Session 4pAA

Architectural Acoustics: Floor Impact Sound in Buildings

Anthony P. Nash, Cochair
Charles M. Salter Associates, 130 Sutter Street, San Francisco, California 94104

Sho Kimura, Cochair
College of Science and Technology, Nihon University, 1-8 Kanda-Surugadai, Chiyoda-ku, Tokyo, 101 Japan

Chair's Introduction—1:55

Invited Papers

2:00

4pAA1. Development of floor impact sound insulation technology in Japan. Sho Kimura (Dept. of Architecture, Sci. & Technol., Nihon Univ., 1-8 Kanda Surugadai Chiyoda-ku, Tokyo, 101 Japan)

With the formulation of sound insulation performance standards by the Architectural Institute of Japan in 1979, the sound insulation ratings for floor impact sound level determined by means of the heavy and soft impact source stipulated by JIS have been used for performance orders and labeling for buildings in general. Furthermore, prediction calculation methods for sound insulation ratings based on the impedance method have also been developed. This has led to an increase in the thickness of standard floor slabs in concrete apartment houses in Japan from 110–120 mm at that time to 150–160 mm and, recently, there are many floor slabs that are as thick as 200 mm, thereby greatly contributing to an improvement in floor impact sound insulation performance. Meanwhile, there has also been progress in research dealing with the improvement of the insulation performance of heavy floor impact sound in wooden buildings. Unitizing highly rigid floor panels and beams and increasing the flexural rigidity of the overall floor cross section, while designing sound-insulated ceilings of the lower storey independent of the floor structure and the independent walls of the room below, have made it possible to achieve the performance of a concrete slab about 150 mm thick.

2:15

4pAA2. Assessment of low-frequency footfall noise in wood-frame multifamily construction. Warren E. Blazier, Jr. (Warren Blazier Assoc., Inc., 2064 Union St., San Francisco, CA 94123)

The impact insulation class (IIC) rating of floor/ceiling constructions fails to identify low-frequency footfall noise problems common in contemporary wood-frame multifamily construction. Because the IIC procedure ignores the impact noise spectrum below 100 Hz, the rating is not influenced by the low-frequency impact spectra produced by a typical live-walker, which usually peak in the region of 15 to 30 Hz and are associated with the "thuds" and "thumps" commonly observed. Data comparing the shape of the typical low-frequency impact noise spectrum produced by a live-walker and that of the standard ISO tapping machine indicate that the tapping machine might potentially be used as the basis for obtaining *low-frequency* impact noise ratings [W. E. Blazier, Jr. and R. B. DuPree, J. Acoust. Soc. Am. **96**, 1521–1532 (1994)]. However, a comparison made between the two sources in the range above 100 Hz shows significant differences in the spectra produced by the tapping machine and a live-walker. This suggests that a more meaningful IIC rating of mid- to high-frequency impact noise might be obtained by cushioning the tips of the hammers in the standard ISO tapping machine, in order to better align the impact spectrum with that of a live-walker.

2:30

4pAA3. Development of a heavy and soft impact source for the assessment of floor impact sound insulation. Hideki Tachibana (Inst. of Industrial Sci., Univ. of Tokyo, Roppongi, Minatoku, Tokyo, 106 Japan) and Hiroshi Tanaka (Nippon Sheet Glass Co., Ltd., Koyodai 4-5, Ryugasakishi, 301 Japan)

In order to examine the impact sound insulation performance of building floors against heavy and soft impacts such as those of children jumping and running, a method whereby a heavy and impact source (an automobile tire) is dropped is specified in JIS A 1418, in addition to the standard method of using the ISO tapping machine. In order to improve the latter method, a research project has been carried out in the Architectural Institute of Japan. After various trial experiments, a new heavy and soft impact source (a rubber ball, weight: 2.5 kg; size: 183 mm in diameter; impact time: 20 ms) has been developed. In this paper, the basic impact characteristics of the source and the experimental results obtained in field measurements are introduced in comparison to the existing impact sources (automobile tire and the tapping machine).

2:45

4pAA4. Stimulus-response measurement of floor impact sound. Anthony Nash (Charles M. Salter Assoc., 130 Sutter St., San Francisco, CA 94104)

In the area of floor impact testing, a number of excitation techniques have been used for evaluating the degree of sound transmission. The ultimate goal of the excitation is to replicate the force and time duration of a human footfall so one can reliably rank the acoustical quality of the floor/ceiling system. The most common excitation methods involve dropping some object from a known height so the kinetic energy delivered to the floor is consistent. The limitation of the energy method is that the force spectrum imparted to the floor can vary if the floor finish has unstable or nonlinear characteristics (e.g., a thick carpet). In order to control this parameter, research laboratories have occasionally used a force sensor to help calibrate the impact forces from dropped objects. This paper will discuss the concept of using a force sensor as one component of a stimulus–response impact sound measurement *in the field*. A unique 13-kg force transducer ''platform'' was constructed in order to perform the field experiments. The platform is capable of measuring large dynamic forces at frequencies up to 100 Hz. Some examples of two-channel field data (force versus sound pressure) will be presented to demonstrate the technique.

3:00

4pAA5. Proposal for a standard heavy impact source for the measurement of floor impact sound insulation performance. Katsuo Inoue and Sho Kimura (Dept. of Architecture, Sci. & Technol., Nihon Univ., 1-8 Kanda Surugadai Chiyoda-ku, Tokyo, 101 Japan)

The floor impact sound generated in multifamily dwellings frequently has a peak in the low-frequency band (20–30 Hz). Because of that, in order to assess the floor impact sound insulation performance in real-life situations, it is necessary to measure floor impact sound using a standard impact source, with an impact force characteristic corresponding to actual impact sources. However, in the current ISO standards, a ''tapping machine'' that is light and stiff with a weak impact force is the only one that has been approved as a standard floor impact source, which cannot be considered adequate as a method of measurement in solving problems of floor impact sound. Actual case examples of the measurement of impact force performance of floor impact sources generated in actual housing units are presented in this paper, together with a verification of the usefulness of the heavy floor impact source stipulated by current JIS standards in Japan and a proposal for the introduction of the same impact force characteristics as ISO standards.

