aerosomes®-ImaRx Pharmaceutical Corp.). The first agent (MRX-115) is based upon lipid-coated microspheres filled with perfluoropropane gas and is in phase III clinical trials (October, 1997) for radiology and cardiology applications. Myocardial perfusion studies show the potential for the agent to detect ischemia in patients with myocardial infarcts. Targeting ligands have been covalently bound to lipids and incorporated into the stabilizing shells on the microbubbles. The first targeted agent planned for clinical trials is MRX-408 (perfluorobutane gas, linear hexapeptide-RGD analog). *In vitro* studies show enhanced visualization of thrombi, and *in vivo* studies in dogs with arterial and venous thrombi show selective enhancement, even in animals injected with heparin. *In vitro* studies have been performed to test sonothrombolysis comparing MRX-115 to MRX-408 with and without urokinase. The targeted contrast agent MRX-408 increases the rate of clot dissolution with ultrasound. Targeted, tissue-specific agents are feasible for ultrasound. A thrombus-specific ultrasound contrast agent has the potential to improve clot characterization and detection as well as clot lysis.

1:30

4pBV3. Targeted acoustic contrast agents: New opportunities for ultrasound in medical diagnosis and therapy. Gregory M. Lanza (Div. of Cardiology, Campus Box 8220, Barnes-Jewish Hospital, Washington Univ., 216 S. Kingshighway Blvd., St. Louis, MO 63110, greg@soundlab.wustl.edu), Kirk D. Wallace, Rebecca L. Trousil, James G. Miller (Washington Univ., St. Louis, MO), James H. Rose (Iowa State Univ., Ames, IA), Patrick J. Gaffney (NIBSC, Herts, UK), Dana R. Abendschein, Christopher S. Hall, Michael J. Scott, Christine A. Lorenz, Ralph Fuhrhop, and Samuel A. Wickline (Barnes-Jewish Hospital, Washington Univ., St. Louis, MO 63110)

For 30 years, medical professionals have sought the ''magic bullet'' which would allow sensitive detection of disease and facilitate localized administration of potent chemotherapeutic agents without systemic toxicities. Ligand targeted ultrasonic contrast agents have the potential to improve diagnostic sensitivity, port drugs to specific pathologic tissues, and provide dosimetry of the delivered therapeutics. The authors have recently developed a ligand directed, lipid encapsulated perfluorocarbon emulsion particle (250-nm diameter). These particles, when bound to biological surfaces, strikingly increase the acoustic reflectivity, although the particles have low inherent echogenicity. Contrast enhancement of the reflectivity of blood thrombi, fibrin clots, and nitrocellulose membranes has been tentatively explained by a transmission line model. Intravenous injection of the perfluorocarbon emulsion particles has localized and acoustically enhanced remote thrombi which were otherwise difficult to detect. The contrast agent has a 1-h circulatory half-life which greatly facilitates successful systemic targeting. The small particle size of these emulsions affords access to tiny capillary beds and vascular endothelial disruptions such as those following angioplasty. Collectively, these studies suggest perfluorocarbon contrast agents can target molecular epitopes within and proximate to the systemic circulation. These agents are creating new and exciting opportunities for ultrasound in medical diagnosis and therapy.

Contributed Papers

1:50

4pBV4. Disruption of contrast agents for monitoring blood flow. J. Brian Fowlkes (Dept. of Radiology, Univ. of Michigan, Medical Ctr., Kresge Res. Bldg. III R3315, Ann Arbor, MI 48109-0553, fowlkes@umich.edu), Richard T. Rhee, David W. Sirkin, and Paul L. Carson

An experimental system is being developed for controlling the flow of ultrasound contrast agent to create abrupt changes in contrast levels in tissues. Such changes can be used to monitor blood flow through bolus passage and to identify specific feeder vessels. Focused ultrasound (2.25 MHz) has been used to interrupt the flow of the contrast agent MRX-115 (ImaRx Pharmaceuticals, Tucson, AZ) in a tube (6 mm diameter, average velocity of 9.45 cm/s) using short bursts 20 cycles in duration with a pulse repetition frequency (PRF) of 0.75–6.0 kHz. Peak rarefactional pressures (PRP) for effective disruption of contrast agent were only 0.6–1.2 MPa. This corresponds to mechanical indices (MI) of 0.4–0.8 and intensity values which are well within the recommended limits for diagnostic ultrasound, up to 90% of contrast agent signal was eliminated and positive boluses were produced by turning the field off and on. Similar results have been achieved with a 1.8-MHz *in vivo* system in a rabbit model. Therefore the potential exists to use common diagnostic ultrasound to produce contrast disruption suitable for short bolus production. [Work was supported in part by the U.S. Army under Grant No. DAMD17-94-J4144.]

2:05

4pBV5. Effects of pulse width, phase, and intensity on echoes from ultrasound contrast agents. Karen E. Morgan, Paul A. Dayton, Katherine W. Ferrara (Dept. of Biomed. Eng., Univ. of Virginia, Box 377, Health Sci., Charlottesville, VA 22908, kej6u@virginia.edu), Alexander L. Klibanov, and Gary H. Brandenburger (Mallinckrodt, Inc., Hazelwood, MO 63042)

Contrast-assisted imaging shows great promise for the imaging of microvascular perfusion, although current techniques cannot differentiate the echoes from tissue and ultrasound contrast agents. Harmonic schemes for imaging with these agents have limited spatial resolution due to the transmission of a narrow-band pulse. The potential for wideband contrast-assisted imaging through a systematic evaluation of the effects of imaging parameters on the received echoes is evaluated. Our results indicate that the echo intensity received from contrast agents following wideband transmission is at least as large as or larger than that for narrow-band transmission, with both exhibiting nonlinear increases with increasing transmitted signal intensity. While the echo intensity at harmonic multiples of the bubble resonant frequency can be larger for narrow-band insonation, echoes received after wideband insonation demonstrate a broadband spectrum with significant amplitude over a very wide range of frequencies. The authors also demonstrate that the transmission of wideband high intensity pulses produces consistent time domain echoes from microbubbles. These echoes are unchanged whether the microbubbles break or remain intact over several pulses. Additionally, if the phase (order of compressional and rarefactional half-cycles) of the transmitted pulse is changed, the polarity of the received signal changes accordingly for a stationary reflector but remains unchanged for a microbubble.

2:20–2:30 Break

2:30

4pBV6. Characterization of ultrasound-contrast agents by short-burst excitation. Oliver D. Kripfgans (Dept. of Radiology, Univ. of Michigan, Medical Ctr., Kresge Res. Bldg. III R3315, Ann Arbor, MI 48109-0553, oliver.kripfgans@umich.edu), J. Brian Fowlkes, and Paul L. Carson

A two-frequency method has been adapted to the use of short ultrasonic bursts for the determination of bubble size. The advantage of a burst method is threefold. The overall energy transfer into the volume of interest is smaller as compared to cw. Second, it allows simple methods for improving spatial resolution. Finally, only one transducer will eventually be required in a pulse-echo mode. The experimental setup is designed to fully compensate for transducer characteristics and sample artifacts such as attenuation through window material and contrast agent solution. This leads to a constant sound pressure exposure for contrast agent in the focal volume. The data acquisition employs the frequency tracking procedure where spectra for narrow-band excition are analyzed for fundamental, second harmonic, and sidebands. A technique of nonconstant imaging frequency is used to avoid ambiguities due to coincidences between higher harmonics of the fundamental and multiples of the sideband frequencies. Measurements with shell-encapsulated microbubbles showed the behavior demonstrated by previous cw methods. Higher harmonics as well as sidebands were present due to the nonlinear dynamics of the contrast agents. [This work was supported in part by USPHS Grant Nos. RO1 DK42290 and 1RO1 HL54201.]

2:45

4pBV7. Acoustic detection of microbubble destruction in gaseous contrast agents. William T. Shi and Flemming Forsberg (Dept. of Radiol., Thomas Jefferson Univ., Philadelphia, PA 19107)

The destruction of microbubbles from ultrasound contrast agents has recently received much attention because the disintegration can be used to create new imaging modalities and may produce adverse bioeffects as

well. Both passive and active acoustic detection techniques were employed in our experiments to investigate the collective and individual behaviors of contrast microbubbles at different insonification levels. With the passive technique, the oscillation and destruction of microbubbles were studied by characterizing the waveforms of scattered acoustic signals and analyzing the harmonic and noise contents in the spectra of the signals. It was observed that discrete subharmonics and superharmonics (up to the 20th order) were generated at low acoustic pressures. When the bubble destruction occurred at high acoustic pressures, the noise level in the spectra was raised and the discrete harmonic structure in the higher frequency domain reduced to broadband noise. The noise spectrum in the higher frequency range was found to broaden as the acoustic pressure increased. A modified ACD (acoustic cavitation detector) with improved spatial resolution was utilized to study the disintegration process of an individual microbubble. The comparison of results from passive and active detections will be presented.

3:00

4pBV8. Acoustic and system parameters affecting destruction of ultrasound contrast agents. Peter P. Chang, Inder Raj S. Makin, and Lawrence A. Crum (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, pchang@apl.washington.edu)

In order to effectively use ultrasound contrast agents (UCAs), it is necessary to tune the ultrasound scanner parameters to a particular agent or change the characteristics of the UCA to target-specific applications. Both efforts are being sought by ultrasound manufacturers and pharmaceutical companies. The destruction of UCAs is usually undesirable, although it can sometimes be used advantageously such as in harmonic imaging mode. In the present work, changes in acoustic and system parameters such as frequency, transmitted acoustic pressure, pulse repetition frequency, flow rate, diluting media, concentration, and type of contrast agents were measured at both first and second harmonic. A prototype Doppler system previously described [Chang *et al.*, Ultrasound Med. Biol. **22**, 1205–1214 (1996)] has been adapted to carry out these measurements. Results with Albunex show that bubble destruction increases with an increase in flow rate, acoustic pressure, PRF, and decreases with an increase in the concentration of the contrast agent and oxygen content in the diluting medium. In conclusion, the rupture of UCAs are influenced by several imaging parameters. The threshold values when destruction of the agents occurs should be known in order to optimize the use of UCAs in clinical settings. [Work supported by DARPA-TRP.]

3:15

4pBV9. Observations of insonified contrast agents *in vitro* and *in vivo*. Paul Dayton, Karen Morgan, Margaretta Allietta, Kathy Ferrara (Univ. of Virginia, Box 377 HSC, Charlottesville, VA 22908), Alexander Klibanov, and Gary Brandenburger (Mallinckrodt, Inc., Hazelwood, MO 63042)

Ultrasound contrast agents, which may include a gas core and lipid or albumin shell, play an increasingly important role in medical imaging. However, a complex set of phenomena have limited quantitative evaluation of tissue perfusion based on echoes from these microbubbles. With the use of optical microscopy, mechanisms of microbubble destruction during insonation are evaluated, as well as the effects of ultrasonic radiation force. Depending on conditions such as acoustic intensity and dissolved gas concentration in the media, several mechanisms can change the microbubble and promote its destruction during insonation. Slow gas diffusion from the bubble is observed with low-intensity transmission, but with several pulses at a higher intensity, the albumin shell may weaken sufficiently to release the gas core. It is demonstrated that the primary acoustic radiation force has a significant effect on microbubbles when the ultrasound is transmitted with a low acoustic pressure and high pulse repetition frequency. Deflection of the contrast agent streamline to the wall of a small vessel is shown both *in vitro* and *in vivo*. Such manipulation of the contrast agents may be desired in drug delivery applications, as it provides a method of concentrating bubbles in a specific area.

3:30

4pBV10. Ultrasound contrast agent enhances vascular damage in mouse intestines. Richard A. Gies and Douglas L. Miller (P7-53, Battelle Pacific Northwest Natl. Lab., P.O. Box 999, Richland, WA, doug.miller@pnl.gov)

Either Albunex®, a gas-body based contrast agent, or 5% albumin in saline without gas bodies was administered to anesthetized hairless mice at 10 ml/kg by retro-orbital injection. The mice then were exposed at the midsection to 1.09 MHz of ultrasound for 100 s, either continuous (cw) or pulsed (pw, 10 μs pulses repeated at 1 kHz), in a 37 °C water bath. After exposure, the intestines were examined to count petechiae and hemorrhages into the intestinal lumen. cw thresholds for statistically significant petechiae production were 0.7 MPa spatial peak pressure amplitude without added gas bodies, and 0.5 MPa with added gas bodies. At 0.7 MPa the mean counts (five mice) were 22 without and 120 with added gas bodies. A statistically significant number of hemorrhages occurred at 0.7 MPa (three per mouse) with the contrast agent. The pw thresholds with added agent were 1.4 MPa for petechiae and 2.8 MPa for hemorrhage. At 2.8 MPa, average counts were 227 petechiae and 2.4 hemorrhages. No statistically significant vascular damage occurred for pw exposure up to 2.8 MPa without contrast agent. In terms of mechanisms, the effects appeared to be partly thermal for cw, but primarily associated with ultrasonic cavitation for pw exposure. [Work supported by NIH CA42947.]

Session 4pEA

Engineering Acoustics: Applications of Acoustic Technology

Robert D. Finch, Chair
Cullen College of Engineering, University of Houston, Mechanical Engineering, Houston, Texas 77204-4792

Contributed Papers

1:00

4pEA1. Determination of the noise attenuation of hearing protectors by numerical modeling of the outer ear. Samir N. V. Gerges and Elizabete Y. Bavastri (Federal Univ. of Santa Catarina Mech. Eng. Dep.-Acoust. and Vibration Lab., Cx. P. 476, Florianópolis, SC, Brazil, gerges@mbox1.ufsc.br)

This paper will address the important practical issues of hearing protectors, used in industry, to protect the workers from high noise levels. Comments on the difficulties for the measurements of hearing protector's noise attenuation is discussed. A new work on the numerical modeling of the outer ear canal, considering the eardrum acoustic impedance, using the finite elements (FEM), infinite FEM for the quantification of the noise attenuation of the protectors is presented. This numerical model considers the geometry of the outer ear, outer ear canal, and the eardrum acoustic characteristics. This numerical model can serve as a quick and low cost tool for the optimization of the protector project and the investigation of the effect of different parameters such as protector insertion, effect of leakage, materials, and others on the protector noise attenuation.

1:15

4pEA2. Improved TV/radio listening for hearing impaired. J. G. H. van Zutphen (Tech. Univ. Twente and Audiol. Dept. of the Free Univ. of Amsterdam, The Netherlands), W. F. Druyvesteyn (Philips Res. Labs., Eindhoven, The Netherlands), and C. H. Slump (Tech. Univ. Twente, The Netherlands)

In case different persons, having different preferences for the loudness level, want to listen to one sound program in one room, problems arise concerning the loudness of the reproduced program. An example: person A has a normal hearing threshold, while person B is hearing impaired. A sound reproduction system, consisting of a small array of loudspeakers at the TV/radio can solve this problem with digital signal processing; the directivity pattern of the array is manipulated such that the main lobe is directed to person B. The investigation and the presentation at the conference contains two parts: (1) physical acoustics: design and testing of the loudspeaker array including digital processing, and (2) subjective evaluation with normal-hearing subjects and hearing impaired subjects. The subjects listen (and view) a TV program consisting of a selection from 120 short spoken sentences. The normal-hearing subjects adjust the loudness level at their listening position. The hearing-impaired subjects test the effect of the array. These listening tests show a clear improvement in the speech intelligibility for the hearing-impaired subjects as expressed in the number of correct understood sentences and in subjective terms as ''easier to understand'' or ''less tiring to listen to.''

1:30

4pEA3. Piezoelectric transducer for a hearing aid using PZT thin film. Hidehiko Yasui, Minoru Kurosawa, Takeshi Morita, Takefumi Kanda, and Toshiro Higuchi (Dept. of Precision Machinery Eng., Grad. School of Eng., Univ. of Tokyo, 7-3-1 Hongo, Bunkyo-ku, Tokyo, 113 Japan, hide@intellect.pe.u-tokyo.ac.jp)

A piezoelectric transducer using a PZT thin film deposited by a hydrothermal method was fabricated for the investigation of ability for an earphone. The PZT material can produce sounds efficiently from an electrical signal. The thin film technology is suitable for the fabrication of a small transducer which is worn in the ear. Especially, the hydrothermal method has merits such as thick film fabrication and possibility of deposition on a round surface. Therefore we can design and fabricate a diaphragm for a small earphone which produces a high enough sound-pressure level for people who have an obstacle for hearing ability. The fabricated transducer had a bimorph construction. The base material was titanium foil 20-μm thick. Both sides of the titanium, PZT thin films 10-μm thick were deposited. The diameter of the transducer was 6 mm. We investigated the ability of the fabricated transducer. Displacement amplitude of the transducer was about 0.9 μm per 5 V driving voltage below the resonance frequency.

1:45

4pEA4. Low-frequency circuit noise in hearing aids. Vishakha W. Rawool (Commun. Disord. and Special Education, Bloomsburg Univ., Bloomsburg, PA 17815)

Some hearing impaired listeners complain about the circuit noise in hearing aids. This is specially true for individuals with high-frequency hearing impairments who have normal hearing in the lower frequencies. In this investigation, the equivalent input noise levels were measured in several different hearing aids: linear, wide dynamic range compression, high-level compression, programmable, and digital aids. For the linear, programmable, and digital aids the measurements were made with two types of frequency responses. One was suitable for a flat moderate hearing impairment and another for a high-frequency hearing impairment. For the linear and compression aids, the noise levels were measured by using two different volume control settings. All the measurements were made by using 11 different special purpose average frequencies ranging from 200 to 5000 Hz. The results suggest generally higher circuit noise in the lower frequencies in all the hearing aids. Changes in volume control settings showed no differences in noise levels. For the programmable aids, the flat versus high-frequency configurations yielded similar noise levels. For the linear and digital aids, the high-frequency configuration yielded higher noise levels in the lower frequencies when compared to the flat configurations.

2:00

4pEA5. High degree of freedom muffler optimization using genetic algorithms: Experimental verification. Siska Pottie and Dick Botteldooren (Information Technol. (INTEC) Univ. Gent st-Pietersnieuwstraat 41 B-9000 Gent, Belgium, pottie@intec.rug.ac.be)

This paper reports on an experimental verification of a nonconventional muffler design. The optimal design is searched using a genetic algorithm (GA). GAs are a powerful optimization technique that are well suited to handle multiparameter optimization problems. The degrees of freedom used to describe the structure is far beyond the usual set. The muffler has a fixed length and width and is described by a set of design rules defining the use of the available space. Different coding alternatives for the GA were analyzed. Best results are obtained with a coding that describes the position and size of the design rules. The performance of the mufflers is characterized by the theoretically achieved sound reduction and by the ease of construction of the optimized muffler. The theoretically achieved sound reduction is calculated using a time domain simulation. In

order to verify the promising theoretical performance of the optimized mufflers, a good performing design is built and tested in the laboratory. General agreement between the theoretical and experimental results is found. Since no conventional muffler design rules are imposed, the optimization is completely open for new approaches which can led to new insights for conventional muffler design.

2:15

4pEA6. Theoretical analysis of reactive silencers with two propagating modes. J. Kergomard and M. Pachebat (Lab. Acoustique, Universite du Maine, Av. O. Messiaen, 72017 Le Mans Cedex 9, France, kergo@laum.univ-lemans.fr)

A general modeling of sound ducts with reactive (or dissipative) treatments considers one main guide and a secondary one (the treatment), with another propagation constant, including discontinuities, coupled to the main one by perforations, i.e., a shunt impedance and a series impedance [see Kergomard *et al.*, Acta Acustica **2**, 1–16 (1994)]. For a periodical system the calculation can be made analytically for a succession of N cells by diagonalization: the impedance matrix of the portion of the duct with treatment can be deduced. At low frequencies only one mode propagates in each duct and the matrices are of fourth order. From the boundary conditions for the treatment the insertion loss (IL) for the main duct is deduced. Examples are an array of coupled resonators, a lining with perforated facing, and a perforated tube muffler with partitions. A conclusion is the proof of the interest of the interferences between two propagating modes of the periodical system: resonances occur and IL is proportional to $\log N$. It is more efficient than a single propagating mode (e.g., in expansion chambers). Compared to the evanescent case (IL proportional to N), it is less efficient but the frequency band is wider. Many facts of practical interest can be deduced.