3:15

4pAA6. Some consequences of including low frequencies in the evaluation of floor impact sound. Jens H. Rindel and Birgit Rasmussen (Dept. of Acoust. Technol., Tech. Univ. of Denmark, Bldg. 352, DK-2800 Lyngby, Denmark)

A method for including frequencies down to 50 Hz in the evaluation of floor impact sound has become available with the new version of ISO 717-2. In addition to the single number quantity for rating the impact sound insulation, a new spectrum adaptation term has been defined. The method has been studied by the Acoustics Group of NKB (Nordic Committee on Building Regulations). The new method has been applied to a large number of recent measuring results from the Nordic countries. It was found that the spectrum adaptation term for the extended frequency range depends on the type of floor construction, and light floor constructions are evaluated less favorably than heavy constructions. Comparison with data from a 14-year-old Swedish survey suggests that the extended frequency range leads to a higher correlation with subjective evaluation of impact noise. The consequences of applying the extended frequency range in future building regulations or in a system for sound classification of dwellings have been considered. However, there are several problems to be solved, among which are a lack of available data for floor constructions at low frequencies, an increased measurement uncertainty, and the fact that some floor constructions are evaluated too favorably by the new method.

3:30

4pAA7. Floor research project at NRC Canada: An update on progress—Measurements with experimental impact devices. A. C. C. Warnock (NRC Canada, M27 Montreal Rd., Ottawa, ON K1A 0R6, Canada)

A major thrhust in this floor project is the collection of low-frequency impact information, below 100 Hz, using special impactors: a tire-dropping machine, two experimental rubber balls supplied by H. Tachibana, and a walker. The frequency range for the standard ISO tapping machine is also extended at 50 Hz. A preliminary analysis of the data collected so far in the project will be presented. Factors that were expected to change the shape of the impact spectrum drastically at low frequencies, so far, have had little or no effect. Changing joist depths from 185 to 285 mm had negligible effect on low-frequency transmission. Changing I-beam depth from 235 to 460 mm did have a significant effect. The best wood joist floor tested to date is still worse than a 150-mm concrete slab. So far in the project, the most effective remedy for excessive low-frequency impact sound is mass.

Contributed Papers

3:45

4pAA8. Floor research project at NRC Canada: An update on progress—Airborne sound and standard tapping machine measurements. A. C. C. Warnock (NRC Canada, M27 Montreal Rd., Ottawa, ON K1A 0R6, Canada)

The extensive research project on sound transmission through floors is still in progress. This paper will present information on standard measurements made according to ASTM E90, ASTM E492, and ISO 140 parts III and VI. Since the last report in 1995, the effects of changing joist length, joist spacing and depth, wood I-beam type and depth, and the effect of adding concrete as a topping on the floor have been studied. Different types of subfloors and different types of gypsum board have been used in several combinations. For a given basic structure, the mass of the panels has the most effect on the single number ratings, STC and IIC, which are determined mainly by the sound transmission loss values around 125 Hz. Many structural changes have little effect around 125 Hz so STC values do not change much, although there are significant changes at other frequencies.

4:00

4pAA9. Review of research reports on floor impact sound in Japan. Hirokazu Fukushima (Environment and Fire Dept., Bldg. Res. Inst., Ministry of Construction Japan, 1 Tatehara Tsukuba-si Ibaraki-pref. 305) and Masahito Yasuoka (Univ. of Tokyo, 1-3 Kagurazaka Shinjuku-ku Tokyo, 162 Japan)

The authors have reviewed technical papers on floor impact sound reported in journals and proceedings published by the Architectual Institute of Japan, the Acoutical Society of Japan, and the Institute of Noise Control Engineering Japan. Research on this subject started in the 1960s in Japan, with considerable delay from the development in this area in North America and Europe. However, in the last decade, approximately 25 papers have appeared on this subject every year, and the total number of the relevant papers in Japan has been found to exceed 400. The number of research works on floor impact sound in Japan is characterized by the use of a soft and heavy impact source (a tire for compact cars). As is widely known, this source is the standard impact source unique to Japan. However, approximately half the recently reported papers deal with noise produced by the tapping machine. This change seems to reflect the Japanese consumers' general choice of floor lining changing from carpets to wooden floor coverings. This report will discuss the Japanese trend in research on floor impact sound, not only through the number of papers but also through the introduction of relevant social background in Japan.

4:15

4pAA10. Prediction of heavy weight impact sound of large-spanned slabs. Takashi Koga and Masanori Tano (Kajima Tech. Res. Inst., Kajima Co. Ltd., Chofu, Tokyo, 182 Japan)

Adaptation of the large-spanned RC slabs, whose area corresponds to an apartment unit, is increasing in Japan. The authors measured heavy-weight impact sound and impedance of such large slabs at over 30 sites. The structural methods of slabs are: unbond, half-precast concrete, void, and the conventional framework method. There were no distinctive differences among structural methods except in case for void slabs. The impact sound level was determined by the thickness of the slab rather than the area of the slab, even if the area of the slab was over 70 m^2. In the case of small slabs, where the first eigenfrequency of the slab vibration is around the 63-Hz octave band, there is a tendency of a high-impact sound level. On the other hand, the lower first eigenfrequency may be a benefit for the impact sound level. The stiffness level of the beam seem to depend on frequency.

4:30

4pAA11. Impact force characteristics by footsteps and the vibration characteristics in housing floors. Hideo Watanabe (Tech. Res. Inst., Toda Corp., 315 Kaname, Tsukuba City, Ibaraki, Japan), Sho Kimura, and Katsuo Inoue (Nihon Univ., 1-8-14, Kanda, Surugadai, Chiyodaku, Tokyo)

Various experimental researches were conducted dealing with physical characteristics and sensation in order to be able to express quantitatively appropriate hardness in terms of comfortability in residential floors. From among these, the results of analyses of impact force time characteristics of a sole by footsteps and the production of a prototype impact source involving the impact force of the heel, which is a representative impact force generated by footsteps are indicated in this paper. Then, using the impact source to measure the amount of dynamic displacement in wooden floors of old construction and in the direct-pasted wood flooring commonly used in recent multifamily housing, the dynamic spring coefficient of traditional wood-construction floor structure, which had never before been clarified, was quantitatively determined.

4:45

4pAA12. 30 years of resilient furring channels. David A. Harris (Acoust. Consultant, Port Ludlow, WA 98365)

This paper reviews the development, testing, and present day use of the venerable resilient furring channel, one of the most successful acoustical products invented. Billions of feet have been produced and ''it's still going.'' Wall and floor/ceiling systems with resilient furring channels were, and still are, among the most practical techniques for improving the airborne and impact sound transmission characteristics of wood frame structures. Performance data on typical assemblies will be presented along with tips to maximize field performance.

5:00

4pAA13. Quest for quiet floor and wall systems. David A. Harris (Acoust. Consultant, Port Ludlow, WA 98365)

Lightweight floors over wood and metal joists are notoriously noisy. Both impact and airborne noise are intrusive. A unique technique has been developed that allows the use of hard surface materials such as vinyl, hardwood, or tile without compromising sound isolation attributes or economics. Recent subjective and objective testing of floor and wall systems that utilize constrained layer damping technology demonstrate significant promise. Lightweight wood wall and floor/ceiling assemblies utilizing these principles exhibit a perception of quiet solidarity typically found in expensive concrete systems. This paper presents a series of tests comparing conventional plywood and gypsumboard systems with similar assemblies containing constrained layer damped subfloor and wall materials.

5:15

4pAA14. Field impact noise test data on a concrete floor structure. Roy L. Richards and Ioana Park (Towne, Richards and Chaudiere, Inc., 105 NE 56th St., Seattle, WA 98105)

A total of 83 field impact insulation tests were performed using a Bruel & Kjaer calibrated tapping machine. The tests generally followed ASTM procedures. All measurements were conducted in the same building, using two adjacent receiving rooms constructed for the purpose of the tests. The floor-ceiling constructions tested consisted of a 7-in. concrete base slab, over which were installed a variety of resilient underlayments and finished floors. Tests were conducted with and without a resiliently suspended gypsum board ceiling. Results were extended at low frequencies to the 50-Hz third-octave band. However, measurements in the 50 to 80-Hz frequency range were often limited by background noise. The field impact insulation class (FIIC) results showed one of the resilient underlayments to be clearly superior to others under a ceramic-tile surface, but not as clearly superior under wood surfaces. The resiliently suspended ceiling improved FIIC ratings by 6 to 15 points, depending on the materials laid on the concrete slab. No tests were conducted on carpeted surfaces.

Session 4pBVa

Bioresponse to Vibration and to Ultrasound: Tactile Communication

Ronald T. Verrillo, Cochair
Institute for Sensory Research, Syracuse University, Merrill Lane, Syracuse, New York 13244-5290

Tohru Ifukube, Cochair
Institute for Electronic Sciences, Hokkaido University, Sapporo, Hokkaido, 060 Japan

Chair's Introduction—2:00

Invited Papers

2:05

4pBVa1. Haptic speech perception: Are the possible pay-offs worth the effort? Lynne E. Bernstein (Spoken Lang. Processes Lab., House Ear Inst., 2100 W. Third St., Los Angeles, CA 90057)

Episodically throughout the 20th century, practical and scientific inquiries into speech perception by touch have been undertaken. While some extraordinary examples of communication by the natural (uninstrumented) method of Tadoma have suggested that touch has great potential for speech perception, artificial vibrotactile devices have produced relatively modest results. Why at this time, in the light of success (at times verging on spectacular) of auditory implants for people with profound hearing losses, should additional resources be invested in development and testing or sensory aids for speech perception via touch? In answer to this question, this presentation will summarize research on speech perception involving touch, outline important practical and clinical justifications for continued interest, and discuss some theoretical issues in speech perception to which studies involving touch can contribute. [Work supported by NIH DC00695 and DC01577.]

2:20

4pBVa2. New discoveries in taction and their implications for tactile communication. Stanley J. Bolanowski, Ronald T. Verrillo, George A. Gescheider, James C. Makous, Bradley M. Pietras, Jason C. Cohen, and Christine M. Checkosky (Inst. for Sensory Res. and Dept. of Bioengineering and Neurosci., Syracuse Univ., Syracuse, NY 13244)

The implementation of tactile aids for the deaf and blind has progressed largely through the development of new technologies and strategies adapted to adequately and efficiently activate previously understood tactile mechanisms and processes. In the past decade, however, important new discoveries have been made relating to the basic mechanisms of taction such as the effects of aging and disease, submodality interactions, skin mechanics, and learning and cortical plasticity. These advances in our knowledge will impact heavily on future developments and applications of tactile aids. Several of these new discoveries will be described in the context of basic tactile mechanisms and in their implications for significant improvements in the effectiveness of aids for tactile communication. [Work supported by NIH.]

2:35

4pBVa3. Fine surface-texture discrimination ability depends on the number of mechanoreceptors participating in the discrimination task. Tetsu Miyaoka, Masahiro Ohka (Dept. of Comput. Sci., Shizuoka Inst. of Sci. and Technol., 2200-2 Toyosawa, Fukuroi, 437 Japan), Takuya Kawamura, and Yasunaga Mitsuya (Nagoya Univ.)