2:30

4pEA7. Observations of characteristic transmission and reflection coefficients of a splitter duct attenuator. M. Terao and H. Sekine (Dept. of Architecture, Kanagawa Univ., Kanagawaku, Yokohama, 221 Japan)

The characteristic transmission coefficients (including the reflection coefficients) of a splitter duct attenuator with two air channels containing higher-order wave modes were investigated. These were evaluated at interfaces taken in the straight duct sections in both sides of the attenuator and were determined experimentally and numerically (by BEM). The determination method introduced here requires the solution of two sets of the simultaneous equations given by the superposition principle of traveling waves. One is that the sound pressure at a point is the sum of the pressures of all the traveling waves that could possibly exist there, and it was applied to decompose the incoming and outgoing waves of all modes at each interface. The other is that each outgoing wave pressure of each mode at each interface equals the sum of the products of the incoming wave pressures of all modes at all interfaces and the corresponding transmission factors. It was applied to determine, without using an echoic termination, each characteristic transmission coefficient by which each incoming wave contributes to each outgoing wave pressure. The transmission coefficients of the attenuator were obtained for the fundamental and the first-order modes. Agreement of those by experiment and numerical results is substantially good.

2:45

4pEA8. Applications of sonic soot cleaning techniques. Jing Tian (Inst. of Acoust., Academia Sinica, Beijing 100080, PROC)

Sonic soot cleaning is a method that is applied to clean the ash or fossil deposit on surfaces of heat exchangers in boilers. The technique includes a set of hardware and software for the designing, production, installation, debugging, and operation of sound wave generators and the controller. Although it has been applied in boilers in power plants, petrochemical works, and general industries worldwide, still most of the correlated basic problems, such as the relationships between sound field and the mass delivery, the heat conduction as well as the combustion process in boilers, have not been well solved. It is almost a purely empirical method that its application is far in the lead of fundamental research. Now in China more than ten institutions and companies are engaged in the research, development, and marketing of sonic soot cleaning techniques. These techniques cover sound generators ranging a frequency domain from infrasound up to ultrasound, under the protection of more than 20 patents. In this paper, several applications are analyzed as examples to demonstrate the effectiveness. Some basic phenomena and unsolved problems are revealed. The course of sonic soot cleaning development is discussed. [Project supported by the Natural Science Fund of China.]

THURSDAY AFTERNOON, 25 JUNE 1998 GRAND CRESCENT (W), 1:00 TO 3:00 P.M.

Session 4pMU

Musical Acoustics: Timbre of Musical Sound II

James H. Irwin, Chair
Electrical and Computer Engineering Technology, ECET, Bradley University, Peoria, Illinois 61625

Contributed Papers

1:00

4pMU1. Validation of a multidimensional distance model for perceptual dissimilarities among musical timbres. Nicolas Misdariis, Bennett Smith, Daniel Presssnitzer, Patrick Susini (Inst. de Recherche et de Coordination Acoustique/Musique (IRCAM), 1 place Igor Stravinsky, F-75004 Paris, France), and Stephen McAdams (IRCAM, F-75004 Paris, France and (CNRS), Univ. René Descartes, EPHE, F-75006 Paris, France)

Several studies dealing with the perception of musical timbre have found significant correlations between acoustical parameters of sounds and their subjective dimensions. Using the conclusions of some of these studies, a calculation method of the perceptual distance between two sounds has been developed. Initially, four parameters are considered: spectral centroid, irregularity of the spectral envelope, attack time, and degree of variation of the spectral envelope over time. For each of these, a transformation factor between the physical axis and the corresponding subjective dimension is obtained by linear regression. After a normalization of the data, the four coefficients then found are those of a linear combination that gives the final distance values. Since this model is based on numerical results derived from experiments that mostly used synthesized sounds, the application to a database of recorded musical instrument sounds needs a strong validation procedure. This procedure involves the adjustment of the coefficients of the first four parameters as well as the eventual introduction of new ones to attain a perceptually relevant distance between two musical sounds. The progress of this research and the results of the database search engine built on this similarity model will be presented and discussed.

1:15

4pMU2. Timbre effects caused by drumstick tip shapes/sizes. James H. Irwin, Jr. (Elec. and Comput. Eng. and Technol., Bradley Univ., Peoria, IL 61625, jhirwin@bradley.edu)

The relationship between the shape/size of a drumstick tip and the resulting initial spectrum produced by hitting a drumhead was investigated. Contrary to tensioned plate theory, and popular opinion, the amplitudes of the various induced partials were not significantly different when the shape/size of the tip was changed dramatically. The work was extended by using various diameter steel balls to strike the drumhead and it was found that the initial spectra did change for extreme differences in ball diameter. Further investigation showed two, or possibly three, timbre groupings. Present results indicate that small diameter balls (less than 15 mm diameter) induce significant energy above the first few partials of the drumhead, medium diameter balls (15 to 25 mm diameter) induce most of their energy in the first few partials, and perhaps large diameter balls (greater than 25 mm diameter) induce little energy above the lowest partial. These effects are linked to the membrane and plate characteristics of the drumhead. Since most drumstick tips have diameters in the medium range little distinction occurs among their spectra. In all cases the spectra were normalized for energy input, and the drum tension and position of impact were fixed.

1:30

4pMU3. On the intentional use of instrument characteristics in contemporary classical compositions. Alexandra Hettergott (Inst. of Musicology, Univ. of Vienna, Spitalgasse 2, 9. Hof, A-1090 Vienna, Austria)

Compositional intention is likewise subjected to the (mechanical) instruments' sound characteristics and technical possibilities, for instance, the possibility to produce impulsive or stationary sounds, to form (pedaled) chord clusters, or to microtonally retune pitch during performance. These differing techniques may additionally allow several (intended) psychoacoustical effects, as a changing fluctuation (beatings, roughness) or pitch strength, which also have a strong impact on timbre: so in the latter case the pitches' varying distinctiveness may be combined with a ''metallic,'' ''plastic,'' or ''wooden'' sound coloring. Consequently, in composing the decision for strings, either *struck* (piano, also prepared), *bowed* (violin, microtonally tuned), or *plucked* (harpsichord), for *flue*, *reed* (organ), or *free-reed* (mouth organ) pipes, etc., is also subjected to (*a priori*) intentional reflections aimed on the very (new) sound experience imagined. By means of music and score examples [W. Rihm, J. Cage, G. Scelsi, G. Ligeti, O. Messiaen, T. Hosokawa] as well as spectra and spectrograms the paper sheds some light on the composers' sounding and purposeful use of the mechanical instruments' possibilities in sound production in order to get (close to) the corresponding aesthetics of expression intended.

1:45

4pMU4. Musical instrument perception in cochlear implant listeners. John J. Galvin III and Fan-Gang Zeng (House Ear Inst., 2100 W. 3rd St., 5th Fl., Los Angeles, CA 90057-1922)

To examine the degree to which the cochlear implant preserves acoustic cues for timbre perception, eight cochlear implant and three normal-hearing listeners participated in a closed-set musical instrument identification experiment. Eleven nonpercussive, single-note instrument samples including woodwinds, brass, and strings were selected from the MUMS Classical Sounds CDs and formed a six-token and a nine-token set condition. The stimuli were normalized to have the same duration and RMS amplitude, and edited to include either the natural attack or the steady portion only. Subjects were trained with trial-by-trial feedback until asymptotic performance was achieved on one pitch (a4) and were tested without feedback on another pitch (c4). In the six-instrument task, both normal and implant listeners scored an averaged 70%–80% correct identification performance for both the natural attack and steady-state conditions. In the nine-instrument task, the implant listeners scored 20%–30% lower than the normal control. These results indicate that in a relatively simple six-instrument task, the cochlear implant can provide sufficient cues to support normal-level timbre perception, whereas the distorted spectral transformation via the cochlear implant fails to support a normal-level timbre perception, as the task complexity is increased to require more spectral resolution in the nine-instrument task.

2:00

4pMU5. Signal processing techniques to analyze the effects of guitar geometry on musical timbre. Dominic DiDomenico, Kevin Morris, and Suzanne Keilson (Loyola College, Dept. of Elec. Eng. and Eng. Sci., 4501 N. Charles St., Baltimore, MD 21210, keilson@loyola.edu)

Musical acousticians seek to answer many questions. What makes a particular instrument sound like it does? How do listeners perceive musical sounds? As an undergraduate design project, three guitars of varying shapes (''box,'' ''triangle,'' ''octagon'') were constructed and then compared to the traditional design. Materials were not varied, only the geometry of the sound boxes. The overall volume of each sound box was approximately equal. Digitized samples of the plucked strings of the constructed and traditional guitar were obtained. Conventional Fourier signal analysis of the notes from the three constructed guitars showed similarities to those of the traditional guitar. Other signal processing methods, such as short-time Fourier transform spectrograms, appropriate for the transient signal of the plucked string, were also investigated. Although the signals were broadly similar, analysis revealed differences, which might be attributed to the different geometries. A survey was conducted to assess listener's preferences for the various instruments. The survey was conducted from taped samples of the plucked strings to correlate the importance of psychoacoustics and the physical signal in assessing instrument quality. Listener preference rated the octagon-shaped guitar as highly as the traditional one. [Work supported by the Loyola College Department of Electrical Engineering and Engineering Science, and the Loyola College Hauber Fellowship Fund.]

2:15

4pMU6. The subjective evaluation of resonating sound-art samples and its relation to psychoacoustic measures. Densil A. Cabrera (Dept. of Architectural and Design Sci., Wilkinson Bldg., G04, Univ. of Sydney, NSW 2006, Australia, densil@arch.usyd.edu.au)

A genre of sound-based art is identified, wherein the artist simply records the sound of a continuously resonating acoustic system, with a minimum of intervention in the process. The sound of such works is often close to steady state, and tends to evolve slowly. The sounds include nearly pure tones, tone combinations (sometimes with prominent beats), and sustained noiselike sounds. Fluctuations due to variations in the energy source (often the wind) also occur. It was thought that the abstract nature and simplicity of these sounds would lead to subjective evaluations based largely on fundamental psychoacoustic criteria. Fifty-three samples, of 20 s duration, were arbitrarily selected from the genre, including works by Alvin Lucier, Gordon Monahan, and Roger Winfield. Subjects rated these stimuli on five continuous bipolar scales: (dis)like, (un)interesting, (un)pleasant, emotional arousal, and emotional valence. The stimuli were measured for loudness, pitch characteristics, roughness, fluctuation, and lateralization. Analysis showed emotional arousal to be the strongest response, with significant correlations to loudness, roughness, and pitch. The results point towards the possibility of constructing a perceptual model of this type of minimalist art form. [Work supported by an Australian Postgraduate Award.]

2:30

4pMU7. The dependence of timbre perception on the acoustics of the listening environment. Dae-Up Jeong (Dept. of Architectural and Design Sci., Wilkinson Bldg., G04, Univ. of Sydney, NSW 2006, Australia, jeong_d@arch.su.edu.au) and Fergus R. Fricke (Univ. of Sydney, NSW 2006, Australia)

The acoustic quality of a room for music cannot easily be determined. As timbre is important in music it is possible that the perceived change in timbral quality of a sound in two spaces may give the basis for a measure of acoustic quality. The smallest perceivable change in the timbre of steady-state stimuli was measured in different listening environments through two experiments featuring a spectral centroid and rise time, respectively. Five component harmonic stimuli at 440 Hz were employed. The centroid of the standard stimuli was fixed at 2.95, in which the median amplitude of each component of the standard was set at 0.5, while those of comparison stimuli were varied from 2.93 to 2.41 in steps of 0.02 with a negative spectral tilt. The smallest perceivable change in the rise time was also measured using complex tones containing ten harmonics of 200 Hz (F_0) with linear spectral slopes of -6 dB/oct. Rise times of comparison stimuli were varied from 5 to 25 ms in 2-ms steps, while the fall time was fixed at 5 ms across the stimuli. Results are presented and comments are made on how the acoustics of the listening environment influences the perception of timbre.

2:45

4pMU8. Perceptual analysis of synthesized struck bars. Vincent Roussarie (Inst. de Recherche et de Coordination Acoustique/Musique (IRCAM), 1 place Igor Stravinsky, F-75004 Paris, France), Stephen McAdams (IRCAM, F-75004 Paris, France), and Antoine Chaigne (Ecole Nationale Supérieure des Télécommunications (ENST), F-75013 Paris, France)

This study is part of a project seeking to understand musical timbre and perceptual phenomena of the sound environment. It concerns the sensitivity of listeners to sounds emitted by simple vibrating structures, such as xylophone bars. Previous studies of timbre have characterized this attribute in terms of a small number of perceptual dimensions of an analytic nature (attack time, spectral centroid, spectral flux). Other studies have suggested a return to the physical characteristics of the sound sources themselves in order to understand their influence on timbre perception. Sounds were synthesized with a computer model of the vibrating bar developed at ENST that allows precise control of physical parameters. Listeners made dissimilarity judgments on all pairs of different sounds. Multidimensional scaling analyses of these data yield a perceptual space with dimensions that correspond strongly to the varied physical parameters (material density and the first term of internal damping) as well as to salient auditory dimensions (pitch for the former and a combination of spectral centroid and decay rate for the latter). These results argue in favor of a privileged processing of source characteristics by the auditory system.

THURSDAY AFTERNOON, 25 JUNE 1998

CASCADE BALLROOM I, SECTION A (W), 1:00 TO 5:00 P.M.

Session 4pNSa

Noise and Architectural Acoustics: Surface Transportation Noise: Trains and Automobiles

Carl E. Hanson, Chair

Harris Miller Miller and Hanson, 15 New England Executive Park, Burlington, Massachusetts 01803-5221

Chair's Introduction—1:00

Invited Papers

1:05

4pNSa1. Diagnosis of noise sources from high-speed trains using the microphone-array technique. B. Barsikow (akustik-data Eng. Office, Kirchblick 9, D-14129 Berlin, Germany)

To reduce noise radiated by high-speed trains, abatement at the source is becoming increasingly important. Techniques for accomplishing this task require an exact knowledge of the individual sources involved and their reduction potential. During the past several years, a number of sound-generating components on ICE trains operated by the DB AG (German Railway) have been investigated by the akustikk-data Engineering Office. Microphone arrays provide a useful technology for investigating noise radiated by individual sources. Among the arrays used for recent measurements are the horizontal and vertical interspersed line arrays as well as X-shaped arrays. Examples measured by each of these instruments are shown. Results of these measurements with arrays yield the strength of each source and their speed exponents. In addition, narrow-band analysis of the sound is possible because the effects of the Doppler shift are not present. With this information, the importance of each source to the total radiated noise can be estimated. Examples of such results are given for rolling noise as well as for components emitting aerodynamic noise. For both types of sources, the effectiveness of various abatement measures is discussed.

1:25

4pNSa2. Noise prediction and control for new Norwegian high-speed trains. Matias Ringheim (Kilde Akustikkas, P.O. Box 229, N-5701, Voss, Norway)

The prediction of future noise levels is an important part of the planning process for high-speed trains and tracks. The first Norwegian high-speed tracks will be opened in 1998, using new 200-km/h electric trains based on the Swedish X2000. Strict noise criteria have been defined and extensive noise control measures are necessary both on the train itself and along the track. The noise prediction was performed several years ago, with the early and fairly simple Nordic prediction method. In the final planning stages some of the more critical results were checked against the revised Nordic method. The method enables the calculation of outdoor energy equivalent levels and maximum levels in octave bands, in simple and complex terrain. The first version of a computer implementation is available. The outdoor results have a form which is suitable for the further calculation of indoor levels, when the building properties are known. The basic aspects of the method, including a summary of available high-speed data, will be presented. Problems related to the prediction and reduction of future noise from new high-speed trains will be summarized.

Contributed Papers

1:45

4pNSa3. Ground-borne vibration measurements of high-speed trains. David A. Towers (Harris Miller Miller & Hanson, Inc., 15 New England Executive Park, Burlington, MA 01803, dtowers@hmmh.com) and Hugh J. Saurenman (Harris Miller Miller & Hanson, Inc., Burlington, MA 01803)

Ground-borne vibration measurements of high-speed train operations were carried out in France for the TGV and Eurostar trains, in Italy for the Pendolino trains, and in Sweden for the X2000 trains. The measurements were made to develop vibration-prediction models for a new guidance manual published by the U.S. Federal Railroad Administration on ''High Speed Ground Transportation Noise and Vibration Impact Assessment.'' The tests included measurements of ground-borne vibration at various distances from the track as well as a procedure to characterize the ground vibration propagation characteristics at each measurement site. The results indicated a wide spread in the vibration data, partly due to differences in the equipment and track condition and partly due to differences in site geology. Further analysis suggested that much of the difference between the trainsets is due to variations in the geology rather than differences in suspension, axle load, or wheel conditions of the trainsets. Applying the characteristics of one type of trainset to different test sites resulted in widely varying ground vibration levels and propagation rates, whereas normalizing the ground vibration from the trainsets to one site substantially reduced the differences in overall vibration level.

2:00

4pNSa4. New noise impact criteria for high-speed ground transportation systems in the United States. Carl E. Hanson (Harris Miller Miller & Hanson, Inc., 15 New England Executive Park, Burlington, MA 01803, chanson@hmmh.com)

New noise impact criteria have been developed by the U. S. Department of Transportation for the assessment of environmental impacts of high-speed rail projects. The environmental impacts of new high-speed rail projects in the United States are the responsibility of the Federal Railroad Administration (FRA). Adoption of these criteria will assure a uniform treatment of noise impacts of new high-speed rail systems throughout the country. The noise impact levels were developed based on well-documented criteria, and on comprehensive social survey data concerning annoyance due to transportation noise. Impact levels are based on the Schultz response curve and include consideration of the combination of project and ambient noise levels. Correction terms are included in the case of high-speed rail (and maglev) to account for startle effects from fast rise times. This presentation discusses the basis for the curves used as thresholds of impact—where noise mitigation is to be considered—and severe impact—where noise mitigation must be implemented. These criteria are included in a new guidance manual published by the FRA on ''High Speed Ground Transportation Noise and Vibration Impact Assessment'' and will be used throughout the USA.

2:15

4pNSa5. Wayside noise measurements of high-speed trains. David A. Towers (Harris Miller Miller & Hanson Inc., 15 New England Executive Park, Burlington, MA 01803, dtowers@hmmh.com)

Wayside noise measurements of high-speed train operations were carried out in the USA for the German ICE, Swedish X2000, and U.S./French RTL-2 Turboliner trains, in France for the TGV and Eurostar trains, in Italy for the Pendolino trains, and in Sweden for the X2000 trains. The measurements in the USA were made as part of the Northeast Corridor Project and the measurements in Europe were made to develop noise-prediction models for a new guidance manual published by the U.S. Federal Railroad Administration on ''High Speed Ground Transportation Noise and Vibration Impact Assessment.'' The results indicate that the TGV, ICE, and RTL-2 trains had similar noise emissions and were typically the quietest of the trains tested. A-weighted noise levels for the X2000 and Pendolino trains averaged about 5 dB higher, and data for the Eurostar trains were scattered over the range for the other trains. Although the higher-speed French trains generated some additional low-frequency aerodynamic noise, this noise did not appear to significantly affect the A-weighted noise levels for train speeds up to nearly 300 kph (186 mph). The results also suggest that at the speeds tested, the observed trains were generally in compliance with the U.S. Federal Railroad Noise Emission Standard.

2:30

4pNSa6. Identification of moving acoustic sources from the pass-by noise. Byoung-Duk Lim and Deok-Ki Kim (Dept. of Mech. Eng., Yeungnam Univ., Gyongsan, Kyungbuk, 712-749, Korea)

The identification of a moving noise source such as a transportation system is usually difficult at high speed since microphones attached to the moving noise source may induce a high level of aerodynamic noise, while the measurement using a fixed microphone suffers from the distortion of the source characteristics due to its motion. In this study a time domain technique based on the finite difference method and regularization is developed for the recovery of the stationary source characteristics from the distorted signal measured at a fixed point. The sources are assumed to be point sources with stationary frequency characteristics, and move along a line at a constant subsonic speed. In the previous work of the present authors recovery of the stationary source signal from a point source requires the information about time origin. For multiple sources it is impossible to set a single time origin that a source signal recovery method using multiple time origins are developed. In the numerical simulations the effects of estimation errors in the parameters, such as the speed of source, time origin, and background noise level, are investigated and the results show the usefulness of this method.

2:45–3:00 Break

3:00

4pNSa7. The acoustical impact of local railway lines. Cristina Pronello (Ing. Cristina Pronello C.so Duca degli Abruzzi n. 24-10129 Torino, Italy, pronello@polito.it)

Local railway lines serve a regional traffic. They have few users since the service offered presents insufficient quality standards. These lines could have more users and they could become metropolitan connections by adopting new vehicles which are faster and more modern. This study has dealt with the current acoustical impact that they produce. Therefore, a number of acoustical measurements have been made along a local railway line. They have been carried out by varying the speed of the trains at a fixed measuring distance of 7.5 m and by varying the measuring distance with a single train speed. The obtained results have shown that current trains produce in the average a greater acoustical impact when compared to modern trains. This has been observed even at slow speeds and with reduced train lengths. Measurements made on a new train, that should soon start its service on the line, have shown that there would be no increase of the produced acoustical level, even with increased speeds. As a consequence, it may be concluded that the current service on the analyzed line could be enhanced by increasing the train speed and the number of runs, maintaining an equivalent daily analogous to the one currently guaranteed.