The purpose of this study was to investigate the relation between fine surface-texture discrimination ability in tactile sensation and the number of mechanoreceptors participating in the discrimination task. Two experiments were performed. In experiment 1 the spatial-summation effect was measured at the fingertips using aluminum oxide abrasive papers as stimuli, with grit values between 600 and 8000. Difference thresholds could not be determined when the area of the papers was 25 mm^2. Thresholds were between 11.3 and 13.5 μm at 100 mm^2, and they were between 3.14 and 5.58 μm at 400 mm^2. In experiment 2 the difference thresholds of the texture-discrimination tasks were measured at the fingertips and the thenar eminence using large-sized (2500 mm^2) abrasive papers with grit values between 400 and 8000. The difference thresholds were smaller at the thenar eminence than at the fingertips because the stimulated area at the thenar eminence was eight times larger than the area at the fingertips. From the results of experiments 1 and 2, it was concluded that the larger the number of mechanoreceptors which participate in the discrimination tasks, the smaller the difference thresholds. [Work supported by the Ministry of Education, Science and Culture, No. 07610093 and Mikiya Science and Technology Foundation.]

4p THU. PM

2:50

4pBVa4. Stimulus factors in perceptual confusability of tactile speech stimuli. Janet M. Weisenberger (Dept. of Speech and Hearing Sci., Ohio State Univ., Columbus, OH 43210)

One possible limiting factor in performance by users of tactile aids for speech perception is the high confusability of tactile speech stimuli. This paper summarizes results of several experiments designed to identify the physical characteristics of tactile speech signals that make two stimuli seem perceptually confusable. First, sets of nonsense syllables were recorded onto digital audiotape and presented to subjects in an identification task. These same stimuli were input to a computer simulation of a multichannel vibrotactile speech aid, the Queen's vocoder, showing the level of activity in each of the 16 vocoder channels over the duration of the stimulus. Perceptual confusion matrices obtained from the identification task were compared to matrices of temporal and spatial coincidences of physical activity in corresponding channels for each stimulus. In further experiments, tactile reception of individual nonsense syllables was compared to degraded auditory perception of the same stimuli. Several different methods of auditory stimulus degradation were employed, including reducing the number of vocoder channels and temporal distortion (smearing). Results are discussed in terms of insights into the processing of complex, spatially distributed vibrotactile stimuli, as well as implications for modifications of tactile aid processors and displays to improve stimulus identifiability. [Work supported by NIH.]

3:05

4pBVa5. The effect of frequency upon tactile apparent movement. Tenji Wake (School of Management, Sci. Univ. of Tokyo, Tokyo, Japan) and Hiromi Wake (Kanagawa Univ., 3-27 Rokkakubashi Kanagawa-ku, Yokohama-shi Kanagawa-Ken 221, Japan)

Tactile apparent movement was observed under the conditions of variation in stimulus duration, frequency, and separation between two stimuli. First, optimal stimulus onset asynchrony (SOA) was determined as a function of stimulus duration using stimuli with the same frequency. Second, optimal SOA was estimated as a function of frequency, by using two stimuli with different frequencies. The optimal SOA for tactile apparent movement increased with the increases of stimulus duration. When two vibrators activate with the same frequency, the optimal SOA depends on frequency. The optimal SOA of the low-frequency stimulus (10 Hz) is longer than that of the high-frequency stimulus (250 Hz). If the first stimulus is 10 Hz and the second stimulus is 200 Hz, the optimal SOA is shorter than two stimuli with same frequency (10 Hz). The first stimulus was kept constant at 10 Hz and the second stimulus was varied in frequency. As the frequency-difference increases, the optimal SOA is decreased, but at the second stimulus of 80 Hz, the optimal SOA is similar to that of two stimuli of 200 Hz. These findings suggest that there are interactions between two channels. [Work supported by the Ministry of Education, Science and Culture, No. 62810008.]

3:20

4pBVa6. Is it possible to obtain the subjective-contour effect in touch? Hiromi Wake (Dept. of Psych., Kanagawa Univ., 3-27 Rokkakubashi Kanagawa-ku, Yokohama-shi Kanagawa-Ken, 221 Japan) and Tenji Wake (Sci. Univ. of Tokyo, Tokyo, Japan)

The so-called subjective-contour effect was examined in touch and compared with vision. The first experiment was designed to establish a criterion of the subjective pattern by modal completion. In experiment 2, free verbal reports were gathered from naive subjects. Verbal reports and rating scores for three characteristics, subjective pattern, depth displacement, and brightness transformation, were gathered from the well-trained subjects in experiment 3. Results were as follows. (1) The effect was strong in vision especially for the pattern. Subjective pattern was reported in touch, but, in most other cases, they understood the pattern to be just a triangle or square as a result of bridging, without perceiving the contiguous surface. (2) Subjective depth was not observed so much even in vision by the naive subjects, although it was clear to the well-trained subjects. When depth displacement was reported, the subjective surface was perceived as more concave than the surface of paper. In conclusion, although the subjective-contour effect was confirmed in touch, it might be somewhat different from that of vision. It is still not clear if the perceived effect in touch is the result of the modal completion as it is in vision. [Work supported by the Ministry of Education, Science and Culture, No. 0310063.]

Contributed Paper

3:35

4pBVa7. A proposal of a new display method for the tactile voice coder: Usage of the swept convex pattern. Chikamune Wada, Shuichi Ino, and Tohru Ifukube (Res. Inst. for Electron. Sci., Hokkaido Univ., Sapporo, 060 Japan)

A fingertip tactile vocoder for the deaf, has been designed, which has a vibrator matrix with 4×16 pins. A sweeping display method was proposed so that the stimuli were moved from right to left on a fingertip surface like electric news tape. From the experimental results, it was proven that an identification rate of the seven Japanese monosyllabic voices increased and reached 55% at a sweep speed of 10 cm/s. Furthermore, in order to improve identification rate, the usage of the convex pattern like Braille has been proposed, in addition to the sweeping display method. It was found that the identification rate of the seven Japanese monosyllabic voices increased by combining the convex pattern with the vibratory pattern, and the increment was about 20%, even though training period was 30 min. From the above results, it was concluded that the usage of the swept convex pattern besides the swept vibratory pattern was effective to increase the identification rate of speech. [Work supported by the Ministry of Education, Science and Culture, Grant-in-Aid for Scientific Research (A)(2) No. 06402067.]