3:15

4pNSa8. Limits to limits? Tor Kihlman and Wolfgang Kropp (Dept. of Appl. Acoust., Chalmers Univ. of Technol., S-41296 Gothenburg, Sweden)

Today, traffic noise due to cars, trains, and airplanes is the main noise source in urban areas. In accordance with the European Commission's Green paper on Future Noise Policy, 22% of the European population are exposed to outdoor A-weighted noise levels higher than 65 dB and more than 45% are exposed to levels between 55 and 65 dB. This corresponds in total to about 250 million people. To improve the situation is a tremendous challenge and the question arises as to whether a good environment be provided (i.e., levels below 55 dB) for all people in today's cities. Can the demands of 55 dBs only be fulfilled by structuring traffic in a smart way and using noise barriers? Or is a clear change in traffic policy needed from individual traffic to public transport? Different cities with different structures and automobile dependencies are compared in a general study based on statistical data. Detailed information about traffic flow, population structure, and land use is analyzed to verify these results.

3:30

4pNSa9. Effects of traffic noise within the Madrid region. Manuel Recuero, Constantino Gil, and Jorge Grundman (Inst. de Investigacion del Automovil (INSIA), Univ. Politecnica de Madrid, Ctra. Valencia km. 7, 28031 Madrid, Spain)

In this work the results of a 20-question survey about the acoustic environment made on the population of 17 towns are presented. Up to 7141 questionnaires were distributed, where 3272 were conducted in population centers with more than 100 000 inhabitants, 2695 in towns below 100 000 and above 50 000 inhabitants, and 1174 in towns with less than 50 000 inhabitants. The aim of this statistical work is to estimate the citizens' opinions about noise sources in their municipalities, the annoyance that these sources produce, where and when the effect seems stronger, and how the noise affects residential areas. There were also questions concerning the opinions about protection against noise in dwellings and the presumed effects of environmental noise. Finally, it was intended to know how important the residents think the environmental noise problem is, how deep their knowledge is about their rights and the law, and what do they think about measures to improve the situation.

3:45

4pNSa10. Measuring the traffic noise in Valencia. E. Gaja, J. L. Manglano, A. Reig, and S. Sancho (Univ. Politécnica de Valencia, Camino de Vera S/N, 46071 Valencia, Spain)

The City of Valencia has approximately 700 000 inhabitants. Due to this large population, the city is subjected to heavy traffic, and, as a consequence, high acoustic pollution. The Townhall of Valencia is very worried about this acoustic pollution and has commissioned the Laboratory of Acoustics in the Universidad Politécnica de Valencia to work on various projects within the city. Due to the work the department has carried out on behalf of the Townhall of Valencia, it has gained experience measuring with the sonometer. An efficient way of measuring noise within the city that avoids wasting long periods of time has been developed. Continuous 24-hour measurements have been taken by sonometers located in different stations spread out around the city and controlled by the Townhall. With these measurements a method that indicates when to measure, in the morning, afternoon, and at night, as well as the exposure time, has been designed. Other parameters, such as characteristics of the traffic, types of streets, etc., have been considered in order to obtain the time and the exposure time of measuring that represents the level of the noise.

4:00

4pNSa11. Characterization of vehicle noise in Hong Kong. W. T. Ng, M. M. F. Yuen, and W. M. To (Dept. of Mech. Eng., Hong Kong Univ. Sci. Technol., Clear Water Bay, Kowloon, Hong Kong, mewmto@usthk.ust.hk)

There is a common perception among residents of high-rise buildings facing expressways that noise emitted from moving vehicles would be dominated by tire/road interaction and at high-frequency regions. Local government officials believe that the impact on residents of low-frequency noise radiated from car engines would be insignificant, especially after the corrections for the *A*-weighting scale are applied. This paper reports on an extensive survey and some *in situ* sound measurements. It was found that road noise in Hong Kong is dominated by low-frequency noise emitted from heavy vehicles in expressways and by low-frequency noise emitted from heavy and light vehicles driven at a speed below 50 km/h in streets. The measured data were characterized by using time-averaged 1/1 octave band analysis, time-averaged 1/3 octave band analysis, and time-frequency analysis. The Doppler effect was clearly observed at low-frequency regions in joint time-frequency distributions. It is suggested that the *A*-weighting scale would not reflect the true annoyance level of traffic noise. Noisiness should be used to quantify the annoyance caused by moving vehicles. A new noise model is proposed to give a realistic description of noise radiated from a moving vehicle.

4:15

4pNSa12. Multivariate analysis of road traffic noise in Gandia (Spain) during 24 hours and its evolution in the last decades. Jose Romero (Lab. Acoust., Dept. Fisica Aplicada U.V., Facultad de Ciencia Fisicas, Valencia, Spain, Romerof@arrakis.es), Alicia Jiminez, Antonio Sanchis, Albert Marin (Polytecnic Univ. of Valencia, Spain), Amando Garcia (Lab. Acoust., Valencia, Spain), and Grover Zurita (Lulea Univ. of Technol., Sweden)

Road traffic noise is one of the most widespread and growing problems in urban areas. While it has long been known that hearing can be damaged by exposure to noise, it is also believed that continual noise, even at low or moderate intensity, can cause psychological discomfort and sleep disorders. Traffic noise levels in urban areas are increasing, and the areas affected by noise are spreading. As a result of these problems, legislation has been introduced by the city council to control the noise produced by individual vehicles with the aim of eventually producing traffic noise levels which are acceptable to the public. In this paper, therefore, aspects which connect the evolution of the road traffic noise in the last decades and the development of the infrastructure of the city are discussed. The study was carried out in Gandia (Spain), which is situated on the East Coast of the country, 65 kms from Valencia and 100 kms from Benidorm. The population has increased from 50 000 to 60 000 in the last 15 years, and the traffic distribution of the city has changed during this time, with a new bridge and some highways around the urban area. The main objective of this paper is to characterize the effects of the road traffic noise during 24 h (between 1983 and 1997), and including such aspects as traffic density and psychological aspects. The analysis part was performed by using multivariate analysis methods (MVA). Multivariate analysis methods can be used to investigate relationships between all the variables by extracting information from data with many variables and treating them simultaneously.

4:30

4pNSa13. Criteria and control for environmental noise emitted from motor vehicles. Ren Wentang (Noise Vib. Control Key Lab., Beijing Municipal Inst. Labour Protection, No. 55 Tao Ran Ting Rd., Beijing, 100054, China)

The present work investigated the environmental noise in a city from energy distribution and annoyance, and showed that the noise from different kinds of in-use vehicles is the main cause of noise pollution, especially in a city in a developing country. The technology and measures of noise control for in-use vehicles are urgently needed. According to measurement results of noise emission from 1000 vehicles based on ISO 5130, the statistical distribution parameters of stationary noise emission for six types of new vehicles and in-use vehicles are obtained. The statistical parameters show that the technology and measures of noise control for exhaust systems of in-use vehicles are key points. Some technological advances and noise-control measures have been proposed and practiced in China. A measurement procedure and system for vehicles in stationary operation is developed to distinguish some noisy vehicles with lost efficacy of a muffler. Regulations concerning stationary noise limits have been published and followed in China subsequent to the investigation. Reasonable design parameters of exhaust systems for different kinds of vehicles are proposed.

4:45

4pNSa14. A mathematical model for the evaluation and prediction of the mean energy level of traffic noise in Caracas. Nila Montbrun, Victor Rastelli, Alexis Bouza (Dept. de Mecánica, Univ. Simón Bolivar, A.P. 89000 Caracas 1080-A, Venezuela), Jenny Montbrun Di Filippo, and Yamilet Sánchez (Univ. Simón Bolivar, A.P. 89000, Caracas 1080-A, Venezuela)

The increase of the population in capital cities has provoked a serious increase in the acoustical pollution, where the real cause is the traffic noise. Over ten years, sound levels have been measured in several points in the city of Caracas. The main objective was to establish a mathematical model for the mean energy level of traffic noise. This model is a modification and improvement of the model that was used in Germany and Italy. Parameters of the fundamental equations were obtained by the use of large experimental data. In the particular case of Venezuela, an innovation was introduced by the consideration of the motorcycle noise. The obtained model can be used to predict noise pollution in cities and contour maps can be made as well.

THURSDAY AFTERNOON, 25 JUNE 1998

CASCADE BALLROOM I, SECTION C (W), 1:10 TO 3:45 P.M.

Session 4pNSb

Noise and Signal Processing in Acoustics: Active Noise and Vibration Control

Clemans A. Powell, Chair
NASA Langley Research Center, MS 462, Hampton, Virginia 23681-0001

Chair's Introduction—1:10

Contributed Papers

1:15

4pNSb1. Active local noise control in open space. Jingnan Guo and Jie Pan (Dept. of Mech. and Mater. Eng., Univ. of Western Australia, Nedlands, WA 6907, Australia)

Local noise control, which creates quiet zones in desired areas, is a practical option of active noise control in open space when the control sources and the primary sources are not able to be very close. In most cases, the local control system attenuates the sound pressure in some areas at the cost of total sound power output increase. As a result, while the control system creates a quiet zone in some areas, it causes the sound pressure to increase in others. Two indicators of control efficiency, quiet zone size and total sound power output, are dependent upon the configuration of the control system. It has been found that for the multichannel control system, there exists a range of optimal configuration of control system, in which the control system can create the largest quiet zone with the lowest increase of total sound power output. Outside this range, the control system either becomes useless in creating large quiet zones, or causes great increase of sound pressure in most areas.

1:30

4pNSb2. Adaptive control of sound transmission with neural network algorithms. Jing Tian, Hai Lin, and Mingkun Cheng (Inst. of Acoust., Academia Sinica, Beijing 100080, PROC)

The Artificial Neural Network is widely used for its self-learning, self-organizing adaptability as well as nonlinearity in nature. Active control of structural sound radiation and sound transmission through panels is widely researched recently for its effectiveness and efficiency in the low-frequency range. In this paper, a neural network-based feedforward adaptive controller using the modified Error Back Propagation Learning Algorithm is presented. The controller is realized with a TMS320C25 DSP Board monitored by an IBM PC compatible, and applied to control the sound transmission through a thin panel between two rooms. Experiments showed that the algorithm was superior in robustness and broadband performance. In the adaptive controller, the structure of a traditional multilayered feedforward neural network (MFNN) is modified. Both filtered-X and filtered-U adaptive filters are discussed. Two auxiliary filters are designed to compensate the secondary signal feedback and the error delay. Random gradient estimation method is used to update the weights of MFNN. Several methods to speed the learning rate are also introduced. The causality of the feedforward controller is analyzed, by which the system behavior is greatly improved. Further strategy to improve the controller is also proposed. [Project supported by the Natural Science Fund of China.]

1:45

4pNSb3. Real-time wave separation in a cylindrical pipe with applications to reflectometry and echo-cancellation. Jean Guerard and Xavier Boutillon (Lab. d'Acoustique Musicale, CNRS-Univ. Paris 6, Case 161, 4 place Jussieu, 75252 Paris Cedex 05, France)

Precise control of the acoustical pressure all along a pipe requires that backward and forward propagating waves are separated in real time. This has been achieved by means of an array of up to five microphones, a high-precision analog calculation, and a real-time numerical post-treatment. The separation factor exceeds 40 dB over several kHz, when special attention to intercalibration issues and to the realization of the analog part is given. The theoretical limit at high frequencies has been reached experimentally. The advantage of a partial analog treatment over a fully numerical one will be discussed. One application consists in reflectometry (and consequently, impedancemetry) without use of the long pipe normally required. Associated with a real-time DSP processor (TMS 320C30), wave separation also permits one to efficiently control the wave reflected by one end. This is done with only one loudspeaker, whose transfer function has been inverted and implemented in the DSP as an IIR filter. This approach of active-control differs from adaptative procedures. Finally, a musical application of the system to a hybrid wind instrument will be presented. [Work supported in part by the French Ministry of Culture.]

2:00

4pNSb4. Active noise control of a single-engine light aircraft cabin. Colin D. Kestell and Colin H. Hansen (Dept. of Mech. Eng., Univ. of Adelaide, Adelaide, SA 5005, Australia, cdkestel@watt.mecheng.adelaide.edu.au)

Active noise control (ANC) shows success and potential in a growing number of commercial applications, one of which is aircraft cabin noise reduction. With the exception of utilizing ANC headsets, light aircraft, which to date offer a high noise environment, have been somewhat overlooked. The importance of weight minimization also prevents installing copious quantities of dampening and insulation materials as a passive noise control measure. While headsets are a pilot's necessity and an obvious target for ''localized'' noise reduction, they are not conducive to either operator or passenger comfort. High noise levels not only render communication difficult but also contribute toward stress and fatigue. A more globalized region of reduced noise will be less restrictive and no doubt provide the occupants with more freedom of movement and overall comfort. Light aircraft operators boasting quieter cabins with a focus on customer comfort will no doubt have a distinct commercial advantage. Using flight trials and laboratory experiments as a basis, this paper will discuss the introduction of ANC into a four-seater Piper Archer. Existing noise levels, objectives, equipment used, methods of approach, and results to date shall be reviewed, as well as the remaining work required to achieve the final goal.

2:15

4pNSb5. Control of exhaust noise from diesel electric locomotives using a hybrid active/passive system. Paul Remington (BBN Technologies, 10 Moulton St., Cambridge, MA 02138), Douglas Hanna, and Scott Knight (BBN Technologies, New London, CT 06320-6147)

Diesel electric locomotives in the US represent a significant source of environmental noise under a variety of operating conditions. There are two primary sources of noise on these locomotives that will have to be reduced before significant reductions can be achieved: exhaust noise and cooling fan noise. Here a system is proposed that uses both active and passive components to control broadband exhaust noise below 3 kHz. The active system utilizes eight actuators and eight residual microphones in a feedforward configuration to control tonal noise below 250 Hz. The passive system is a compact exhaust silencer designed to control exhaust noise above that frequency while still fitting within the limited space available beneath the locomotive hood. Performance estimates for both the active and passive system will be presented along with the results of field tests of a breadboard active system on an F59PHI locomotive. [Work was supported under contract to the Federal Railroad Administration.]

2:30–2:45 Break

2:45

4pNSb6. Anticipated effectiveness of active noise control in propeller aircraft interiors as determined by sound quality tests. Clemans A. Powell and Brenda M. Sullivan (NASA Langley Res. Ctr., M.S. 462, Hampton, VA 23681-0001, c.a.powell@larc.nasa.gov)

Active noise control technology offers the potential for weight-efficient aircraft interior noise reduction, particularly for propeller airplanes. However, there is little information on how passengers respond to this type of interior noise treatment. This paper presents results of two experiments which use sound quality engineering practices to determine the subjective effectiveness of hypothetical active noise control systems in a range of propeller airplanes. Binaural recordings were made in five different propeller airplanes and a commercial jet using an acoustic mannequin and a digital recording system. The noise stimuli for the tests were prepared by digitally modifying the propeller tones in recordings of each airplane type to simulate active noise control systems with a range of effectiveness and complexity. The noise stimuli were presented to the test subjects through electrostatic headphones to preserve the realism, spatiality, and directionality provided by the binaural system. The two tests differed by the type of judgments made by the subjects: pair comparisons in the first test and numerical category scaling in the second. Changes in subjective response relative to changes in measured sound characteristics afforded by the hypothetical active noise control schemes are presented and are contrasted between the two test methods.

3:00

4pNSb7. Robust feedback active noise control algorithm for impulsive additive noise. Sang-Wook Lee and Koeng-Mo Sung (School of Elec. Eng., Seoul Natl. Univ., Seoul 151-742, Korea, lsw@acoustics.snu.ac.kr)

Active noise control (ANC) is a method to reduce the unwanted noise level by introducing secondary noise which has the same amplitude and antiphase with the primary noise. The feedback ANC algorithm is a kind of linear predictor and it uses only one sensor to get the information about the error signal and reference signal. The filtered-x LMS (least mean square) algorithm is widely used for implementing the ANC system because of its simplicity and good performance. But when the additive noise is impulsive, the performance of LMS-type algorithms are known to be lowered. In this paper, the robust feedback ANC algorithm is proposed when impulsive additive noise is present. Instead of the LMS algorithm, the proportion-sign algorithm (PSA) which is a mixture of LMS and dual sign algorithm (DSA) is used to estimate the primary noise field robustly when impulsive additive noise is present. Computer simulations were performed under impulsive additive noise circumstances and showed better performance with the proposed algorithm than that with the conventional LMS algorithm.

3:15

4pNSb8. Active control of structural sound using an active constrained layer damping treatment. Jun Yang, Jing Tian, and Xiaodong Li (Inst. of Acoust., Academia Sinica, Beijing, 100080, PROC)

In this paper active constrained layer damping (ACLD) treatments are applied to control the noise transmitted into a cavity through a panel. Five of the enclosure walls are rigid and the other wall, which is flexible, is a simply supported panel. The panel is treated with ACLD, which consists of a viscoelastic shear layer sandwiched between a piezoelectric constraining cover sheet and the panel. The piezoelectric layer is utilized to control the panel vibration and the shear deformation of the viscoelastic layer to enhance its energy dissipation characteristics. The coupled structural sound equation of the composite panel and the cavity is derived and analyzed by the Galerkin approximation. The results indicate that combining active and passive controls, as in the case of ACLD treatment, is more effective in controlling the noise transmission. Moreover, such a control strategy is found to require lower control inputs than when the active and passive control actions are used separately.

3:30

4pNSb9. Active noise barrier efficiency improvement using multirate signal processing. Jelena Certic, Slobodan Kovacevic, and Petar Pravica (Univ. of Belgrade, Dept. of Elec. Eng., Bulevar Revolucije 73, Belgrade, Yugoslavia, cetic02996P@buef31.etf.bg.ac.yu)

In this paper, implementation of multirate signal processing techniques in an active noise barrier system was considered. Such a modification of a conventional multiple-error filtered-x LMS algorithm is supposed to give improved overall system performance, by separately controlling two different frequency ranges. The control system consists of one detection microphone, three secondary sources, and three error sensors. The primary signal was a random noise signal with a spectrum that coincides a typical road traffic noise spectrum. Locations of components were the ones found to be optimal in our previous work. Computer simulation was developed to compare the attenuation achieved with a control system that uses a multiple-error filtered-x LMS algorithm and the one involving multirate signal processing. Results of the computer simulation confirmed that attenuation is increased using multirate signal processing.

THURSDAY AFTERNOON, 25 JUNE 1998 ASPEN ROOM (S), 1:00 TO 3:45 P.M.

Session 4pPAa

Physical Acoustics and Bioresponse to Vibration/Biomedical Ultrasound: Cavitation Dynamics: In Memoriam Hugh Flynn II

Charles C. Church, Cochair
Acusphere, Inc., 38 Sidney Street, Cambridge, Massachusetts 02139

Werner Lauterborn, Cochair
University of Goettingen, Drittes Physikalisches Institute, Buergerstr. 42-44, D-37073 Gottingen, Germany

Invited Papers

1:00

4pPAa1. Acoustically induced cavitation fusion. Lawrence A. Crum (Appl. Phys. Lab., Univ. of Washington, Seattle, WA 98105)

In 1982, Hugh Flynn was issued a patent (No. 4,333,796) for a Method of Generating Energy by Acoustically Induced Cavitation Fusion and Reactor Therefor. Although it was largely ignored at the time, there have been several recent papers that treat the subject of acoustically induced fusion as quite plausible. Such prescience in the area of cavitation research was typical of Hugh's work. This patent shows such an enormous grasp of detail, and suggests Hugh must have given a great deal of thought and energy to its composition. The author will review some of the interesting aspects of this patent as well as describe some of the most recent activity on this topic. He will also reflect on his many rewarding and stimulating experiences with Hugh over the past 30 years.

1:20

4pPAa2. The search for cavitation *in vivo*. Edwin L. Carstensen, Sheryl Gracewski, and Diane Dalecki (Depts. of Elec. Eng. and Mech. Eng. and the Rochester Ctr. for Biomed. Ultrasound, Univ. of Rochester, Rochester, NY 14627)

Until the mid-1970s, it was generally assumed that with the short pulses of ultrasound used in medical diagnosis, there was little need for concern about the possibility of inertial cavitation *in vivo*. Conversations with Hugh Flynn at about that time made it clear that complacency in this respect was not warranted. Flynn carried out a comprehensive theoretical study of the response of microbubbles to diagnostically representative pulse exposures culminating in a publication, which is now the classic basis for an understanding of cavitation in diagnostic medicine. Robert Apfel independently published similar conclusions at about the same time. Flynn's contributions continued in a collaboration with Charles Church. Experimental evidence that cavitation *in vivo* might be relevant to the use of diagnostic ultrasound began with observations of the killing of fruit fly larvae with microsecond length pulses. Later, the mammalian lung was shown to be particularly vulnerable to these exposures. Noting that purely positive pulses were as damaging as purely negative pulses, Ding and Gracewski showed that inertial cavitation is greatly inhibited by bubble constraints that might occur in tissue. Most recently, hemolysis and hemorrhage associated with the use of contrast agents have provided nearly incontrovertible evidence of the occurrence of cavitation *in vivo*.

1:40

4pPAa3. Nonlinear dynamics of bubbles with surfactants. Robert Apfel, Xiaohui Chen, and Jeffrey Ketterling (Yale Univ., New Haven, CT 06520-8286)

Hugh Flynn pioneered many aspects of cavitational phenomena. He even got a patent on the idea that large bubble collapse could be used for inertially induced thermonuclear fusion. Some researchers currently working in the area of sonoluminescence are asking the same questions. One of the key requirements for confining the energy in the collapse of a bubble is maintaining spherical symmetry. Instabilities, such as Rayleigh–Taylor and Kelvin–Helmholtz, can compromise this requirement. Therefore it is useful to examine the role of surfactants in conferring stability on the collapse of bubbles. Codes that have recently permitted the description of superoscillations of liquid drops with surfactants in air [e.g., Apfel *et al.*, Phys. Rev. Lett. **78**, 1912–1915 (1997)] can be applied to the case of a bubble in a liquid. Of particular interest are insoluble surfactants, such as bovine serum albumin, which do not reduce the surface tension that much, but which greatly increase the surface damping. The role of damping in discouraging or delaying instability of collapsing bubbles in an acoustic field will be presented. [Work supported in part by NASA through Contract No. 958722 managed by the Jet Propulsion Laboratory.]

2:00

4pPAa4. Response curves of bubbles. Werner H. Lauterborn and Robert Mettin (Drittes Physikalisches Institut, Universität Göttingen, D-37073 Göttingen, Germany)

A spherical bubble in a liquid is a nonlinear oscillator that can be set into radial (and surface) oscillations by a sound field. The nonlinear response depends on a set of parameters, mainly the bubble radius, the driving amplitude, and the driving frequency. The response is involved. It consists of an intertwined set of nonlinear resonances, that besides turning over, comprises period doubling sequences to chaos when going up and down in bubble radius, driving frequency, or driving amplitude. A specific feature is the giant resonance appearing for small bubbles and being the host for single bubble sonoluminescence. A set of response curves will be given to acquaint the listener with some understanding of how bubbles behave in a sound field.

2:20–2:30 Break

Contributed Papers

2:30

4pPAa5. Bubble dynamics in non-Newtonian fluids. John Allen (Dept. of Mech. Eng., Univ. of Washington, Seattle, WA 98195) and Ronald A. Roy (Boston Univ., Boston, MA 02215)

Nonlinear oscillations of gas bubbles in viscoelastic non-Newtonian fluids remains a relatively unexplored area, but one of increasing importance with the growing use of high intensity ultrasound and contrast agents for imaging enhancement. Previous work by the authors focused on an analytical and numerical study of bubbles governed by linear viscoelastic constitutive equations. This work was constrained by the small deformation assumptions inherent in the equations. New numerical studies of objective viscoelastic constitutive equations reveal the limitations of the linear viscoelastic models and show many novel results. In particular, the risk of bioeffects from medical ultrasound and harmonic imaging applications are stressed. Questions about the trace of the stress tensor are also discussed. [Work supported by NIH and DARPA.]

2:45

4pPAa6. Giant resonance in the dynamics of small bubbles. Iskander Sh. Akhatov (Ufa (Bashkortostan) Branch of Russian Acad. of Sci., K. Marx St. 6, Ufa, 450000, Russia, iskander@ncan.ufanet.ru), Claus-Dieter Ohl, Robert Mettin, Ulrich Parlitz, and Werner Lauterborn (Universität Göttingen, Göttingen, Germany)

For bubble oscillations under medium and large pressure amplitudes a complicated scenario of bifurcations and coexisting (chaotic) attractors exists. However, for very small bubbles in a very strong sound field the dynamics becomes regular and a new type of strong resonance with a thresholdlike increase in oscillation amplitude occurs [E. A. Neppiras and B. E. Noltingk, Proc. Phys. Soc. London Ser. B **64**, 1032 (1951); H. G. Flynn, in *Physical Acoustics Vol. 1*, edited by W. P. Mason (1964), p. 57; W. Lauterborn, Acustica **20**, 105 (1968)]. This phenomenon has a strong influence on many properties of cavitation bubbles. The following aspects are considered: rectified diffusion, stability of the bubble under SBSL conditions, primary and secondary Bjerknes forces, and interpretation of bubble size measurements.

3:00

4pPAa7. Cavitation bubble dynamics induced by ultrasound waves. J-L. Laborde (EDF-DER ADEI, Rte. de Sens-Ecuelles, 77818 Moret-sur-Loing Cedex, France, jean-luc.laborde@edfgdf.fr), C. Bouyer (EDF-DER ADEI, 77818 Moret-sur-Loing Cedex, France), J-P. Caltagirone (ENSCPB-MASTER, 33402 Talence Cedex, France), and A. Gérard (LMP, Université de Boreaux I, 33405 Talence Cedex, France)

Propagation of power ultrasound (from 20 to 800 kHz) through a liquid initiates a not-so-well-known phenomenon called acoustic cavitation. Inceptions and germs grow into bubbles which collapse, possibly giving rise to extreme conditions of temperature and pressure (assessed to be up to 10 000 K and 500 atm). For instance, these conditions initiate and greatly enhance chemical reactions. A high-speed film shot at 500 fps clearly identifies stable and transient cavitation and shows bubble population phenomena. Clouds of bubbles grow up to ten times their emergence size during 15 ms, and move at velocity around 50 cm/s. Mathematical modeling is performed as a new approach to predict where active bubbles are and how intense cavitation is. Then, a first computation, based on Euler's linear acoustic equations, is used to calculate the pressure field in the case of a cylindrical sonoreactor. A second calculation based on fluid dynamics equations is undertaken as CFD codes are very interesting because they also provide velocity and temperature fields, and two phase flows (liquid and bubbles) could be modeled. The comparison with experimental observations (photographs, high-speed film) and measurements (Particule Image Velocity, temperature) shows good agreement with both calculations.

3:15

4pPAa8. Cavitation and capillary wave from a parametric decay scheme. Masanori Sato (Honda Electron. Co., Ltd., 20 Oyamazuka, Otwa-Cho, Toyohashi, Aichi, 441-31 Japan) and Toshitaka Fujii (Toyohashi Univ. of Technol., Hibari-ga-oka, Tempaku-Cho, Toyohashi, Aichi, 441 Japan)

Acoustic cavitation in liquids and capillary waves which cause ultrasonic atomization are considered to be related to the 1/2 subharmonic of ultrasonic surface waves via parametric decay instability. The cavitation bubble oscillation mode is an asymmetric surface bubble oscillation that does not easily emit ultrasonic waves into water and accumulates acoustic energy from longitudinal waves. The capillary waves accumulate acoustic energy in a surface of liquid, enough to be used not only for atomization but also the separation of water-ethanol solution. These were confirmed by experiments using ultrasonic longitudinal waves in water.

3:30

4pPAa9. An optical transducer for the study of the pressure field around a collapsing cavitation bubble. David C. Emmony and Robin D. Alcock (Dept. of Phys., Loughborough Univ., Loughborough, Leicestershire LE11 3TU, UK, DCEmmony@lboro.ac.uk)

An optical reflection transducer using a critical angle technique has been built to study the pressures developed around a single cavitation bubble in water. The bubble was generated at different distances from a planar glass surface, and total internal reflection at the glass water interface gives the pressure through the change in refractive index in both the water and the glass. A simple hemispherical pmma block system has been built to prove the feasibility of the system. In this case the change of the critical angle is not linear, but the use of a modified equilateral glass prism gives the pressure as a function of time around cavitation bubbles with maximum radii of approximately 1.0 mm and gamma values down to 0.3 with a temporal resolution only limited by the photodiode detection electronics ($\sim$10 ns). The transducer shows the development of positive pressure as well as the migration of the bubble to the surface and is capable of recording the rarefaction waves due to free surface reflection.

THURSDAY AFTERNOON, 25 JUNE 1998 METROPOLITAN BALLROOM (S), 12:45 TO 3:45 P.M.

Session 4pPAb

Physical Acoustics: General Topics (Poster Session)

Michael R. Bailey, Chair
Applied Physics Laboratory, University of Washington, 1013 NE 40th Street, Seattle, Washington 98105

Contributed Papers

All posters will be on display from 12:45 p.m. to 3:45 p.m. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 12:45 p.m. to 2:15 p.m. and contributors of even-numbered papers will be at their posters from 2:15 p.m. to 3:45 p.m. To allow for extended viewing time, posters will remain on display until 12:00 noon on Friday, 26 June.

4pPAb1. Enhancement of acoustic cavitation effects by simultaneous multifrequency excitation. Qiping Fang, Beixing He, and Zhongmao Lin (Inst. of Acoust., Chinese Acad. of Sci., Beijing 100080, PROC, qpfang@public.fhnet.cn.net)

Acoustic cavitation can produce many effects. Promoting or inhibiting cavitation depends on different practical purposes. The production of as many cavitation activities as possible is always expected in sonochemistry. Here, the enhancement of acoustic cavitation effects by simultaneous multifrequency excitation, with frequencies in the power ultrasound ranges (20–100 kHz), is studied. One ultrasonic transducer is used to produce multiple ultrasonic frequencies at the same time, rather than several transducers which work at respective frequencies. A probe hydrophone is utilized to measure the subharmonic emission of cavitation and distribution of the acoustic field. The chemical effect of cavitation is determined by the method of iodine release. The results show that simultaneous multifrequency excitation can reduce the cavitation threshold and enhance the chemical effect of cavitation. On the basis of the cavitation bubble dynamics theory, the theoretical analysis proposed attempts to explain this phenomenon, and further discussions are presented. [Work supported by National Natural Science Foundation of China.]

4pPAb2. Nonlinear wave propagation in cylindrical air-filled tubes. Ludovic Menguy and Joël Gilbert (Inst. d'Acoustique et de Mécanique, LAUM UMR CNRS 6617, Ave. O. Messian, BP 535, 72017 Le Mans Cedex, France)

Three dimensional conservation laws are reformulated in a dimensionless form for a cylindrical air-filled tube. Three nondimensional numbers therefore appear, and are used to evaluate the order of magnitude of each term. Simplifying leads to two quasi-unidimensional generalized Burgers equations, which take into account both nonlinear phenomena and viscothermal losses. Special attention is paid to the validity of the different approximations we proposed. The method of multiple scales indicates that the interaction between the two opposite propagative waves is noncumulative and remains a *local* effect. The two Burgers equations are therefore *globally* independent. A numerical computation in the frequency domain solves the Burgers equations in the case of a stationary wave. The signal is supposed to be known at the two ends of the tube (or impedance instead of signal for one extremity). The computation uses the harmonic balance method to calculate the two opposite propagative waves which makes it possible to verify the right conditions in the two extremities. The signal everywhere in the tube is so pieced together. Some experiments are performed with a siren as source in one extremity, and measurements are compared with theoretical results.

4pPAb3. Numerical simulation with experimental validation for nonlinear standing wave phenomena. Zhichi Zhu, Anqi Zhou, Dongtao Huang (Dept. of Eng. Mech., Tsinghua Univ., Beijing, 100084, PROC, zzc-dem@mail.tsinghua.edu.cn), and Ke Liu (Inst. of Acoust., Academia Sinica, Beijing 100080, PROC)

The study of nonlinear standing waves is of great significance to nonlinear acoustics. The strong nonlinear standing wave phenomena are constantly encountered in the experimental studies of acoustics, such as multiplying growth of higher harmonics, saturation of harmonics, bifurcation, and chaos. The experimental study of strong nonlinear standing waves is highly restricted by the sound source, which is very difficult to build up when its sound power becomes very strong. Therefore, in this paper, a numerical approach to the problem was adopted. By the use of Euler equations and MacCormack fourth-order difference method, the multiplying growth and saturation of higher harmonics in a nonlinear standing wave have been numerically simulated. By increasing the sound-pressure level of the excited source from 125 to 190 dB, the entire development process of the nonlinear standing wave was clearly illustrated. Some new interesting results were obtained from the simulation. For example, the sound-pressure level at zero frequency enlarges quickly with increasing the intensity of the excited source. A detailed comparison between numerical simulation and the relevant experimental results shows that the numerical investigation is successful. [Work supported by NSFC.]

4pPAb4. Acoustics wave propagation from an underground waveguide. Stanislav A. Kostarev (Tunneling Assoc., Sadovaja-Spasskaja 21, 107217, Moscow, Russia)

A simple enough model of an underground acoustic waveguide is proposed. A special structure of nonhomogeneity of the subsurface layer in the soil may lead to the capture of acoustic waves within a channel. In this case, a sound signal will propagate with a sufficiently small decay in comparison with homogeneous media (due to the lack of discrepancy of the wavefront). In this case vibration is observed at large distances from the source. A maximum value of the acoustic level is reached near the axes of the waveguide. In practice, this effect may be significant for the case of vibration generation for buildings on the piles. Under these conditions, enhanced values of vibration may take place even when the vibration of the ground surface is negligible.

4pPAb5. Diffraction of a sound wave on an open end of a half-infinite waveguide with impedance walls and impedance flanges. Evgeni L. Shenderov (Res. and Development Inst. ''Morfizpribor,'' 46, Chkalovski pr. 197376, St. Petersburg, Russia, shend@fs.spb.su)

Diffraction of a plane sound wave on an open end of a waveguide with impedance walls, connected with impedance flanges, is considered. A plane wave is incident upon the waveguide from open space. As a result of diffraction, part of the sound energy is scattered in the half-space and the other part of the energy penetrates into the waveguide. Results of the solution of this problem can be used for calculation of the amplitude of a sound wave, scattered by a hole, in particular, in the backward direction. Besides, one can calculate the sound energy penetrating into the waveguide. Absorption of the energy in the waveguide can be taken into account by introducing impedances with nonzero real parts. The distortion of the directivity pattern of a system of receivers located in the waveguide can also be obtained. The solution of the problem has been carried out by the integral equation method using Green's function of the half-space with an impedance boundary. The equation was solved by expanding an unknown function in the eigenfunctions of the waveguide. It reduces to an infinite set of linear algebraic equations. Results of computation of bistatic diagrams of scattering on the hole and diagrams of backward scattering are presented.

4pPAb6. Dependence on waveguide width of elastic convolver efficiency. Yasuhiko Nakagawa and Suguru Kitabayashi (Faculty of Eng., Yamanashi Univ., Kofu, 400 Japan)

An elastic convolver based on a nonlinear effect in surface acoustic wave (SAW) on a Y-Z$\cdot$LiNbO$_3$ is widely used as a functional device in the area of communication system. Therefore, the efficiency, which is an important parameter for the SAW convolver, is quite low. There have been many reports on electrode structures and convolver materials for improving the efficiency. In this work, theoretical and experimental results are reported for the optimum width of a $(\Delta = v/v)$ waveguide which has high convolver efficiency. The theoretical calculation was obtained by using the scalar potential approximation. The efficiency has a peak at a waveguide width of around 1λ. In order to increase the power density multistrip beam width compressors have been used. Performance of a device having a 4-μs waveguide length, a central frequency of 43 MHz and using 10:1 beam width compressors is described. The validity of the theory is proven experimentally, and, therefore, the optimum waveguide width for the convolver efficiency in a conventional elastic convolver is around one wavelength of the input SAW.

4pPAb7. A new time-domain approach for nonlinear wave propagation: Comparison with the KZK equation approach in the case of an unfocused cw beam. Jahangir Tavakkoli (Inst. of Biomed. Eng., Univ. of Toronto, Toronto, ON M5S 3G9, Canada), Oleg A. Sapozhnikov (Moscow State Univ., Moscow 119899, Russia), Remi Souchon, and Dominique Cathignol (INSERM, 69424 Lyon Cedex 03, France)

A time-domain numerical model for simulating acoustic pulse focusing in soft tissue has already been presented. In this model, the main effects responsible for finite-amplitude wave propagation, i.e., diffraction, frequency-dependent attenuation, and nonlinearity, were taken into account. Here, a comparison between the results of this model with those of the KZK model is presented. Using this model, the acoustic pressure field of a plane circular transducer was simulated in water for a large range of cw-mode input excitation levels. The on- and off-axis pressure-time waveforms and their corresponding harmonic components were calculated using this model and the KZK model, and very good agreement was obtained between the two models. Furthermore, these results were verified by previously published experimental measurements. At present, all implementations of the KZK model are limited to planar or weakly focused sources. This study, along with this group's previous work, reveals the validity of this finite-amplitude wave propagation model for a wide degree of source focusing (from a planar to a highly focused source), and in cw as well as PW modes. It is also shown that the computation time for this model is much less than for the KZK, using the same input parameters.

4pPAb8. Description of nonlinear planar traveling waves in a gas filled tube. Michal Bednarik (CTU-FEE, Technicka 2, 16627 Prague, Czech Republic)

This paper deals with problems for the description of nonlinear planar traveling waves in a gas-filled tube. These waves, which are influenced by the tube-wall boundary layer, can be described by both the Burgers equation and the Khokhlov–Zabolotskaya–Kuznecov's equation with the inclusion of a hereditary integral known as the fractional derivate of order $\frac{1}{2}$. Emphasis is placed on comparison of applicability of both the mentioned equations with respect to experimental data. The applicability of the Burgers model equation is limited by the cutoff frequency of the tube which is exceeded due to the growth of higher harmonic components that is caused by the nonlinear distortion of the primary harmonic shape of waves. The next part of this contribution is dedicated to the presentation of some new approximate solutions of the Burgers equation with a fractional derivate and a brief description of used numerical methods.

4pPAb9. Vibratory gyro-sensor using a trident tuning fork resonator with lateral width set in parallel with a rotationary axis. Yoshiro Tomikawa, Toshiharu Ogasawara, and Norikazu Ishida (Dept. of Elec. Eng. and Information Sci., Yamagata Univ., 4-3-16 Johnan, Yonezawa, Yamagata, 992 Japan)

This paper deals with a new type of piezoelectric vibratory gyro-sensor using a trident tuning fork resonator, which is supported laterally at its base portion, the width direction of which is in parallel with a rotationary axis. The vibratory gyro-sensor is an important and attractive device as an angular velocity sensor, especially, in the vehicle stability control (VSC) system schemed for a practical use in the not so far future. Therefore, many researchers have been engaged in development of such a key device in the VSC system. Some types of piezoelectric vibratory gyro-sensors have also been investigated; for example, two types of vibratory gyro-sensors have been studied using a trident tuning fork resonator: one of them was vertically supported at its base portion and the other was flatly supported at the same portion. The gyro-sensor dealt with here is one of such types of gyro-sensors; however, its operation principle is different from theirs. That is, this new gyro-sensor is aimed at being constructed using two-mode coupling of flexural and torsional vibrations in tuning fork arms. In this paper, the structure and operation principle of such a new trident tuning fork resonator gyro-sensor are described, with simulated results of vibrational characteristics for designing the resonator, and moreover, experimental results to confirm its operation principle are shown.

4pPAb10. Effects of diffraction on the sensitivity of needle-type ultrasonic receivers. Gerald R. Harris, Paul M. Gammell, and Jeffrey M. Porter (FDA/CDRH, 9200 Corporate Blvd., HFZ-132, Rockville, MD 20850, grh@cdrh.fda.gov)

The low-frequency sensitivity of piezoelectric receivers usually is assumed to be represented accurately by the -3-dB cutoff frequency $f_{\rm co}=(2\pi R_i C_i)^{-1}$, where R_i is the loading (e.g., amplifier) resistance and C_i is the total capacitance. For typical needle-type hydrophones such as used in medical ultrasound exposimetry, $f_{\rm co}$ is less than 50 kHz. However, theoretical studies have shown that diffraction effects at the needle tip can cause a low-frequency rolloff in sensitivity at frequencies much higher than that predicted by this simple electrical model. To examine this effect, broadband frequency response measurements of several needle-type hydrophones were made in the frequency range 0.2–2 MHz. The active sensor material was polyvinylidene fluoride, and needle diameters ranged from a few tenths of a millimeter to approximately 1 mm. In all cases the sensitivity decreased with decreasing frequency, with the -3 dB points all lying above 400 kHz. Such behavior calls the use of these devices into question when accurate knowledge of the pressure waveform is required, particularly with regard to measuring the peak rarefactional pressure in pulsed waveforms displaying significant finite amplitude distortion.