Session 4pBVb

Bioresponse to Vibration and to Ultrasound: Effects of Vibration on Man and Animals

Ryusuke Hosoda, Cochair
Department of Marine System Engineering, Osaka Prefecture University, 1-1 Gakuenmati, Sakai, Osaka, 590 Japan

Gunnar Rasmussen, Cochair
GRAS, Sound and Vibration Applications, Skelstedet 10B, Vedbaek 2950, Denmark

Chair's Introduction—4:00

Invited Papers

4:05

4pBVb1. Measurement of physiological and psychological responses during exposure to low-frequency whole-body oscillations. Naonosuke Takarada, Masashi Takesada, Ryusuke Hosoda, and Masakazu Arima (Osaka Prefecture Univ., Dept. of Marine System Eng., Osaka, Japan)

Human responses to low-frequency whole-body oscillations were measured using a ship-motion simulator. An electroencephalogram (EEG), an electrocardiogram (ECG), perspiration, face-surface temperature, facial expressions, and the respiration of subjects were measured as physiological effects, and ride quality and motion sickness symptoms as the psychological effects. Before and after exposure to the oscillations, subjects were asked to reply to questionnaire surveys prepared to measure the psychological changes, to take blood samples, and to have blood pressure and sense of balance inspections. Measured results of physiological changes such as EEG, ECG, etc. have been analyzed by means of spectral analysis, wavelet analysis function analysis, and so on. The results of the questionnaire survey of the semantic differential method (SD method) were analyzed by introducing fuzzy integral and fuzzy measures as well as the traditional statistical method. Correlation analyses using all the measured results were carried out. From the results of analyses, it has been made clear that (1) subjects are classified into several types, and (2) psychological effects are important to evaluate the motion sickness incidence and ride quality, for example.

4:20

4pBVb2. Stability of man gazing at a moving picture. Hajime Takada (Dept. of Mech. Eng. and Material Sci., Yokohama Natl. Univ., 156 Tokiwadai, Hodogaya, Yokohama, 240 Japan) and Motoki Ohba (Komatsu Co. Ltd.)

This paper is concerned with movements and stability of a man gazing at a moving picture in a darkroom. These experiments were studied to clarify the relation between visual information and postures. A worker may indirectly be exposed to vibration when the objects around him swing, for example by an earthquake. This is not only a problem for workers but also, when he operates a large plant, it is a problem for the system if he operates it erroneously. The bar picture was made with a computer and swings right and left in front of a subject in the first case, and, in the second case, the bar pictures start suddenly to the right side or to the left side and, after a constant time, they suddenly stop. As a result, one may stand more unstably under the following conditions: when the bars suddenly move or stop and when the equilibrium line of the bar vibration deviates right or left than when the line exits in the constant position. A man can stand more stably after a few trials than at the first or second trial. A man can swing his center of gravity when the bar swings in front of him. The picture is different from that which can make him unstable.

Contributed Paper

4:35

4pBVb3. Whole body vibration on fast moving trains. Gunnar Rasmussen (G. R. A. S. Sound and Vib. Applications, Skelstedet 10B, 2950 Vedbæk, Denmark)

A series of measurements has been carried out on trains in order to determine the comfort level for crew and passengers. Different trains and varying rail sections were evaluated at train speeds of 120 to 180 km/h. The main problem area is the comfort of standing persons and especially the crew. The measurements were carried out sampling the time varying signal from an x-y-z transducer mounted directly on the floor. The measurements were carried out from 0.1 to 100 Hz and stored directly onto a PC hard disk for later processing. The processing was carried out according to ISO 2631 and ISO 8041. Having the raw data signal available enables one to process the data using different time constants and weighting curves as well as root-mean-square and fourth-power values. A linear integrated value of the entire duration of a trip has little correlation with the perceived comfort. A subjective evaluation using a group of standing people judging their comfort level, on a 5-stage scale from very comfortable to very uncomfortable, shows good agreement with the running rms acceleration level using a 10-s time constant. The lateral exposure level in the Y direction (90° to direction of travel) for a comfortable ride was 0.4 m/s^2.

Session 4pEA

Engineering Acoustics: Acoustical Measurements and Instruments

James M. Powers, Cochair
Naval Undersea Warfare Center, Code 2131, New London, Connecticut 06320

Takeshi Fujimori, Cochair
Electrotechnical Laboratory, Umezono 1-1-4, Tsukuba, Ibaraki, 305 Japan

Contributed Papers

2:00

4pEA1. Numerical and experimental study of radiated fields of high-power ultrasonic transducers for industrial applications in gases. C. Campos-Pozuelo (Inst. de Acústica, CSIC, Serrano, 144, 28006 Madrid, Spain), A. Lavie, B. Dubus (ISEN, Cedex, France), G. Rodriguez-Corral, and J. A. Gallego-Juárez (Inst. de Acústica, CSIC, 28006 Madrid, Spain)

Flexural axisymmetric vibrating plates with an extensive surface and stepped profile have been shown to be very useful high-power ultrasonic radiators. In fact, the stepped profile provides a highly directional or focused radiation which makes them highly efficient for gases and interphase applications. In this work we present a numerical–experimental study of the displacement distribution and the radiated field of various transducers (directional and focusing transducers working at different frequencies). The numerical model is based on a mixed finite element-boundary element method. The method basically consists of calculating the distribution of displacements of the transducers by the finite element method by using the code ATILA while the corresponding radiated field is calculated by the boundary element method by means of the code EQI. The influence of the loading medium on the displacement distribution is also calculated by the boundary element method. In this way the near and far field of the transducers is calculated as well as their directivity pattern and vibration distribution. The results obtained are compared with experimental data.