4pPAb11. Study of a volcano-profiled ultrasound field. Gonghuan Du, Enyu Wang, and Yu Zhang (Inst. of Acoust., Nanjing Univ., Nanjing 210093, PROC)

As is known, the volcano-profiled ultrasound field is of interest in the application of ultrasonic therapy for cancer. This paper indicates that when the radial distribution of vibration velocity on the surface of the source satisfies the expression $q(\xi)=(B\xi^2)^n\exp(-B\xi^2)$, the radiation field, which may be expressed analytically, will maintain approximately the volcano profile in the near field. By employing the theory of the electric field and controlling the shape of the back electrode, a volcano-profiled distribution of the normal component of the electric field and accordingly a same-shaped distribution of the vibration velocity are achieved on the surface of the transducer. In order to improve the radiation efficiency of the transducer, the same material (ceramics) as that of the piezoelectric component, however, unpolarized, is chosen for the back electrode. A once-shaping technique is utilized to make the special-shaped back electrode. The test of the ultrasound field is of satisfying result. The last part of the article gives a brief theoretical description of a nonlinear volcano field. [Work supported by NSF of China.]

4pPAb12. The second harmonic component in the focused sound field diffracted by a straight edge. Shigemi Saito and Jung-Soon Kim (Faculty of Marine Sci. and Technol., Tokai Univ., 3-20-1 Orido, Shimizu, Shizuoka, 424 Japan, ssaito@scc.u-tokai.ac.jp)

To analyze the characteristics of the acoustical imaging utilizing higher harmonic components in the focused sound field, the influence of a diffracting edge put into the focal region on the fundamental and second harmonic fields is theoretically and experimentally investigated. A straight rigid edge put normal to the acoustic axis is assumed to diffract a focused Gaussian beam. The second harmonic sound generated both in front of and beyond the edge is taken into account to calculate the diffracted field through the manner using the Green's function for the governing approximate equation. In the experiment, a knife edge is inserted normal to the sound beam formed by the 1.9-MHz focusing Gaussian source employing a concave piezoelectric transducer with an asteroid electrode. The axial field detected by a needle-type hydrophone as well as the signal obtained with another concave transducer is discussed regarding the amplitudes and phase difference of the fundamental and second harmonic components.

4pPAb13. Acoustic streaming and temperature elevation in a high viscous fluid by irradiation of ultrasound beams. Tomoo Kamakura, Hai-Ying Huang (Dept. of Electron., Univ. of Electro-Commun., 1-5-1, Chofugaoka, Chofu-shi, 182 Japan, kamakura@ee.uec.ac.jp), and Kazuhisa Matsuda (Koganei Tech. High School, Tokyo, 184 Japan)

Sound energy losses due to viscosity and heat conduction in a fluid cause the medium movement as acoustic streaming and cause temperature elevation in beams. In most fluids, the buoyancy force by heat expansion is generally weaker than the driving force of the streaming, so the temperature field is basically determined by the heat transfer equation with convective terms. As some physical parameters such as viscosity depend generally on temperature, the sound field is changed with the irradiation time of cw beams. In addition to the viscosity, two other parameters of sound absorption and sound speed are taken account of their temperature dependency in the present theory. The hydrodynamic flow, forced convection heat transfer, and wave equations are simultaneously solved for axisymmetric ultrasound beams by a finite difference scheme. Numerical examples of the streaming velocity and temperature rise are given. They demonstrate that sound pressure amplitude is intrinsically changed with time and sound self-action is generated. The present method of numerical calculation is easily extended to the problems of focusing and multilayered beam systems.

4pPAb14. Inversion of velocity statistical parameters from travel time measurements. Bertrand Iooss (Ecole des Mines de Paris, Ctr. de Géostatistique, 77305 Fontainebleau, France, iooss@cg.ensmp.fr)

Velocity estimation remains one of the main problems of seismic exploration works. Below a certain range of velocity heterogeneities, deterministic methods become impracticable and the small-scale fluctuations of the velocity field can be described as a random field defined by its first and second statistical moments. It reduces the problem to the determination of a few unknowns, like the mean value, the correlation lengths in the various directions of space, the standard deviation, and the type of correlation function. Theoretical formulas link the statistical moments of an anisotropic random velocity field and those of the travel times of an acoustic plane or spherical wave [A. Ishimaru, ''Wave propagation and scattering in random media'' (1978)]. They are based on the smooth perturbation method of Rytov and on the parabolic approximation which neglects the heterogeneities with scales smaller than the wavelength. Validity domains of these approximations are studied. Therefore, inversion formulas are derived and inversion procedures of the velocity statistical parameters are developed. Finally, comparisons are made between theoretical predictions and synthetic results performed via finite difference algorithms.

4pPAb15. Inverse scattering for porous media with rigid frame. Claude Depollier, Zine el Abidine Fellah, and Achour Aknine (Lab. Acoustique, Univ. Le Maine, B.P. 535, Le Mans Cedex 9, France)

The sound propagation in porous materials having a rigid frame filled by air is well described by the equivalent fluid model. In this framework, the interactions between fluid and structure are taken into account in two response factors, the dynamic tortuosity $\alpha(\omega)$ and the dynamic compressibility $\beta(\omega)$ defined by the equations

$$\rho\alpha(\omega)\frac{\partial\langle\mathbf{v}\rangle}{\partial t}=-\nabla\langle p\rangle,\quad \frac{\beta(\omega)}{K_a}\frac{\partial\langle p\rangle}{\partial t}=-\nabla\langle\mathbf{v}\rangle.$$

These functions are related to the thermal exchanges between fluid and frame and to the geometry of the pores and characterize the sound propagation in the porous media. For homogenous porous materials, these response factors can be measured by different methods, but when the material is inhomogeneous, these methods generally fail. A method is proposed for computing the sound field within a one-dimensional porous medium characterized by a spatially varying tortuosity and compessibility profiles. The spatial variation of the medium is assumed to be along the direction of propagation, i.e., it is stratified. Scattering operators and propagation operators are defined and their properties are studied. An inversion algorithm which leads to the reconstruction of the tortuosity and the compressibility is presented. Numerical results are compared to experimental data.

4pPAb16. A^*-wave spatial resonances on thin cylindrical shells: Experimental study. Loïc Martinez, Jean Duclos, and Alain Tinel (L.A.U.E., Pl. R. Schuman, BP 4006, 76610 Le Havre cedex, France)

An experimental study of a thin cylindrical shell in contact with two different fluids is proposed. Using surface wave analysis methods (SWAM), the A-wave and A^*-wave attenuation and phase velocity are first measured. The experiments show that when the internal fluid has a sound velocity slower than the external fluid, internal reflection of energy inside the shell generates an A^*-wave in multiple space and time positions. Using those results as a basis hypothesis, the 2-D $Ksi(k,\omega)$ wavenumber-frequency response of the fluid filled target is performed. On this Ksi representation, the multiple space A^*-wave echoes cause multiple space resonances on a Ksi-cut versus k. Those numerical results are investigated using SWAM. The so-identified multiple spatial resonances correspond very well to the earliest shell whispering gallery waves families found by Veksler *et al.* in terms of RST.

4pPAb17. Diffraction and conversion of the *A* wave on a *T* structure. Bruno Morvan, Alain Tinel, and Jean Duclos (L.A.U.E, Universite du Havre, Pl. R. Schuman, BP 4006, 76610 Le Havre, France)

The A wave propagates without attenuation in an immersed plane plate. So it can be potentially used in nondestructive inspection for great dimension structure. The A wave propagation is well understood in canonical structures (cylindrical shell, infinite plate). In this paper, a T structure is studied: a brazed perpendicular plate positioned on another one. This stucture is identified as an attached substructure like stiffeners, bulkheads, rib, etc. The experimental study has been performed with a sample made of different metals (stainless steel, brass) and with plates of different thicknesses (frequency-thickness product $\sim$0.5 to 1.5 MHz mm). The A wave is generated at the end of the plate by the conversion of a pulse bulk wave. A second immersed transducer collects the A-wave diffracted signal, at the plate's junction, at a given observation angle. It should be noted that the A wave is reflected and transmitted; it is also converted into a Lamb wave on the principal plate and the perpendicular plate. The spectral analysis of received signal, for different angles, allows the reconstitution of the Lamb wave dispersion curves.

4pPAb18. Formulation of a boundary value problem by a new principle of diffraction. M. Ueda (Dept. of Intl. Development Eng., Tokyo Inst. of Tech., 2-12-1 O-okayama, Meguro-ku, Tokyo, 152 Japan, ueda@ide.titech.ac.jp)

A new diffraction principle, that is formulated from a viewpoint centered at an observation point by considering wave propagation in a space seen by the observer virtually, has been proposed [M. Ueda, J. Acoust. Soc. Am. **95**, 2354–2361 (1994)]. This new approach allows us to grasp a simple understandable concept of diffraction and to formulate the first rigorous and unique representation of diffracted waves, that is, it satisfies both the wave equation and hard or soft boundary conditions for 3D objects. Thus the diffraction problem for scalar waves is solved by this approach. The presentation is expressed in a form of integral equation. Thus it is another problem to formulate the algorithm to find the solution. In this paper the numerical procedures to formulate the algorithm are described. The boundary value problem is formulated by a new principle of diffraction. The structure of the response function, that shows mutual dependence of potentials at the surface of the object, is made clear by this new formulation and it reveals a special role of the edge point in the diffraction process. The usefulness of the response function is made clear by the analysis of diffraction arisen from a semi-infinite plane.

4pPAb19. Modelization of Kirchhoff scattering by a sound ray algorithm. Jean J. Embrechts (Natl. Fund Sci. Res., Dept. of Acoust., Univ. Liege, Sart Tilman, B28, B4000 Liege 1, Belgium, jjembrechts@ulg.ac.be)

The problem of the modelization of sound scattering by rough surfaces arises in room acoustics computer simulations, when sound diffusion is taken into account, as well as in other acoustical fields of interest. It is shown here how a sound ray algorithm can be applied to simulate a scattering process obeying the Kirchhoff tangent plane approximation. The Kirchhoff formulation for a random rough surface is briefly recalled, with the shadowing function proposed by Bass and Fuks. The scattered intensity is directly proportional to the statistical distribution of the surface slopes, which here is kept as general as possible. Then, it is established that this formulation is strictly equivalent to the result of a suitable sound ray algorithm, when the wavelength approaches zero in the Kirchhoff formulation. This is demonstrated in theory, and then applied to several Gaussian rough surfaces. Moreover, hints are given to extend the mathematical analysis of the sound ray technique to the second-order reflections on the rough surface. This is viewed as a way to take into account multiple scattering, which is neglected by the Kirchhoff approximation. [Work supported by FNRS, Natl. Fund for Scientific Research.]

4pPAb20. Analytical method for radiation and scattering problems in noncanonical domains. Victor T. Grinchenko (Inst. of Hydromechan. NAS of Ukraine, 8/4 Zhelyabov St., 252057 Kiev, Ukraine, vgr@ihm.kiev.ua)

One of the tendencies in modern acoustics is to use analytical solutions of the Helmholtz equation to present the sound field around radiators and scatterers of noncanonical shapes. The T matrix and internal source density methods are examples. In the present paper a method called a partial domain method is discussed. An essential point of the method is the notion of general solution of the boundary-value problem for the Helmholtz equation. The boundaries of the objects under consideration are parts of coordinate surfaces in separable systems. However, surfaces may be either coordinate surfaces of different types (as in the case of a finite cylinder) or coordinate surfaces of different systems (as in the case of a cylinder with spherical lids). Theoretical problems of both the method and numerical implementation of one are discussed. Consideration of such aspects of the method as singularity of the field near corner points, nonuniqueness of the wave functions, and extension of boundary conditions gives a basis to make the computation process more effective. The numerical results for some cases of radiation by finite cylinder and cylinder with hemispherical lids are used to illustrate the method.

Session 4pPP

Psychological and Physiological Acoustics: Auditory Attention II (Poster Session)

Christine R. Mason, Chair

Department of Communication and Hearing Research, Boston University, Boston, Massachusetts 02215

Contributed Papers

All posters will be on display from 1:00 p.m. to 5:00 p.m. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 1:00 p.m. to 3:00 p.m. and contributors of even-numbered papers will be at their posters from 3:00 p.m. to 5:00 p.m. To allow for extended viewing time, posters will remain on display until 5:00 p.m. on Friday, 26 June. A cash bar will be set up near the poster session from 5:00 p.m. to 7:30 p.m. preceding the banquet. In order to provide extended viewing of posters in session 5aPP, authors of posters in that session have been requested to display their posters beginning at 1:00 p.m. in Grand Ballroom II.

4pPP1. The auditory attentional blink in a congenitally blind population. Kim M. Goddard, Elzbieta B. Slawinski, and Matthew I. Isaak (Dept. of Psych., Univ. of Calgary, Calgary, AB T2N 1N4, Canada, kgoddard@acs.ucalgary.ca)

The attentional blink (AB), a temporary information processing deficit that follows an attended-to target, has been demonstrated most recently using pure-tone stimuli in a rapid auditory presentation (RAP) paradigm. The current study compared this auditory AB in a group of congenitally blind individuals to nonblind individuals. RAP streams were created using 25 equally loud, randomly generated 1000–2490-Hz tones (11 tones/s). Tones were 85 ms in duration and were separated by a 5-ms silent interval. Target and probe tones were 1500-, 2000-, or 2500-Hz tones increased in intensity by approximately 10 dB SPL relative to the stream tones. In the experimental condition, participants identified the target and probe according to pitch (low, medium, or high). In the control condition, participants performed only the probe task. Probe detection accuracy was measured. Results revealed that in the control condition, participants reported a single loud tone with near ceiling accuracy. In contrast, in the experimental condition, probe detection accuracy was impaired for approximately 270 ms following the target and this impairment was significantly attenuated in the blind population. Findings are discussed in terms of compensatory attentional and temporal processing mechanisms. [Work supported by NSERC.]

4pPP2. The use of visible speech cues (speechreading) for directing auditory attention: Reducing temporal and spectral uncertainty in auditory detection of spoken sentences. Ken W. Grant and Philip F. Seitz (Walter Reed Army Medical Ctr., Army Audiol. and Speech Ctr., Washington, DC 20307-5001)

It is well established that auditory-visual speech recognition is far superior to auditory-only speech recognition. Classic accounts of the benefits of speechreading to speech recognition treat auditory and visual channels as independent sources of information that are integrated early in the speech perception process, most likely at a precategorical stage. The question addressed in this study was whether visible movements of the speech articulators could be used to improve the detection of speech in noise, thus demonstrating an influence of speechreading on the processing of low-level auditory cues. Subjects were required to detect the presence of spoken sentences in noise under three conditions: auditory-only, auditory-visual with a visually matched sentence, and auditory-visual with a visually unmatched sentence. The potential benefits of congruent visual speech cues to auditory detection will be discussed in terms of a reduction of signal uncertainty, in both temporal and spectral domains. In effect, the visual channel may serve to inform the auditory system about when (in time) and where (in frequency) to expect changes in the acoustic signal. [Work supported by NIH Grant DC00792.]

4pPP3. Order effects in the measurement of auditory thresholds during bimodal divided attention. Vishakha W. Rawool (Commun. Disord. and Special Education, Bloomsburg Univ., Bloomsburg, PA 17815)

This study investigated the effect of a visual distraction task on the detection of warbled tones presented at 0.5, 1, 2, and 4 kHz in two orders of presentation. Fourteen subjects responded to warbled tones in two conditions. In the first (attention) condition, the subjects pressed a button everytime they heard the tone. In the second (distraction) condition, they were asked to respond to the tones by turning their heads towards the speaker while solving a cardboard jigsaw puzzle as quickly as possible. The order of attention and distraction conditions was random. Following each correct response, visual reinforcement was presented. For seven of the subjects, the order of presentation of the stimuli was 4, 2, 1, and 0.5 kHz (order 1). For the remaining seven subjects, the order of presentation was 0.5, 1, 2, and 4 kHz (order 2). The same order was maintained in the attention and distraction conditions. Results showed a significant order, attention–distraction, and frequency interaction. The results suggest that distraction can worsen the thresholds at the beginning of the task, but it may cause improvement in thresholds over time. These results will be discussed with reference to dual task attention limits and intersensory facilitation.

4pPP4. The role of memory in the dual-task: Evidence from a frequency/amplitude judgment. Erick Gallun, Anne-Marie Bonnel, and Ervin R. Hafter (Dept. of Psych., Univ. of California, Berkeley, CA 94720, erick@ear.berkeley.edu)

Subjects judged changes in the amplitude and/or frequency of a 100-ms signal either within a trial (experiment one) or between trials (experiment two). Within-trial changes featured roving levels and frequencies, whereas between trial changes were always from the same pool of four stimuli. The cost of making two judgments as opposed to only one was measured by comparing performance in the three conditions (judge amplitude, judge frequency, or judge both). The cost of roving levels and frequencies was measured by comparing performance between the two experiments. It was found that two judgments resulted in a decrease in performance only when the rove was not employed. This agrees with findings in the bimodal dual-task suggesting that the cost of dividing attention is a result of the use of long-term memory, and that the sensory-trace does not incur a cost in the dual-task. [Research supported by the Nat. Inst. of Health (NIDCD 07787), USA, Univ. of California, USA, and CNRS, France.]

4pPP5. Monitoring the simultaneous presentation of multiple spatialized speech signals in the free field. W. Todd Nelson (AL/CFBA, 2610 Seventh St., Bldg. 441, Wright-Patterson AFB, OH 45433-7901), Robert S. Bolia (Systems Res. Labs., Dayton, OH 45440), Mark A. Ericson, and Richard L. McKinley (AL/CFBA, Wright-Patterson AFB, OH 45433-7901)

The effect of spatial auditory information on listeners' ability to detect, identify, and monitor the simultaneous presentation of multiple speech was evaluated in the free field. Factorial combinations of four variables, including the number of localized speech signals, the angular separation of the speech signals, the location of the speech signals along the horizontal plane, and the sex of the speaker were employed using a completely within-subjects design. Participants were required to detect the presentation of a critical speech signal against a background of nonsignal speech events. Speech stimuli were derived from a coordinated call sign test which consisted of a call sign (''Ringo''), a color (''red''), and a number (''five''). In addition to having high face validity for aviation communication tasks, this measure has been successfully employed in competing message experiments. The experiment was conducted at the USAF Armstrong Laboratory's Auditory Localization Facility—a 277-speaker geodesic sphere housed within an anechoic chamber. Performance efficiency was evaluated in terms of percentage of correct detections and false alarms, reaction time, and signal detection theory indices of perceptual sensitivity and response bias. Implications for the design of spatial auitory displays to enhance communication effectiveness and situation awareness are discussed.

4pPP6. Sequential interactions in discrimination of target frequency increments trailed by irrelevant frequency increments: Effects of target duration and target-irrelevant frequency separation. Blas Espinoza-Varas and Hyunsook Jang (Commun. Sci. and Disord., Univ. of Oklahoma Health Sci. Ctr., Oklahoma City, OK 73190, blas-espinoza-varas@UOKHSC.edu)

Discrimination thresholds were measured for target frequency increments trailed by irrelevant frequency increments. Pairs of pure tones (t_1,t_2) separated by 20 ms of silence were presented on each interval of a three-interval, two-alternative, forced-choice task. Duration was 40 or 80 ms for t_1, and 80 ms for t_2; frequency was 1500 Hz for t_1, and either 500, 1500, or 2500 Hz for t_2; level was 70 dB SPL for t_1 and t_2. On each trial, a ''standard'' pair of tones (interval 1) was followed by two ''comparison'' pairs (interval 2 and 3). One (randomly chosen) ''comparison'' pair contained an adaptively varied target increment in the frequency of t_1. In addition, both comparison stimuli contained identical irrelevant increments in t_2 frequency. Listeners had to determine which ''comparison'' interval contained the target increment. Irrelevant frequency increments elevated target-frequency discrimination thresholds. Threshold elevation was greater with 40-ms than with 80-ms targets. Equal-frequency conditions (1500 Hz for t_1 and t_2) yielded larger threshold elevation than unequal frequency conditions (1500 and 500 Hz or 1500 and 2500 Hz for t_1 and t_2). Irrelevant-increment effects decreased with training. [Funded by OCAST Project HR4-064.]