2:15

4pEA2. Development of bubble breaking equipment by aerial ultrasonic waves. Hisao Iwata, Sueshige Nakamura, Sadahiro Abe, and Yasushi Ito (Mitsubishi Heavy Industries, Ltd., Nagoya R&D Ctr., Iwatsuka-cho, Nakamura-ku, Nagoya, 453 Japan)

In a filling machine for carbonated drinks, many bubbles are generated, causing the vessels to overflow. As a result, only a small volume of drink is left in the vessel, and the vessel is dirtied. For breaking these bubbles, some bubble breaking methods were investigated experimentally. Finally, it was discovered that the method using aerial ultrasonic waves was the best of all for the filling machines. A bubble breaking machine utilizing aerial ultrasonic waves has been developed. It consists of an ultrasonic vibrator, a metallic horn, a rectangular plate, partition plates, and a reflection plate. The rectangular plate is designed to vibrate in the stripes mode, which is the most efficient radiation mode. The partition plates are set perpendicular to the node lines of the stripes mode, so many pistons are produced by the partition plates. The reflection plate concentrates sound radiation from one piston at one particular point. The equipment was applied to a filling machine of carbonated drinks and the desired results were obtained.

2:30

4pEA3. A new method for measuring the bulk modulus of compliant acoustic materials. François M. Guillot and Jacek Jarzynski (George W. Woodruff School of Mech. Eng., Georgia Inst. of Technol., Atlanta, GA 30332-0405)

In this method the sample is compressed, and a dual-beam laser Doppler vibrometer is used to measure the resulting strain. The sample is suspended in air by thin threads. A transient acoustic signal is used to compress the sample, where the wavelength of the acoustic signal in air is significantly greater than the sample dimensions. A rigid reflector is placed behind the sample to double the acoustic pressure. Data are presented for a rubber–air composite material. Extension of the above method to measurements as a function of temperature and pressure is described. The frequency range for this method is ∼100–3000 Hz.

2:45

4pEA4. Vibroviscometer using a triangular bimorph transducer. Tsutomu Kobayashi, Hidekazu Tai (Dept. of Elec. Eng., College of Eng., Nihon Univ., Nakagawara 1, Tokusada, Tamura-mati, Koriyama, 963 Japan), and Sadayuki Ueha (Tokyo Inst. of Technol., Yokohama, 226 Japan)

A new vibroviscometer is proposed, and it was fabricated in order to prove its feasibility. The viscometer consists of a triangular bimorph transducer and a circular thin plate. The bimorph transducer consists of two piezoelectric plates and a metal shim plate, and the shim plate is sandwiched in between the piezoelectric plates so that two pairs of piezoelectric–metal-plate (unimorph) transducers can be used separately as driving and detecting transducers. The circular plate is adhered on the acute tip of the triangular transducer at a right angle to the transducer surface. As the measurement procedure, only the circular plate driven by one unimorph transducer is immersed in the liquid and the output of the other unimorph transducer is monitored to be compared with the driving signal. It was found that the difference in the electrical phase between the driving and the monitored signals appears to be in proportion to the viscosity of liquids and that the difference remarkably appears at the range of the antiresonance frequency between the first and the second resonance frequency of the triangular bimorph. The reason for this is that the viscosity of the liquid exerts a retarding force on the plate. As for the results of the experiments regarding the frequency characteristics, the temperature characteristics, and the measurement of the viscosity for many liquids, it was proved that this phase difference method using the triangular bimorph can be used successfully for precise measurements of the viscosity.

3:00

4pEA5. Filtration technique enhanced by electro-acoustic methods. Timo Tuori (VTT Energy, P.O. Box 1603, FIN-40101 Jyväskylä, Finland), Eliisa Järvelä, Hannu Sekki, Hannu Mursunen, Tuulia Väärämäki, and Juha Heikkinen

Results from an experimental study of electric and/or ultrasonic field assisted filtration are presented. Both electric and ultrasonic fields can reduce fouling of the filtration medium and have significant influence on filtration capacity. The extent of filtration improvement is affected mostly by particle size, surface charge, acoustic frequency, and field strengths. Theoretical examinations of the use of electric field and/or ultrasonic field to enhance filtration efficiency were laid out. Some aspects regarding orthokinetic interaction in acoustic agglomeration were considered. Energy consumptions of the filtrations of different suspensions used in experiments were also determined. For pyrite suspension the best results with ultrasound were obtained when filtration was done as follows: 15-s negative pressure filtration, 15-s negative pressure filtration with 22-kHz ultrasonic burst (0.5 s) and 20-s drying with negative pressure. In this case the moisture content of cake was 7% while in the reference cake it was 14%. The cake capacity was fourfold greater than in the reference case. The best results for phosphoric acid were obtained when combined use was made of the ultrasound and the electric field. In this case the filtration capacity increased fifteenfold while the use of ultrasound on its own increased the filtration capacity tenfold.

3:15

4pEA6. The removal of small particles using high-frequency ultrasonics. Ahmed A. Busnaina (Microcontamination Res. Lab., Clarkson Univ., Potsdam, NY 13699-5729)

High-frequency (near 1 MHz) liquid-based ultrasonic cleaning (known as megasonic cleaning) is widely used in industry for the removal of particulate contamination. Megasonic cleaning is an effective but poorly understood particle removal technology. It is found that acoustic streaming, microstreaming, and stable cavitation aid in particle removal. Results show that removal efficiencies near ten power is one of the most important parameters in megasonic cleaning. The megasonic input power has a greater effect on the removal efficiency than does temperature. An unexpected decrease in cleaning efficiency at high powers is reported, and attributed to energy attenuation due to increased bubble activity. This is consistent with previous results generated using different megasonic equipment. The results also show that when the optimum reported power is used, the removal efficiency will be high over a wide range of bath temperatures. [Work supported, in part, by the New York State Center for Advanced Materials Processing, Clarkson University.]

3:30

4pEA7. Piezoelectric copolymer array for nondestructive inspection. Hajime Yuasa and Kuniyuki Masazumi (Akishima Labs., Mitsui Zosen, Inc., 1-50, Tsutsujigaoka 1-chome Akishima, Tokyo, 196 Japan)

Ultrasonic sensors which use copolymer membranes, just like other single-crystal elements and ceramic elements, are able to transmit and receive with the same element. However, for meeting the required specifications to identify welding defects around 0.1 mm, there were many difficulties and problems in manufacturing an integrated type of sensor with functions for both transmitting and receiving, as well as from a costwise consideration. Consequently, to overcome these problems, the following new concepts were introduced for quick implementation. As far as the thin copolymer membrane is concerned, the reception part and the transmission part were not separated but integrated, and an independent electrode was installed for each function. The beam-receiving surface of the reception part of the membrane was made so as to form an overall earth electrode, while on the opposite surface, a number of comblike electrodes were installed in parallel. Furthermore, because the electrical impedance of the piezoelectric copolymer sensor is low, careful attention should be paid to matching between the cable and the electrical circuit. With this device, the installation of a preamplifier adjacent to the receiving element solved the problem. The developed ultrasonic sensor has a 1-channel transmitter with a frequency of 20 MHz and high-density receivers with 56 channels on a 20-mm-long membrane.

3:45

4pEA8. Laser ultrasonic evaluation of solder connection in integrated chips. Jerry Rosson Smith, Jr. (Dept. Mech. Eng., Georgia Inst. of Technol., Atlanta, GA 30332)

Currently, no on-line techniques exist for the inspection of solder connections in surface mount technology (SMT) chips or in flip chips. This research investigated the use of laser-induced ultrasound in test chip samples to determine solder joint integrity. The ultrasonic vibrations were detected interferometrically and analyzed to determine chip mounting quality. Analysis was conducted in both the time and frequency domains. A controlled sample of flip chips, several (unidentified) of which were processed in a manner that tended to create poor connections, were tested and the vibrational signatures were compared. Comparison of the vibrational signatures showed a single flip chip with an abnormal signature. Detailed visual inspection demonstrated that the suspect chip was indeed poorly connected. Tests involving SMT consisted of obtaining the signatures of chips with high-quality connections. These connections were systematically damaged to simulate poor joints. Signatures were then taken from the newly damaged chips and compared with the undamaged signatures. Physical models and interpretations concerning experimental results have been developed. Future phases of research are outlined, including other promising results and capabilities. [Work supported by PRC.]

4:00

4pEA9. Numerical simulation of a combined acoustic preconditioning—Electrostatic precipitation (APEP) system and comparison with experimental data. T. L. Hoffmann, E. Riera-Franco de Sarabia, and J. A. Gallego-Juárez (Inst. de Acúst. (CSIC), Serrano 144, 28006 Madrid, Spain)

Based on a recently developed computer code (PAIS—particle agglomeration and interaction simulation), numerical simulations are presented which model the process of acoustic agglomeration using two different agglomeration kernels: the classical orthokinetic and the new acoustic drag interaction kernel. The results of the simulations are contrasted with experimental data. Acoustic particle agglomeration is an effect that occurs when an aerosol is sonified with high-intensity sound (140–165 dB). Experimental investigations show that acoustic agglomeration may increase the efficiency of conventional particle filtering devices (such as cyclone filters or electrostatic precipitators) by modifying the aerosol's particle size distribution. However, existing models based on the classical orthokinetic kernel fail to predict certain important features of acoustic agglomeration. The new PAIS code overcomes this limitation by implementing acoustic drag interaction, a hydrodynamic effect that was only recently found to play an important role in the acoustic agglomeration process. As a result, new model predictions based on both kernels are compared with experimentally determined data. Important differences in the simulation with either kernel are pointed out. It is shown that the acoustic drag interaction kernel, in contrast to the classical model, accounts for the important agglomerations occurring between same and similarly sized particles.

4:15

4pEA10. Driving circuit of vibratory gyrosensor using damped capacitance elimination techniques. Katsutoshi Sakurai, Akira Satoh, Kazumasa Ohnishi (Magnetic Devices Div., Alps Electric Co., Ltd., 1-3-5, Higashitakami, Nagaoka, Niigata, 940 Japan), Yoshiro Tomikawa (Yamagata Univ., 4-3-16 Jonan, Yonezawa, Yamagata, 992 Japan), and Chiharu Kusakabe (Yamagata Univ., 1-4-12 Koshirakawa, Yamagata, 990 Japan)

In a vibratory gyrosensor, a high-quality factor and no frequency difference between driving and detecting frequencies are necessary to construct a high-sensitivity gyrosensor. In order to adjust driving and detecting

frequencies and driving position of the resonator in the gyrosensor, mechanical trimming, which means the partial cutting toward the resonator, is effective; however, such a trimming is complicated. For a method of electrical changing of the resonance frequencies of the resonator, it is well known to insert a condenser between the driving circuit and the driving electrode of the resonator; however, that method is usually used for a slight adjustment of the resonance frequency. On the other hand, a method was devised to change the resonance frequency of the piezoelectric vibrator, using damped capacitance elimination techniques, and it was utilized to a driving circuit of the vibratory gyrosensor. In this paper, sensitivity characteristics of the gyrosensor, whose driving frequency is changed using the damped capacitance elimination circuit, are reported.

4:30

4pEA11. Piezoelectric vibratory gyrosensor using a trident tuning-fork resonator. Akira Satoh (Dept. of Elec. Eng. and Information Sci., Faculty of Eng., Yamagata Univ., 4-3-16 Johnan, Yonezawa, Yamagata, 992 Japan and Magnetic Devices Div., Alps Electric Co., Ltd., 1-3-5 Higashitakami, Nagaoka, Niigata, 940 Japan), Yoshiro Tomikawa (Yamagata Univ., 4-3-16 Johnan, Yonezawa, Yamagata, 992 Japan), and Katsutoshi Sakurai (Alps Electric Co., Ltd., 1-3-5 Higashitakami, Nagaoka, Niigata, 940 Japan)

A piezoelectric vibratory gyrosensor, constructed using a trident tuning-fork resonator, is characterized by superior support construction. That is, for a trident tuning-fork, the vibrational loss with its support condition becomes so small that a gyrosensor has good performance compared with other existing type resonators. First, the vibration modes of a trident tuning-fork have been investigated, and suitable clamping conditions are clarified using finite-element analysis. Then, the method of adjusting the resonance frequencies between driving and detecting vibration modes and the influences of arm imbalance were studied. Moreover, when detecting the angular rates using a prototype trident tuning-fork resonator, the leakage output signal was apparently eliminated by connecting a differential circuit. As a result, a small-sized piezoelectric vibratory gyrosensor having good gyro-characteristics could be measured.