4pPP7. Interactions of pairs of target and context tones based on relative frequency relation and degree of frequency uncertainty. Donna L. Neff and Rebecca L. Wrage (Boys Town Natl. Res. Hospital, 555 N. 30th St., Omaha, NE 68131, neff@boystown.org)

Four normal-hearing listeners completed 2IFC sample-discrimination tasks for frequency, in which they judged which of two tone pairs was drawn from the higher of two Gaussian frequency distributions. The center frequencies of the two target distributions were placed at low (400/500 Hz), middle (1128/1410 Hz), or high (3200/4000 Hz) frequency regions. All distributions were equally spaced and equivariant on a logarithmic frequency scale. Pairs of extraneous (to-be-ignored) context tones were added above, flanking, and below the low-, middle-, and high-frequency target regions, respectively. Across conditions, target and context tones used the same frequency regions, but never occupied the same frequency region within condition. Context tones were fixed in frequency, varied between two known frequencies, or were Gaussian distributed. A level jitter was added to derive perceptual weighting functions across stimuli. The results showed little detrimental effect of fixed-frequency context tones, but large individual differences in the effects of known pairs of tones or tones with Gaussian variation. Weighting functions and patterns of frequency interactions among stimuli will be discussed. [Work supported by NIDCD.]

4pPP8. Effects of tonal masker uncertainty on detection. Elzbieta B. Slawinski (Dept of Psych., Univ. of Calgary, Calgary, AB T2N 1N4, Canada, eslawins@acs.ucalgary.ca) and Bertram Scharf (Ctr. de Recherche en Neurosci. Cognit., CNRS, Marseille, France)

The present study was inspired by results of Allen and Wightman [J. Speech Hear. Res. **38**, 503–511 (1995)]. The goal of the present study was to explore the effects of uncertainty of a weak tonal masker (or distractor) on the detection of a tone in noise. The thresholds of three listeners were measured for 1-kHz tone burst (350-ms duration) in broadband noise (300–1800 Hz, at 60 dB SPL). A 2IFC tracking procedure (3 down, 1 up) with six interleaved tracks was used. On each trial, the distractor frequency was selected randomly from six frequencies (525, 800, 925, 1075, 1280, and 1600 Hz) outside of the critical band surrounding the signal. The distractor came on simultaneously with the signal in the signal-plus-noise interval and also at the corresponding moment in the noise-alone interval. The preliminary results indicated that thresholds for the signal increased in the presence of distractors by 3–10 dB, depending on the frequency of the distractors and type of noise. Thresholds were higher for distractors close to the signal frequency, more so in intermittent noise than in continuous noise.

4pPP9. Identification of brief auditory patterns. Gerald Kidd, Jr., Christine R. Mason, and Chung-Yiu P. Chiu (Dept. of Commun. Disord. and Hearing Res. Ctr., Boston Univ., Boston, MA 02215, gkidd@bu.edu)

The identification of brief nonspeech sounds poses a challenging problem for machine-based recognition systems, especially at low signal-to-noise ratios. In contrast, human observers are thought to be remarkably adept at such tasks. However, there are few systematic studies in the literature that have attempted to quantify human psychophysical performance, or the factors that influence performance, for the identification of brief nonspeech sounds in noise. This study examined human identification of brief sequences of tones arranged in six frequency patterns in masked conditions. The stimuli and methods have been described previously [Kidd, Jr. *et al.*, J. Acoust. Soc. Am. **98**, 1977–1986 (1995)]. Several acoustical factors were varied including pattern duration, signal-to-noise ratio, and range of frequencies forming the patterns. Identification performance declined as duration was shortened below 100 ms even when signal energy, or detectability, were equated. For brief patterns at low signal-to-noise ratios, increasing the range of frequencies comprising the patterns improved identification performance. These results provide insight into the limits of human performance in the identification of brief sounds and may prove useful in the design of algorithms for machine-based recognition. [Supported by ONR and NIH.]

4pPP10. Selective attending to auditory streams in complex sequences: Frequency and/or temporal expectations? Carolyn Drake, Renaud Brochard, and Matthieu Adenier (Lab. de Psych. Exp., CNRS URA 316, Univ. Rene Descartes, 28 rue Serpente, 75006, Paris, France)

Both frequency and temporal information influence the perceptual organization of complex sequences: stream segregation is more likely with wider frequency separations and at faster rates. The influence of frequency and temporal information in directing attention to a stream within a complex sequence is investigated. Complex sequences of three subsequences (each with a specific frequency/tempo combination) were preceded by a cue sequence containing different types of information (FT=frequency + tempo, F=frequency, T=tempo, S=silence). The ability to selectively attend to one subsequence was measured by correct detections of a small temporal irregularity in the cued subsequence. When stream segregation was easy (large frequency separations), performance was similar for FT and F (93%), intermediate for T (79%) and random for S (64%): frequency information facilitated selective attending more than temporal information. When stream segregation was hard (small frequency separations), performance was poorer and similar for all conditions (65%): temporal information became predominant when frequency information was less useful. Thus listeners adapt their attending strategies to coincide with the most pertinent information. Results are discussed in terms of diverging predictions of the theories of Bregman [1990] and Jones [1993].

4pPP11. Application of a peripheral model to an attentional filter of missing-fundamental tone. Hiromitsu Miyazono (Dept. of Administration, Pref. Univ. of Kumamoto, 3-1-100 Tsukide Kumamoto, 862 Japan, miyazono@pu-kumamoto.ac.jp), Tsuyoshi Usagawa, and Masanao Ebata (Kumamoto Univ., 2-39-1 Kurokami Kumamoto, 860 Japan)

An attentional filter measured by the probe-signal method is compared with the peripheral auditory filter measured by the notch-noise method. The shape of attentional filter is estimated by the roex (rounded exponential) filter model. However, the attention can be given on a frequency without the signal on the actual frequency. In this study, the attentional filter was measured using a missing-fundamental tone which has no energy on an actual frequency. The missing-fundamental tone was produced by a harmonic signal with 13 components from 1000 Hz to 4000 Hz. The fundamental frequency was set to 250 Hz. The psychometric function is measured on various frequencies and the shift of the function from that of control condition is regarded as the effect of the attentional filter. The filter shape is estimated by the roex model and compared with that of the pure tone. In the frequency region apart from the fundamental, the dynamic range of the filter is smaller than that of pure tone. Near the fundamental, the correlation between the measured value by the probe-signal method and the estimated value by the roex filter for the missing-fundamental tone is smaller than that for the pure tone.

4pPP12. Short-term auditory memory interference: The effect of speech pitch salience. Kazuo Ueda and Naoko Seo (Dept. of Psych., Kyoto Pref. Univ., Hangi-cho, Shimogamo, Kyoto, 606 Japan, h50015@sakura.kudpc.kyoto-u.ac.jp)

The interference effect of speech pitch salience on tone pitch recognition and digit recall was observed. For the pitch recognition task, 11 subjects had to recognize pitches of two harmonic complex tones separated by a 5-s retention interval. During the interval, no sounds or six randomly chosen intervening sounds, either complex tones from a semitone scale or spoken digits, were presented. The speech included natural stimuli and synthesized stimuli with the original and with flattened pitch contours. For the digit recall task, the subjects had to recall serially the intervening digits. They had to perform both tasks in the dual task conditions. The results were: (1) the synthesized speech was equally intelligible as natural speech, (2) both pitch recognition errors and digit recall errors were significantly increased in the dual task conditions compared to the single task conditions, and (3) the synthesized speech with the flattened pitch contours as well as the complex tones had the most deteriorating effect on tone pitch recognition. Thus it became obvious that the weak interference effect of natural speech on tone pitch recognition was due to their pitch ambiguity. [Work supported by Grant-in Aid for Scientific Research and by Nagai Research Fund.]

4pPP13. Changing frequency after-effect of a linear frequency glide. Takuro Kayahara (Intelligent Modeling Lab., Univ. of Tokyo, 2-11-16, Yayoi, Bunkyo-ku, Tokyo, 113 Japan, kayahara@iml.u-tokyo.ac.jp)

After listening to a linear frequency glide in a single direction ten times, subjective frequency change of stationary sine wave in the opposite direction was observed (changing frequency after-effect). Adapting glides had a fixed frequency excursion (ΔF: 960–1040 Hz), five different durations (250, 500, 750, 1000, or 1250 ms), and glided either upward or downward in frequency. Test glides starting from 1000 Hz had three different durations (500, 750, or 1000 ms), and its ΔF increased or decreased according to subject's response. ΔF at subjective stationary frequency was measured using a three-category (''up,'' ''down,'' or ''no change'' in frequency) double-staircase procedure. In the ipsilateral condition (presenting both adapting and test stimuli to the same ear), ΔF in the same direction with the adapting glides was obtained and maximum ΔF (12 Hz) was observed when the duration of each test glide was the same as adapting glides, although, in the contralateral condition, ΔF was very small or none. These results suggest that there is a feature extractor which is selective in changing frequency and its direction at the level before integration of interaural information, and this extractor changes its own property after a relatively short time exposure of a single direction frequency glide.

4pPP14. Dynamic intensity change influences perceived pitch: Attentional differences between musicians and nonmusicians. John G. Neuhoff (Lafayette College, Easton, PA 18042)

The influence of dynamic intensity change on the terminal pitch of ascending frequency glides was tested as a function of terminal frequency and musical experience. Previous work has shown that dynamic intensity change influences perceived pitch [J. G. Neuhoff and M. K. McBeath, J. Exp. Psychol.: Human Percept. Perform. **22**, 970–985 (1996)]. In the current study, musicians and nonmusicians were presented with dynamic tones that changed in both frequency and intensity, and made real-time judgments about perceived pitch change. Results show that for both musicians and nonmusicians, the influence of intensity change on perceived pitch lessened as the terminal frequency approached an octave. Musicians were less susceptible to the effects of intensity change for falling intensity but not for rising intensity. The findings suggest that musicians may have better selective attention to dynamic auditory stimulus dimensions, and the influence of a tonal schema may reduce the influence of intensity change on perceived pitch. The results imply an interplay of musical and nonmusical listening strategies and demonstrate that the two are not mutually exclusive.

4pPP15. Temporal properties of loudness recalibration. Dan Mapes-Riordan and William A. Yost (Parmly Hearing Inst., Loyola Univ. Chicago, 6525 N. Sheridan Rd., Chicago, IL 60626)

Loudness recalibration occurs when a loud (recalibration) tone at frequency $f1$ precedes quieter test tones at frequencies $f1$ and $f2$. Recalibration is a reduction in the perceived loudness of the test tone at $f1$ due to the addition of the recalibration tone. Mapes-Riordan and Yost [J.

Acoust. Soc. Am. **101**, 3170(A) (1997)] showed that the temporal onset of loudness recalibration is relatively fast. The current set of experiments addressed the temporal decay properties of loudness recalibration. The first set of experiments examined the ''local'' temporal characteristics by measuring how the average amount of loudness recalibration varies with the duration of silent gap between the recalibration tone and the first comparison tone. A second set of experiments examined the ''global'' nature of the decay of loudness recalibration by inducing recalibration and then monitoring the decay using an adaptive tracking procedure. The results of these experiments will be discussed in terms of how assimilation processes influence loudness recalibration, what factors make the decay of loudness recalibration relatively slow, and the effect of attention on loudness recalibration. [Work supported by a Program Project Grant from NIDCD.]

THURSDAY AFTERNOON, 25 JUNE 1998 GRAND BALLROOM A (S), 1:00 TO 3:45 P.M.

Session 4pSA

Structural Acoustics and Vibration: Vibrations of Complex Structures I

Donald B. Bliss, Chair

Department of Mechanical Engineering and Material Science, Duke University, Hudson Hall, Science Drive, Box 90300, Durham, North Carolina 27708

Contributed Papers

1:00

4pSA1. Uncertainty in modeling the dynamics of structures in midfrequency range. Kai-Ulrich Machens and Eike Brechlin (Inst. of Tech. Acoust., Univ. of Technol., Berlin, Germany)

Based on measurements of the energy contents in single struts in a complex truss structure (109 struts, 35 joints) [Machens, Dyer. Acustica **81** (1995)], a numerical simulation of the truss dynamics using the direct dynamic stiffness method was performed. Great care was taken to account for flexural, longitudinal, and torsional waves in the struts, moments of inertia and masses of the joints, complex coupling stiffness (translational and rotational) including a loss factor between joints and struts, as well as material damping in the struts. The struts were modeled as Euler–Bernoulli–Beams and additionally as Rayleigh–Timoshenko–Beams. Frequency averaging proved to be important in the midfrequency range as a meaningful comparison with the measured data depends on a high accuracy in modeling and choice of material parameters. The broadband frequency averaged energy in single struts show a high sensitivity towards various factors such as small random variations of the geometry and of the complex coupling stiffnesses as well as the use of first- and second-order beam theory. The results show that a truss, which is built up from simple elements, becomes fuzzy merely by its high number of coupled elements.

1:15

4pSA2. Reduction of fluid-loaded ribbed shells to equivalent uniform structures for mid- to high-frequency structural acoustic analysis. Donald B. Bliss (Dept. of Mech. Eng. and Mater. Sci., Duke Univ., Durham, NC 27708) and Linda P. Franzoni (North Carolina State Univ., Raleigh, NC 27695)

The acoustic and vibratory behavior of fluid-loaded shells with ribs and other structural discontinuities is of great interest in naval applications. These structural discontinuities lead to complex phenomena, especially at mid- to high-frequencies where shell wavelengths are shorter than the discontinuity spacing. The methods of homogenization and local/global decomposition can be used to divide this problem into two parts. In the global problem, periodic discontinuities are replaced by an equivalent distributed suspension with slowly varying properties. This problem can be solved much more efficiently than the original problem since all rapidly varying scales have been removed. The local problem is solved separately and independently, except for amplitude information from the global problem. The local problem formulation provides transfer function information that defines the suspension in the global problem. Once the formulation has been developed for a specific structure, the global problem is solved first, and the local solution can be reconstructed afterward. The two constituent parts are then recombined to form a composite solution. As an illustration, this methodology is used to solve for Bloch waves on a structure with identical periodic discontinuities. The application of the theory to nonidentical discontinuities is also explained.

1:30

4pSA3. Toward a design methodology for equipment emulators in the shock testing of large structures. Pierre DuPont and J. Gregory McDaniel (Dept. of Aerosp. and Mech. Eng., Boston Univ., Boston, MA 02215)

In a variety of situations, an undesired shock excitation is applied to a master structure which supports shock-sensitive equipment. Often, one wishes to design and test a master structure which transmits the least amount of shock energy to the attached equipment. In scaled testing of new designs, a major task is to design and construct ''equipment emulators,'' inexpensive mechanical systems which approximately mimic the dynamic behavior of the actual full-scale equipment *as seen by the master structure*. This presentation will present new methodologies for designing equipment emulators, assessing their fidelity, and interpreting test data taken in the presence of imperfect emulators. Starting with frequency-domain impedance descriptions of the master structure and actual equipment at the attachment points, this work develops sensitivity metrics which directly relate the fidelity of the emulator to its structural complexity. These ideas may provide a path by which experimentalists can efficiently arrive at conceptual designs of emulators which promise a specified degree of fidelity in terms of system velocities and their associated shock spectra. The ideas are illustrated on simple examples and applied to the emulation of commercial-grade electronic cabinets for the testing of novel ship deck structures. [Work supported by ONR.]

1:45

4pSA4. A new finite-element analysis approach for the prediction of minimum and maximum vibroacoustic response of solid–fluid systems. Jean-Sébastien Genot, François Charron, and Noureddine Atalla (Univ. of Sherbrooke, Dept. of Mech. Eng., Sherbrooke, QC J1K 2R1, Canada)

Engineers are aware that identical products do not have necessarily the same dynamic response because of small differences in their dimensions or material properties. The classical finite-element method cannot take account of this fact. Statistical tools are needed. In order to consider the input data variability, new modules have been integrated in a finite-element code to estimate the maximum and minimum response levels (mean quadratic velocity and radiated power) of a structure. Up to now,

numerical modules were generally based on Monte Carlo or perturbation methods. Both have limiting and complementary drawbacks: The Monte Carlo method is accurate but far too slow, even for small finite-element models, while the perturbation scheme is fast, but unable to predict nonlinear behavior of the dynamic response properly with respect to the input variables. The developed approach takes advantage of the response modal decomposition to identify intermediate variables which are nearly linear. The intermediate variables (eigenfrequencies, dynamic factors) are easily interpolated, thus requiring little computation. Results are presented to prove the efficiency of the method.

2:00

4pSA5. FEM aided structural intensity measurement method for thick body. Yoshikazu Koike, Chie Aramaki, Kentaro Nakamura, and Sadayuki Ueha (Precision and Intelligence Lab., Tokyo Inst. of Technol., 4259 Nagatsuda, Midori-ku, Yokohama, 226 Japan)

Many studies about structural intensity(SI) measurement, which is useful to know transmission paths of the vibration energy in a body, have been reported. Most of the works, however, have been done only for thin bodies, such as beams or plates because of the difficulty in measuring interior vibration distribution in thick structures. In this paper, a new SI measurement method applicable to thick bodies is proposed. The entire vibration distribution in the structure is calculated by dynamic analysis of the finite element method (FEM) from the partial surface vibration data if the elastic parameters, the sound velocity, sizes of a body under study, and the locations of excitation and absorption are given *a priori*. The entire SI distribution can be estimated using the results. Based on this method, several SI measurements for thick vibrating bodies have been successfully carried. The merits of this method are as follows. (1) Only surface vibration data of the structure is needed to estimate the entire SI distribution. (2) It is applicable for arbitrary 3-D structures if the conditions mentioned above are satisfied *a priori*. The principle and the experimental results are described in the report.

2:15

4pSA6. Resilient mounting of engines. Anders Nilsson, Leif Kari, and Romain Haettel (MWL, Dept. of Vehicle Eng., KTH, 10044 Stockholm, Sweden)

In any type of vehicle a number of mechanical sources generate structure-borne noise. The energy is transmitted from excitation points to surrounding structures. These radiate noise externally and internally. The most commonly used method to reduce the radiated noise is to mount the sources resiliently to the supporting foundation. The insertion loss due to the resilient mounting can in many cases be extremely poor. This could be due to the break down of the stiffness of the foundation in the audible frequency range. At the same time the stiffness of the mounts could be increased drastically as compared to the static stiffness. A test rig and measurement procedures for determining the dynamic properties of mounts are described. Measured and predicted results are compared for rubber mounts with simple geometries. The insertion loss of a resilient mounting very much depends on the location of the engine feet on a foundation. Case studies relating to passenger vessels and catamarans built with aluminum and sandwiched are presented. Reduction of both noise and weight is discussed.

2:30–2:45 Break

2:45

4pSA7. Assessment of construction vibration impacts on historic structures. Chetlur G. Balachandran (Parsons Brinckerhoff, Inc., One Penn Plaza, New York, NY 10119)

This paper addresses construction-related ground vibration impacts to historic structures located in an area where three options are being considered for providing capacity improvement to an existing roadway. Historic structures within the project area are: a Fort, a Legislature building, an electrical substation, and an auxiliary building. The above fragile structures are more likely to be affected by construction-related ground vibration impacts than by operational traffic-induced vibration impacts. The buildings are in close proximity to the existing heavily traveled roadway and are already experiencing moderate levels of ground vibration from road traffic. Potential vibration effects from heavy construction equipment include annoyance to people inside these buildings and potential for minor damage to the buildings. The particular concern in the present study is to comply with ground-vibration criteria to safeguard identified structures within the project corridor against construction-induced ground vibration. The various national and international criteria that were used to assess vibration impacts together with the results of vibration impact assessment will be presented. Construction of the roadway alternative that would cause the least vibration impact was identified and mitigation was considered.

3:00

4pSA8. Hybrid control for vibration and acoustics. Robert L. Clark (Dept. of Mech. Eng., Duke Univ., Box 90300, Durham, NC 27708) and Dennis Bernstein (Univ. of Michigan, Ann Arbor, MI 48109-2118)

The ''standard problem,'' frequently discussed in the controls literature, is detailed for application to noise and vibration control. Specifically, the feedback, feedforward, and hybrid (combination of feedback and feedforward) control system architectures are developed in the framework of the standard problem. The hybrid measurement structure and problem was formulated to demonstrate that the LQG compensator design for control of stochastic inputs is separable with respect to inputs which can be measured directly. Thus, one can design a feedback controller in the absence of measurable input disturbances if the objective is to combine both feedback and feedforward or adaptive feedforward control for stochastic and measurable inputs, respectively. Once the feedback controller is designed for the stochastic inputs, the feedforward controller required for the measurable input disturbances can be designed in a subsequent formulation of the control problem. Realizing that the design of such compensators is separable serves to simplify the hybrid design process.

3:15

4pSA9. Active control of total structural intensity in a T beam: General case. Sabih I. Hayek and Jungyun Won (Active Vib. Control Lab., Dept. of Eng. Sci. and Mech., Penn State Univ., University Park, PA 16802, sihesm@engr.psu.edu)

The active control of power flow in a T beam was presented at the 134th Meeting of the Acoustical Society of America. In that paper, the mechanical noise source was a shear force in the plane of the T beam. In this paper, the active control of structural intensity in the T beam is achieved for a general shear mechanical noise source that has in-plane and out-of-plane components. The reduction of the total power flow at a control point located midway in the vertical leg of the T beam is achieved through control actuators that are located at one end of the straight part of the T beam. The vector control actuators have components in three-dimensional space in various combinations of normal forces, shear forces, torques, and moments. The efficiency of the structural intensity reduction is considered for local versus global control. Various control strategies are compared which are also aimed at minimizing the cost function that represents the total power requirement for the control process versus the noise input power.

3:30

4pSA10. Vibration isolation of two elastic structures using active compliance at the isolation mounts. Kenneth E. Jones and Y. F. Hwang (Carderock Div., Naval Surface Warfare Ctr., 9500 MacArthur Blvd., West Bethesda, MD 20817-5700)

A mathematical model of the vibration isolation of two elastically connected structures (via isolation mounts) employing active springs is presented. Two elastic structures are connected by an arbitrary number of springs, each made active by the inclusion of a piezoelectric pad at one of

the spring and structure interfaces. Single-input–single-output feedback controllers at each spring support seek to minimize the force transmitted from one structure to the other. Solutions are based upon the formalism of Lagrange's equations using the normal modes of the uncoupled structures as the generalized coordinates. The constraints by the springs and supports are enforced by Lagrange multipliers. A 2-D numerical example of two elastic beams is presented. One beam, driven by a harmonic disturbance force, is coupled to the second elastically supported beam through three active springs. A beam-relative displacement sensor colocated at each spring is the input to an arbitrary feedback filter which can incorporate practical amplifier characteristics and actuator dynamics. Results of the numerical simulation of the feedback control case are discussed.

THURSDAY AFTERNOON, 25 JUNE 1998 GRAND BALLROOM III (W), 1:00 TO 4:50 P.M.

Session 4pSC

Speech Communication: A Half-Century of Speech Research

Patricia K. Kuhl, Chair
University of Washington, Eagleson Hall 204, Box #354875, Seattle, Washington 98195

Chair's Introduction—1:00

Invited Papers

1:05

4pSC1. Fifty-four years in speech research. Gunnar Fant (Dept. of Speech, Music and Hearing, KTH, Stockholm, S-10044, Sweden, gunnar@speech.kth.se)

This is a brief presentation of my scientific career with outlooks on the development of speech research in the last 50 years. It started with an electrical engineering thesis at KTH in 1944–1945. The topic was intelligibility loss as a function of bandwidth reduction in telephony and related problems in assessing the effects of various types of hearing loss. In 1945–1948 I was employed by the Ericsson Telephone Company where I conducted basic studies of the spectral characteristics of Swedish speech sounds. This early period and my two years at MIT, 1949–1951, provided an entry to a pioneering era of speech research with new tools for speech analysis and synthesis and speech production modeling. It was during this time I started my work on the acoustic theory of speech production and established my cooperation with the linguists Roman Jakobson and Morris Halle and with Ken Stevens. I have followed the advent and the progress of speech research through the analog, the digital, and the computer age. Speech research now has the challenge to provide a deeper and more integrated understanding of all levels of the speech communication process. This is the ultimate requirement for the realization of advanced applications.

1:25

4pSC2. Toward models for human production and perception of speech. Kenneth N. Stevens (Res. Lab. of Elec. and Dept. EECS, MIT, 50 Vassar St., 36-517, Cambridge, MA 02139)

Two of the goals of research in speech communication are to develop models of normal speech production and normal speech perception. Related objectives are to uncover the process by which children acquire the knowledge implicit in these models and to determine how the models are modified for disordered speech. Even partial achievement of these goals can have significant practical consequences, including machine recognition and synthesis of speech, and improved methods for diagnosis and remediation of speech disorders. In this paper, a current view of a framework for models of speech production and perception will be described, and some of the steps that have led to refinement of these models over the past 50-odd years will be described. Advances have been made in quantifying acoustic mechanisms of speech production and in specifying the nature of the discrete linguistic representation of an utterance in memory. From studies of speech perception and speech motor control, some understanding has been gained of how properties of the sound are related to the linguistic representation. There are large deficiencies, however, in our understanding of variability in speech due to speaker differences, speaking style, and context. [Work supported in part by NIH Grants DC00075 and DC02525.]

1:45

4pSC3. Junctures in speech communication. J. L. Flanagan (CAIP Ctr., Rutgers Univ., 96 Frelinghuysen Rd., Piscataway, NJ 08854-8088)

Several junctures in research on speech communication seem salient over the past 50 years. Under the goad of expensive long-distance transmission, and the criticality of voice communication in a world in turmoil, bandwidth compression methods received strong emphasis in the 1940s. Because transoceanic communication was by HF radio, privacy and encryption were also central issues. These incentives resulted in the Channel Vocoder and a variety of offshoots. By the 1960s, digital computers were being deployed widely and sampled-data theory was well established, providing the systems researcher with a new environment for experimentation—one in which concepts could be rapidly simulated and evaluated. Bandwidth compression continued to be a focus, as computer simulation techniques progressed from classical filter emulation to new designs based solely on digital theory. By the 1980s, broadband connectivity (including transoceanic cable) was becoming pervasive and economical enough that emphasis on bandwidth conservation waned. Concomitantly, digital machines were expanding in capability—but were markedly limited by their inability to

communicate with human users in natural ways. The focus consequently shifted to human/machine communication—especially by conversational means. As we approach the year 2000, a strong research emphasis remains on human/machine communication—but with expanded aspirations for natural interaction. Computer interfaces with multiple modalities—utilizing sight, sound, and touch in combination—are beginning to serve multiple collaborating users. In the same era, explosive deployment of two technologies—cellular telephony and computer networking—has rejuvenated interest in bandwidth conservation and coding for privacy—with many of the early vocoder concepts evolving to sophisticated digital forms implemented on single-chip processors. As we move toward 2020, a special emphasis is likely to be on multilingual communication. And in this time, continued advances in computing, acoustic signal processing, and language modeling should carry translating telephony far beyond the ''phrase book'' stage.

2:05

4pSC4. Three questions for a theory of speech. Alvin M. Liberman (Haskins Labs., 270 Crown St., New Haven, CT 06511, liberman@haskins.yale.edu)

On reviewing my 50 years of research, I see that in trying to head theory in the right direction I have been led ever more compellingly by the need to find plausible answers to three questions: (1) What is the basis for the parity between primary motor and perceptual representations that explains how, in the evolution of speech, production and perception managed to proceed in lock step, thus allowing the parties to each two-way exchange to communicate, then as now, in a common code? (2) How does speech meet the requirement, unique to its combinatorial mode, that discrete, invariant, categorical, and distinctly linguistic segments be produced and perceived in strings at rates as high as 20 segments per second? (3) What biological and psychological properties account for the vast difference in the difficulty of acquisition and use between speech and the reading or writing of its alphabetic transcription? Two theories of speech—one quite conventional, the other much less so—provide answers that differ greatly in their plausibility.

2:25

4pSC5. Speech research at the I. P. Pavlov Institute in Leningrad/St. Petersburg. L. A. Chistovich (Early Intervention Inst., St. Petersburg, Russia), J. M. Pickett (Windy Hill Lab, Surry, ME), and R. J. Porter, Jr. (Univ. of New Orleans, New Orleans, LA)

The ''Leningrad Group,'' led by L. A. Christovich and V. A. Kozhevnikov, carried out an extended research program to describe relations between speech perception and speech production and build a model of perception. The work first explored auditory processing mechanisms that could afford closely interdependent connections between auditory information and the control signals for motor production. The resulting model for auditory analysis employed amplitude modulation detectors as well as spectral measurements. The model was evaluated in perception tests with listeners mimicking and shadowing speech sounds and was also employed in a spectrographic display and a scheme for automatic speech recognition.

2:45–3:00 Break

3:00

4pSC6. Speech research from acoustic transmission to gestural analysis. Katherine S. Harris (Dept. of Speech and Hearing Sci., City Univ. of New York, 33 West 42 St., New York, NY 10036, loumau@erols.com)

My training was in experimental psychology at Harvard, where the focus of the department's speech work was the characteristics of sounds affecting their error-free transmission, essentially an extension of the Bell Telephone Laboratories' approach to telephone communication. At Haskins, I began research on the fricative sounds, and discovered the important role of the dynamic characteristics of the sounds in distinguishing among them. In turn, this focus led to an interest in the role of formant movement in speech perception, and ultimately to a study of the characteristics of speech movements themselves. Over the last 50-year period, thanks to some of those on this panel, we have developed a good understanding of the static characteristics of the vocal tract as a producer of speech sounds and the acoustics of speech. However, we still have but a rudimentary account of how the moving articulators develop a dynamic output, and how segmental and suprasegmental transformations of the phonetic message affect this output. Recent instrumental advances should allow the development of an understanding of the dynamics of articulation that parallels our understanding of speech acoustics.

3:20

4pSC7. Chasing ideas in phonetics. Peter Ladefoged (Phonet. Lab., Linguist. Dept., UCLA, Los Angeles, CA 90095-1543, oldfogey@ucla.edu)

In my early career I never stayed long enough in a particular field to be contradicted. I started as a poet learning about the sounds of words with David Abercrombie. Then, remembering my background in physics, I moved to studying acoustic phonetics. From there I became a pseudo-psychologist testing perceptual theories, until a meeting with a physiologist led to work on the respiratory muscles used in speech. Eventually I landed in Africa teaching English phonetics and learning about African languages. So, by the time I was asked to set up a lab at UCLA, I was a specialist in nothing. However, I was able to use my background to describe the sounds of a wide range of languages, becoming a sort of linguist. Computers and bright students led to other ways of analyzing sounds. Building a research group who felt that they had a stake in the development of the lab taught me their varied ideas from statistics to engineering, and the philosophy of linguistics. Now, still looking for the growing edge of the field, I think it might be in the physically observable activity of the brain; but perhaps this is because I have never been a pseudo-neurologist.

3:40

4pSC8. Language and speech. Victoria Fromkin (Dept. of Linguist., Univ. of California, Los Angeles, CA 90024)

It is exciting to note that I have been involved for thirty-six of these important fifty years of phonetic research. In 1962 I entered my first class in Phonetics and Phonology taught by the newly arrived Peter Ladefoged from Edinburgh. That course, and Peter himself, set my goals which have lasted all these years—the relationship between the cognitive knowledge that lay behind our ability to speak and understand. The field of phonetics is varied and researchers have their own agenda and questions to answer. While one's goals may be practical or theoretical, the answers we all find help us understand the incredibly unique ability of the human animal to communicate in a special, linguistic way. Whether studying normal or abnormal speech (speech errors and aphasia), child language, or language decay, we gain insights into the complexities of the process. In recent years, due to new technological tools such as CAT, MRI, fMRI, PET, ERPs, etc., we have the chance to ask new questions. But still it takes an understanding of the linguistic systems underlying the ''speech chain'' to know what questions to ask. Linguists therefore are now, as they have always been, important in the research into speech communication.

4:00

4pSC9. Communication between minds: The ultimate goal of speech communication and the target of research for the next half-century. Hiroya Fujisaki (Dept. of Appl. Electron., Sci. Univ. of Tokyo, 2641 Yamazaki, Noda, 278 Japan)

Needless to say, speech is a manifestation of the information intended by the speaker and is perceived by the listener who tries to retrieve it. Elucidation of the entire process of speech communication thus requires a multidisciplinary approach, as was eminently illustrated by the work of Chiba and Kajiyama about a half-century ago [*The Vowel, Its Nature and Structure*, Tokyo-Kaiseikan (1941)]. Through the efforts of distinguished researchers, supported by the rapid progress in electronics and computers as well as in other means, the ensuing half-century has seen remarkable advances in our understanding of the linguistic, physiological, articulatory, acoustic, sensory, and cognitive processes in normal, developmental, and pathological cases, as well as in technologies for its utilization. As a new millenium of man–machine symbiosis is entered into, however, still deeper understanding becomes necessary in order to establish a friendly human–machine interface through spoken language, often in multimodal and multilingual environments. In the author's view, this can only be achieved by investigating and modeling the mind itself as the ultimate source and destination of information. To that end, speech science will need closer cooperation with related disciplines such as brain science, cognitive science, and artificial intelligence.

4:20–4:50 General Discussion

THURSDAY AFTERNOON, 25 JUNE 1998

CASCADE BALLROOM I, SECTION B (W), 1:30 TO 5:20 P.M.

Session 4pSP

Signal Processing in Acoustics, Engineering Acoustics, Architectural Acoustics and Education in Acoustics: Acoustics in Multimedia—Systems Issues II

David I. Havelock, Chair
National Research Council, M-36 Montreal Road, Ottawa, ON K1A 0R6, Canada

Invited Papers

1:30

4pSP1. Acoustics and psychoacoustics of workstation audio systems. Floyd E. Toole (Harman Intl. Industries, Inc., 8500 Balboa Blvd., Northridge, CA 91329, ftoole@harman.com)

Workstations are widely used as production environments for multimedia, film, and audio programs. For many consumers, they are also the delivery systems for these programs, as well as for others, like music, television, and films, intended for reproduction in various conventional systems and environments. Listeners in workstations therefore must be able to perceive sounds with the appropriate sense of timbre, direction, distance, and ambiance whether the program was created in two-channel stereo, or any of the multichannel surround or binaural 3-D formats. The space constraints, and the usual restriction to two loudspeakers, put special demands on the design of these audio systems. Binaural image-steering techniques will be the basis for most interactive games and, through crosstalk-cancellation, binaural technology will enable realistic replications of two, four, and five-channel audio for music, films, etc. This paper reviews the requirements for workstation audio systems, discusses the key enabling technologies, and summarizes what is known about the appropriate technical specifications, measurement techniques, and subjective evaluation criteria and methodologies for evaluating workstation audio systems.

2:00

4pSP2. Using a personal computer platform to develop an information-rich learning environment for instruction in acoustics. Robert Celmer (Acoust. Prog. and Lab., College of Eng., Univ. of Hartford, 200 Bloomfield Ave., West Hartford, CT 06117, celmer@uhavax.hartford.edu)

Teaching students the subject of acoustics involves large amounts of technical information, including equations, derivations, and example problems. It also requires the explanation of a multitude of concepts, many of which are often difficult for students to grasp and equally frustrating for the instructor to disseminate. Part of the difficulty can be traced to the use of *text* (a visually based medium) to teach about *sound* (an aurally based medium). This presentation will describe the development of multimedia techniques for in-class presentation of acoustics instruction. Some of the materials were developed using certain authoring applications, drawing and animation programs, sound manipulation software, as well as 3D CAD and spectral analysis applets. In addition to the classroom materials, self-paced exercises have been developed to review the material in a student-centered learning environment, along with scheduled interaction time with the instructor for questions, clarifications, and test administration. Demonstrations of the materials for the instruction of acoustical concepts, as well as case studies, will be presented.

Contributed Papers

2:30

4pSP3. The impact of system latency on dynamic performance in virtual acoustic environments. Elizabeth M. Wenzel (NASA–Ames Res. Ctr., M.S. 262-2, Moffett Field, CA 94035-1000, bwenzel@mail.arc.nasa.gov)

Engineering constraints that may be encountered when implementing interactive virtual acoustic displays are examined. In particular, system parameters such as the update rate and total system latency are defined and the impact they may have on perception is discussed. For example, examination of the head motions that listeners used to aid localization in a previous study suggests that some head motions may be as fast as about 175°/s for short time periods. Analysis of latencies in virtual acoustic environments (VAEs) suggests that: (1) commonly specified parameters such as the audio update rate determine only the best-case latency possible in a VAE, (2) total system latency and individual latencies of system components, including head-trackers, are frequently not measured by VAE developers, and (3) typical system latencies may result in undersampling of relative listener-source motion of 175°/s as well as positional instability in the simulated source. To clearly specify the dynamic performance of a particular VAE, users and developers need to make measurements to average system latency, update rate, and their variability using standardized rendering scenarios. Psychoacoustic parameters such as the minimum audible movement angle can then be used as target guidelines to assess whether a given system meets perceptual requirements.

2:50

4pSP4. Immersive audio for desktop systems. Chris Kyriakakis, Tomlinson Holman, Hartmut Neven, and Christoph von der Malsburg (USC, 3740 McClintock Ave., EEB432, Los Angeles, CA 90089-2564)

Numerous applications are envisioned for integrated media workstations to create, edit, and accurately monitor digital video and audio. There are two major classes of limitations that impede the performance of current desktop loudspeaker-based sound systems. The first encompasses problems that arise due to the local acoustical environment, such as early reflections from the CRT and nearby flat surfaces, as well as the design characteristics and performance of the loudspeakers. The second class involves limitations that arise from variations in human listening characteristics and listener movement relative to the loudspeakers. In this paper several findings are presented that are based on acoustical, psychoacoustical, and signal processing methods for delivering accurate sound field representations. Furthermore, a novel vision-based method for accurate listener tracking is presented that eliminates the need for headgear or tethered magnetic devices. Current findings show that this algorithm can be implemented without imposing a significant computational overhead to the host processor. Research directions include adaptive real-time HRTF synthesis based on the tracking information, as well as vision-based pinna shape classification for improved performance.

3:10–3:20 Break

3:20

4pSP5. Distributed-mode loudspeakers and their impact on intelligibility in multimedia and sound distribution. Peter Mapp (Peter Mapp Assoc., Colchester, UK) and Henry Azima (New Transducers Ltd., Stonehill, Huntingdon, PE18 6ED, UK)

A new class of acoustic radiator has been investigated and shown to exhibit unusual radiation characteristics that can produce enhanced intelligibility under given listening conditions. The improvement mechanisms are investigated and shown to be related to a number of unique features. These include transient response, synergetic decay, lower inherent distortion, directivity, and the diffusivity of acoustic radiation. In a sound system design, it is desirable to produce an even coverage in both spatial and spectral domains. Conventional radiators, the majority of which work under a pistonic regime, exhibit two important characteristics that limit their effectiveness in real-world applications. These are the well-known narrowing of the acoustic radiation (dispersion) angle with increasing frequency and the coherency of the radiation. In contrast, the distributed-mode radiators, an emerging new technology, tend to overcome these obstacles and offer a new approach to improved sound distribution and multimedia. They combine a flat power response with an essentially constant and wide directivity (independent of frequency), augmented by a spatially and temporally diffuse radiation characteristic. The fast transient response results in a signal broadcast with very high clarity. The diffuse nature of the radiation suggests a lower degree of boundary interaction and spectral interference.

3:40

4pSP6. The measurement of coloration in electroacoustic enhancement systems. Mark Poletti (Industrial Res. Ltd., P.O. Box 31-310, Lower Hutt, New Zealand)

The use of electroacoustic systems to enhance the acoustics of concert halls is becoming increasingly prevalent. Regenerative systems are often used for obtaining a source position independent enhancement of reverberation time. However, these systems can produce unnatural acoustics due to the unequal enhancement of the response at different frequencies, a phenomenon termed coloration. Coloration alters the natural Rayleigh statistics of the transfer function magnitudes, and the variation in statistics can be used as a method for quantifying it. This approach avoids the difficulties of examining only the peaks in the transfer function magnitude. A method is developed for measuring the deviation of room transfer functions from Rayleigh statistics. Variance measures are derived. The underlying probability density function is modeled by fitting a three-parameter density termed the generalized gamma function. The maximum deviation of the modeled cumulative distribution from the Rayleigh distribution provides an alternative measure of coloration. The sensitivity of the measures is improved by eliminating the dominant early part of the impulse response, which is largely unaffected by non-in-line systems. An assisted reverberation system simulator is used to simulate the VRA system, which includes a unitary multichannel reverberator in the feedback loop to provide room volume expansion. The simulations show that the coloration measures increase with loop gain as expected, and are independent of other parameters in the system such as the secondary room reverberation time.

4:00

4pSP7. Criteria for accurate acoustic emulation. David McGrath (Lake DSP pty. Ltd., Sydney, Australia)

The technique of auralization was pioneered by acoustics researchers in the field of concert hall simulation, where an accurate rendering of the acoustics implied that all characteristics of an acoustic space were reproduced, wherever possible. As the convolution technology is pushed into more cost-effective implementations, the broader potential for this technology is being realized. Now the criteria for acoustic emulation must be reevaluated to take into account a number of new factors. This paper explores many relevant criteria for repeatability testing and characterizing the performance of equipment in the presence of realistic three-dimensional noise fields and complex acoustic environments. Such an acoustic system can be accurately emulated using real-time simulation hardware, if a complete characterization of the system can be obtained. This paper discusses methods that are appropriate for the implementation of accurate acoustic emulation, with the emphasis on real world problems and solutions. It is shown how the entire acoustic signal path can be replaced by an appropriate low latency FIR filter, defined using new techniques for impulse response measurement.

4:20

4pSP8. Multichannel audio signal compression and quality assessment for AV communications. Jin-Woo Hong, Dae-Young Jang, and Seong-Han Kim (Electron. and Telecommunications Res. Inst., Yusong, P.O. Box 106, Taejon, 305-600, Korea, jwhong@etri.re.kr)

This paper describes an algorithm for multichannel audio signal compression using the psychoacoustic model, an implementation and the result on quality assessment of real-time operating multichannel audio codec, and a multiplexing technique of synchronization for AV communication. The algorithm which is based on the MPEG-2 audio standard, utilizes psychoacoustic modeling, sub-band analysis and synthesis, and bit allocation for information as key technology. The audio codec was implemented on real-time operation by using a flexible DSP system in which an encoder and a decoder are independently designed for processing a formatted digital audio bit stream, and can compress the multichannel audio data of about 3848 kbits/s into 384 kbits/s. A subjective test for quality assessment of multichannel audio codec was performed using double blind and triple stimuli with hidden reference method. In the test condition with 17 subjects and ten test materials, the test result with 95% confidence intervalshows that all test materials except one (pitch pipe sound) are awarded mean difference grades better than -0.6 and this audio codec is acceptable for AV communication or broadcasting. Finally, a multiplexing method to control time stamp for synchronization of AV communication is, in this paper, introduced from the implementation of HDTV system using this codec.

4:40

4pSP9. Reproduction of CD by providing ultrasonic atmosphere. Jouji Suzuki, Shin'ichiro Kaneko (Dept. of Information and Comput. Sci., Saitama Univ., 255 Shimo-okubo, Urawa, 338 Japan), and Kiyomitsu Ohno (Nihon Tech. Ctr., Tachikawa, Tokyo, 190 Japan)

It is said that the LP sound reproduced by a well-prepared system is superior to the sound of a CD and the difference stems from whether the signal is recorded by analog or digital form. Considering master tape of LP is recorded in digital form, the main difference exists in the bandwidth of the reproduced signal. The signal of a CD is restricted to 20 kHz. On the other hand, the spectrum of an LP signal is extended to 100 kHz. Though the upper limit of hearing acuity of a human is 20 kHz, superior components will contribute to offer more rich and realistic atmosphere. In order to produce a higher-frequency component from CD sound, the folding spectrum of the sampling frequency of 44.1 kHz is utilized. The output signal of a DA converter of a CD player is filtered with BPF of 24–42 kHz. This signal is added to the CD output and presented to listeners. This system is named PULSA (providing ultrasonic atmosphere). A preference test is conducted on conventional CD sound and the sound by PULSA through a wideband static headphone on three minuets of orchestra and opera. It is revealed that PULSA sound is better than the conventional by the ratio of 65 to 35.

5:00

4pSP10. A multi-processor system for the production of virtual sound fields. Felipe Orduña-Bustamante, Ricardo R. Boullosa, and Antonio P. López (Centro de Instrumentos UNAM, Circuito Exterior CU, CP 04510, México DF, Mexico, felipe@aleph.cinstrum.unam.mx)

A multi-processor system is described for the production of virtual sound fields using small arrays of up to four high-quality loudspeakers and up to four control microphones distributed around an artificial head and torso in a lightly damped listening room. The digital signal processing engine consists of four floating point processors interconnected for parallel computation, together with digital and analog converters. The performance of the system is analyzed in terms of the tradeoff between the maximum sampling rates that can be used and the complexity of the digital filter matrices in the different operating modes of the system such as system identification, single-input multiple-error adaptive filtering, and full multichannel filtering with fixed coefficients.

4p THU. PM

Session 4pUW

Underwater Acoustics: 3-D Propagation Effects II: Where are We Today in Models and Measurements?

Michael B. Porter, Cochair
Scripps Institute of Oceanography, Marine Physical Laboratory, MC 0205, 8602 La Jolla Shores Drive, La Jolla, California 92037-0205

Alexandra I. Tolstoy, Cochair
4224 Walalae Avenue, Suite 5-260, Honolulu, Hawaii 96816

Chair's Introduction—1:00

Invited Papers

1:05

4pUW1. The 3-D effect on underwater acoustic propagation in the offshore area of Taiwan's coast. Chi-Fang Chen, Jang-Jia Lin (Dept. of Naval Architecture and Ocean Eng., Natl. Taiwan Univ., 73 Chou-Shan Rd., Taipei, Taiwan, ROC), Chung-Wu Wang (Naval Oceanograph. and Hydrographic Office, ROC), and Ding Lee (Yale Univ./Naval Undersea Warfare Ctr., New Haven, CT)

Lately 3-D effects on underwater acoustic propagation have frequently been reported. The major causes for the 3-D effects are the variations in azimuth of bottom topography or water column properties. The offshore regions of Taiwan's coast are of the similar nature. There are submarine canyons in the offshore region of southwest Taiwan. The acoustic propagation is studied numerically using FOR3D. The 3-D effect is found to be more serious along the axis of the canyon's axis than cross the canyon. The field test is also conducted in the region; the measurements are well matched with the numerical results calculated by FOR3D, which indicates the actual 3-D effect exists in the real ocean environment. [Work supported by National Science Council of Republic of China.]

1:25

4pUW2. High-resolution, three-dimensional measurements of low-frequency sound propagation in shallow water. George V. Frisk, Kyle M. Becker, Laurence N. Connor, James A. Doutt, and Cynthia J. Sellers (Dept. of Appl. Ocean Phys. and Eng., Woods Hole Oceanogr. Inst., Woods Hole, MA 02543)

An overview is presented of the modal mapping experiment (MOMAX), which was conducted in March 1997 in the vicinity of the East Coast *STRATAFORM* site. Both fixed and moving source configurations were used to transmit several pure tones in the frequency range 50–300 Hz. The magnitudes and phases of these signals were recorded on several freely drifting buoys, each containing a hydrophone, GPS and acoustic navigation, and radio telemetry. High-resolution, three-dimensional measurements of the sound field were made out to ranges of 10 km and illustrate the influence of the laterally varying seabed, measured with a chirp sonar system. The precision navigation also enabled the creation of a two-dimensional, synthetic aperture planar array, parallel to the ocean surface. The pressure field data measured on the array were transformed into the wave number domain, where the lateral variability of the waveguide manifests itself in the spatially evolving spectral content of the modal field. Finally, the phases of the measured signals show remarkable stability and regularity, even in the context of complex, multimodal fields. This behavior can be exploited to make accurate estimates of the relative source/receiver speed from measurements of the time rate-of-change of the phase. [Work supported by ONR.]

1:45

4pUW3. A fully global hydroacoustic monitoring system for the Comprehensive Nuclear-Test-Ban Treaty—Plans and progress. Martin W. Lawrence and Marta Galindo Arranz (Provisional Tech. Secretariat, Comprehensive Nuclear-Test-Ban Treaty Organization, Vienna Intl. Ctr., A-1400 Vienna, Austria, mlawrence@ctbto.un.or.at)

The first ever fully global hydroacoustic monitoring system is being planned and implemented for use in the verification of a new international treaty, the Comprehensive Nuclear-Test-Ban Treaty (CTBT). This system will provide hydroacoustic monitoring of all the world's oceans for 24 h a day, every day of the year, into the indefinite future. This unique resource will utilize two types of station. One type will be based on a hydrophone at the SOFAR axis depth, cabled back to shore. The other will be based on a seismometer on a small island using detection of the T-phase signal. This latter station relies on a signal which has propagated predominantly through the ocean, but has been converted to seismic energy at the margin of the island. Although this network is being installed for monitoring of nuclear explosions, it will also be a unique resource for scientific investigation of various phenomena. Data will be available to all State Signatories to the CTBT.

2:05

4pUW4. Validation of source region energy partition calculations with small-scale explosive experiments. Douglas B. Clarke, Philip E. Harben, Steven L. Hunter, and Donald W. Rock (Lawrence Livermore Natl. Lab., Livermore, CA 94551)

The decrease in signal energy as the location of a nuclear explosion varies from deep in the ocean to above the ocean surface is a concern for the planned ocean monitoring component of the Comprehensive Test Ban Treaty. Small-scale experiments were designed to validate predictions of energy coupling by nuclear explosions in the ''source region,'' the origin of the signals that propagate in the deep underwater sound (SOFAR) channel. The experiments were performed in a biologically dead lake at a scale length of 1/50 (1/125 000 in explosive energy) relative to one kiloton using 6.82-kg charges of Pentolite 50/50. The acoustic energy coupled into the water was monitored at a 60-m range by a hydrophone string with eight piezoelectric sensors spaced from near-surface to a 30-m depth. Useful data were obtained at five burst locations: 5, 2, 0, −2, and −15 m. Results from the experiments and new calculations support the predicted energy partitioning for above-surface explosions with model and experiment peak pressures agreeing within a factor of two over three orders of magnitude variation. [Work performed under the auspices of the U. S. Department of Energy by the Lawrence Livermore National Laboratory under Contract W-7405-ENG-48.]

2:20

4pUW5. Three-dimensional propagation modeling in shallow water. Gregory J. Orris and John S. Perkins (Naval Res. Lab., Washington, DC 20375)

Much interest exists in problems of shallow-water propagation, where three-dimensional spatially dependent waveguide effects are sometimes very large. In these cases the three-dimensional dependence of the acoustic field can be accounted for in a variety of ways other than solving the often intractable fully coupled problem. There exist several pseudo three-dimensional models that are based on the parabolic equation method (e.g., $N\times2$D PE [J. S. Perkins and R. N. Bear, J. Acoust. Soc. Am. **72**, 515–522 (1982)], the adiabatic PE [M. D. Collins, J. Acoust. Soc. Am. **93**, 2269–2278 (1993)], the coupled-mode PE [A. T. Abawi *et al.*, J. Acoust. Soc. Am. **102**, 233–238 (1997)] and the spectral PE [G. J. Orris and M. D. Collins, J. Acoust. Soc. Am. **96**, 3499–3503 (1994)]). Yet, because these models are extensions of efficient two-dimensional models they have explicit and implicit limitations on their applicability. Some of the advantages and limitations of these pseudo three-dimensional models will be demonstrated through their application to data collected during the SWellEx-96 and RDS-1 matched-field experiments.

2:35–2:40 Break

2:40

4pUW6. 4-D modeling of sound propagation in shallow water with anisotropic sediment layers. X. Tang, M. Badiey (Marine Studies, Univ. of Delaware, Newark, DE 19716), and W. L. Siegmann (Rensselaer Polytechnic Inst., Troy, NY 12180)

A highly efficient 4-D PE model is developed to study broadband sound propagation at the Atlantic Generating Station (AGS) site. Extensive core data has shown that the sediment at this AGS site consists of multiple layers which vary in both range and azimuth. Consequently, sound propagation in the water column in this area is significantly affected by the complex sediment structure due to the fact that the layered sediment acts like a bandpass filter in which broadband acoustic energy is selectively broken into narrow-band components propagating in different layers, usually referred as sediment mode trapping. The details of the range, azimuth, frequency, and bandwidth dependences as well as their relationships to the sediment structure of such mode trapping effect is numerically explored by performing the 4-D PE model. The empirical orthogonal function representation method is employed to transfer the geoacoustic parameters of the sediment core data into PE inputs. The broadband travel time results at eight different bearings are compared with the existing experimental data. The importance of the 3-D effects is also examined by comparing the numerical results of azimuthal coupled and uncoupled models.

2:55

4pUW7. Wave number extraction techniques for a three-dimensionally varying shallow-water waveguide: A comparison. Kyle M. Becker and George V. Frisk (Dept. of Appl. Ocean Phys. and Eng., Woods Hole Oceanogr. Inst., Woods Hole, MA 02543, kbecker@whoi.edu)

Theory shows that the wave number spectrum of a propagating normal mode field is a function of position for a complex, shallow-water waveguide whose acoustic properties vary in three spatial dimensions. By describing the spatially varying content of the modal field, a direct measure of the propagation characteristics of the waveguide can be made. Consequently, to obtain greater sensitivity of the spectrum as a function of position, it is of interest to resolve the contributions from individual modes using as short a data record as possible. To this end, several wave number extraction methods have been applied to synthetic data, as well as real data measured using a synthetic aperture planar array created by several freely drifting buoys. Wave number content as a function of position was extracted using Hankel transforms, best linear estimator techniques, and autoregressive algorithms. Results obtained for frequencies in the range 50–300 Hz are compared for resolution and consistency using data records of various lengths. Based on these results, recommendations are made for the selection of an appropriate extraction algorithm. [Work supported by ONR.]

3:10

4pUW8. Range-dependent matched-field source localization and tracking in shallow water on a continental slope. Lieutenant(N) Martin L. Taillefer and N. Ross Chapman (SEOS, Univ. of Victoria, P.O. Box 3055, Victoria, BC V8W 3P6, Canada)

This paper describes matched-field source localization and tracking of a sound source towed over a continental slope off the western coast of Vancouver Island. The data were collected using a multielement vertical line array (MEVA) as part of the PACIFIC SHELF sea trial carried out during September 1993. The MEVA is a 16-element array with sensors evenly spaced at depths between 90 and 315 m. The target was towed along a linear radial track over a steep slope out to ranges of 5.5–6 km from the array, and then along a navigational arc in water depths from 250 m at the array to 750 m. The target emitted three continuous wave tones in the band 45–72 Hz. In order to model the environment as accurately as possible, the replica fields were calculated using an adiabatic normal-mode approximation which accounted for elastic wave propagation in the bottom. In addition, a bathymetric database was compiled from various echosoundings of the area. Using conventional linear MFP based on two-dimensional replica fields to determine ambiguity surfaces for any target radial from the array, the target was tracked in range, depth, and bearing along the entire experimental track, with Bartlett processor values as large as 0.87.

3:25

4pUW9. Experimental investigation of matched-field processing in a wedgelike shallow-water environment. Paul A. Baxley (Space and Naval Warfare Systems Ctr., San Diego, Code D881, 49575 Gate Rd. Rm. 170, San Diego, CA 92152-6435, baxley@spawar.navy.mil)

Underwater acoustic propagation in shallow-water wedgelike environments will experience ''bending'' out of the vertical plane containing the source and receiver. Consequently, the time for the energy to travel from the source to the receiver will be altered, or some paths may miss the receiver altogether. Localization estimates from array processors will be in error if this horizontal multipath propagation is ignored. This phenomenon is investigated experimentally via the analysis of vertical-line-array source-tow data recorded in a steep wedgelike environment near San Clemente Island during the fourth Shallow Water evaluation cell Experiment (SWellEX-4). The source-tows examined were approximately in the cross-slope direction. The objective of this study is (1) to observe the horizontal ''refraction'' effects via an examination of the matched-field localization errors resulting from a neglect of that phenomenon, and (2) to study the feasibility of including modeled horizontal refraction effects into the matched-field processor to enhance localization performance. Geoacoustic parameter inversion was performed at short range so that mismatch in this parameter would not be misinterpreted as a horizontal ''refraction'' effect. A three-dimensional Gaussian beam program was used to generate replica fields, and to perform investigative simulations for the experimental environment. [Work supported by ONR/SPAWARSYSCEN San Diego.]

THURSDAY AFTERNOON, 25 JUNE 1998 GRAND BALLROOM A & B (S), 4:00 TO 5:00 P.M.

Session 4pPL

Plenary Lecture

Sabih I. Hayek, Chair
Department of Engineering Mechanics, Pennsylvania State University, 227 Hammond Building, University Park, Pennsylvania 16802

Chair's Introduction—4:00

Invited Paper

4:05

4pPL1. Trends in modeling of structural-acoustics systems with structural complexity in low- and medium-frequency ranges.
Christian Soize (ONERA, BP 72, 92322 Chatillon Cedex, France, soize@onera.fr)

This paper gives a comprehensive survey of a theoretical approach for predicting the frequency response functions of general complex structural acoustics systems (CSAS) in the low- and medium-frequency ranges. For such a CSAS, the complex structure is made of a master structure coupled with internal substructures presenting structural complexity and corresponding to a large number of secondary subsystems, such as equipment units or internal secondary structures, attached to the master structure. The master structure is a general 3-D dissipative structure with an arbitrary bounded geometry and is made of an anisotropic, inhomogeneous, viscoelastic medium. This complex structure is coupled with an external acoustic fluid via the master structure. The approach is based on the use of fuzzy structure theory, introduced by the author in 1985, including recent developments concerning identification of the fuzzy substructure model parameters. This theory allows the effects of the internal structural complexity on the master structure to be modeled. In addition, the construction of an intrinsic reduced model of the frequency response functions of this CSAS adapted to the medium-frequency range (recently proposed by the author) is presented, which allows simplifications in the calculation of the responses to any deterministic or random excitations.

Meeting of Accredited Standards Committee S3 on Bioacoustics

L. S. Finegold, Chair S3
USAF Armstrong Laboratory, Noise Effects Branch AL/OEBN, 2610 Seventh Street, Wright Patterson Air Force Base, Ohio 43433-7901

R. F. Burkard, Vice Chair S3
Hearing Research Laboratory, State University of New York at Buffalo, 215 Parker Hall, Buffalo, New York 14214

P. D. Schomer, Chair, U. S. Technical Advisory Group (TAG) for ISO/TC 43, Acoustics
U.S. CERL, P.O. Box 9005, Champaign, Illinois 61826-9005

J. Erdreich, Chair, U. S. Technical Advisory Group (TAG) for ISO/TC 108/SC4, Human Exposure to Mechanical Vibration and Shock
Ostergaard Acoustical Associates, 100 Executive Drive, West Orange, New Jersey 07052

H. E. von Gierke, Vice Chair, U. S. Technical Advisory Group (TAG) for ISO/TC 43, Acoustics and ISO/TC 108/SC4, Human Exposure to Mechanical Vibration and Shock
1325 Meadow Lane, Yellow Springs, Ohio 45387

V. Nedzelnitsky, U.S. Technical Advisor (TA) for IEC/TC 29, Electroacoustics
National Institute of Standards and Technology (NIST), Building 233, Room A149, Gaithersburg, Maryland 20899

Accredited Standards Committee S3 on Bioacoustics. The current status of standards under preparation will be discussed. In addition to those topics of interest, including hearing conservation, noise, dosimeters, hearing aids, etc., consideration will be given to new standards which might be needed over the next few years. Open discussion of committee reports is encouraged.

SCOPE OF S3: Standards, specifications, methods of measurement and test, and terminology in the fields of mechanical shock and physiological acoustics, including aspects of general acoustics, shock, and vibration which pertain to biological safety, tolerance, and comfort.

THURSDAY AFTERNOON, 25 JUNE 1998 ST. HELENS ROOM (W), 3:30 P.M.

Meeting of Accredited Standards Committee S1 on Acoustics

J. P. Seiler, Chair S1
U. S. Department of Labor, Cochran Mill Road, P.O. Box 18233, Building 038, Pittsburgh, Pennsylvania 15236

G. S. K. Wong, Vice Chair S1
Institute for National Measurement Standards, (INMS), National Research Council, Ottawa, ON K1A 0R6, Canada

P. D. Schomer, Chair, U. S. Technical Advisory Group (TAG) for ISO/TC 43, Acoustics
U. S. CERL, P.O. Box 9005, Champaign, Illinois 61826-9005

H. E. von Gierke, Vice Chair, U. S. Technical Advisory Group (TAG) for ISO/TC 43, Acoustics
1325 Meadow Lane, Yellow Springs, Ohio 45387

V. Nedzelnitsky, U. S. Technical Advisor (TA) for IEC/TC 29, Electroacoustics
National Institute of Standards and Technology (NIST), Building 233, Room A149, Gaithersburg, Maryland 20899

Accredited Standards Committee S1 on Acoustics. Working group chairs will report on their preparation of standards on methods of measurement and testing, and terminology, in physical acoustics, electroacoustics, sonics, ultrasonics, and underwater sound. Work in progress includes measurement of noise sources, noise dosimeters, integrating sound-level meters, and revision and extension of sound level meter specifications. Open discussion of committee reports is encouraged.

SCOPE OF S1: Standards, specifications, methods of measurement and test and terminology in the field of physical acoustics including architectural acoustics, electroacoustics, sonics and ultrasonics, and underwater sound, but excluding those aspects which pertain to biological safety, tolerance and comfort.