4:45

4pEA12. Phase criterion in the feedback cycle of low-speed edgetones. Young-Pil Kwon (Dept. of Mech. Eng., Soong Sil Univ., Sangdodong, Seoul 156-743, Korea)

The phase criterion of the feedback cycle of low-speed edgetones has been obtained using the jet–edge interaction model in which the reaction of the wedge to the impinging jet is modeled by an array of dipoles. The edgetone is produced by the feedback loop between the downstream-convected unstable flow issued from the nozzle and the upstream-propagating sonic wave due to the wedge reaction. By estimating the phase difference between the downstream flow and the upstream wave, and by imposing the phase-locking condition at the nozzle lip, it is found that the phase factor should be $p=-1/4$ in the feedback relation $h/\Lambda+h/\lambda=n+p$, where h is the stand-off distance, Λ and λ are the wavelengths of downstream- and upstream-propagating disturbances, respectively, and n is a stage number for the ladder-like characteristics of frequency. The proposed criterion $p=-1/4$ is just the opposite, in phase, to $p=1/4$ that has been regarded as an established characteristic of edgetones. However, $p=-1/4$ is verified substantially in comparison with available experimental data and found to be effective up to a wedge angle of 90°. [Work supported by KOSEF.]

5:00

4pEA13. Computations of acoustic wave strength and vorticity in shear flow. Laurine Leep-Apolloni (CAE Dept., Ford Res. Lab., Dearborn, MI 48121) and David R. Dowling (Univ. of Michigan, Ann Arbor, MI 48109)

While exact techniques are available for plane-wave propagation in uniform flow, acoustic propagation in nonuniform flow has proven much less tractable. A numerical technique has been developed to treat the simplest possible case: acoustic plane waves launched into a uniform shear. The computations are based on the Euler and continuity equations, using only the standard linearization for small acoustic amplitudes. When nondimensionalized, these equations produce a single governing parameter: m/f, where m is the flow's shear rate, and f is the acoustic frequency. Numerical solution for acoustic fluctuation velocity and density at each grid point allows wavefront location and orientation as well as wave strength to be determined. The results for wavefront location and orientation from the current numerical method are compared to a WKB solution obtained from the simplest nonuniform flow approximation which uses a nonuniform speed of sound in lieu of a nonuniform background flow. Gradients of acoustic field variables along the wavefronts, which are not predicted by the standard approximation, are also presented. The acoustic vorticity field is examined to study the rotational nature of the plane waves after interaction with the shear.

5:15

4pEA14. Further work on numerical simulations of aerosound from jets using the Lighthill theory with the k-ε turbulence model and the large eddy simulation (LES) procedure. William C. Meecham (Dept. of Mech. and Aerosp. Eng., Univ. of California, Los Angeles, CA 90095) and Seungbae Lee (INHA Univ., Inchon, Korea)

Acoustic mechanical power from fluctuating fluid flows forms but a tiny part (about 10^{-5}) of the power within the flow. The familiar, far-field solution to the Lighthill equation is used and is written as

$$\rho(x,t)=\frac{x_i x_j}{4\pi c^4|x|^3}\int_v\left[\frac{\partial^2 T_{ij}(\mathbf{y},t)}{\partial t^2}\right]dV,$$

where $T_{ij}=\rho U_i U_j$. An order of magnitude analysis gives the following intensity: $I=3E-5*\rho_0 L^2 U^3 \mathrm{Ma}^5/(4\pi x^2)$, where Ma is the Mach number and L is the scale of the turbulence. Direct numerical simulations for aerosound are limited to Re less than 2500 (so to not fully turbulent flows). The fluid flow must be calculated. This is done here in two steps. First, the time-honored k-ε turbulence model is used and then the LES technique. For this, one filters the equations, leaving only large scales. For the subgrid scales modeled, the accepted Smagorinsky model is used with a deductive model devised previously. Results are compared with the equation displayed above and the quadrupole character is checked for subsonic jets.

Session 4pMU

Musical Acoustics: Performance

Kengo Ohgushi, Cochair
Faculty of Music, Kyoto City University of Arts, 13-6 Kutsukake, Ohe, Nishikyo-ku, Kyoto, Kyoto, 610-11 Japan

Edward C. Carterette, Cochair
Department of Psychology, University of California, Los Angeles, California 90095

Invited Papers

2:00

4pMU1. The perception of rhythm in musical performance and poetry recital. Neil Todd (Dept. of Psych., Univ. of Manchester, Oxford Rd., Manchester M13 9PL, UK)

Over the last 2 decades much work has been done on the nature of expression in musical performance. Expression, which takes the form of subtle transformations of timing, dynamics, pitch and timbre, etc., may be measured fairly accurately when a score-based representation is available. It is generally agreed that the major function of expression in these cases is the communication of the performer's structural/motional interpretation of the piece being performed. Indeed, it is possible to demonstrate that the greater part of expressive variation may be accounted for by the two major components of rhythmic structure, namely, meter and grouping. However, little attention has been paid as to how the perception of rhythm may be influenced by expression. In this paper the perception of rhythm in expressive performance is reviewed from the point of view of a recent auditory-motor theory of rhythm perception [N. P. Todd and G. J. Brown, ''Vizualization of rhythm, time and metre,'' Artificial Intelligence Rev. (1996)]. The similarity between musical performance and the prosody of poetry recital is also discussed in light of a recent experiment to test the behavior of the model with speech. This is consistent with the view that rhythmic expression has a common origin in music and speech.

2:15

4pMU2. A measure of dissimilarity for comparison of dynamics, agogics, and pedaling in several piano performances. Tomoyasu Taguti (Faculty of Sci., Konan Univ., Okamoto, Higashinada, Kobe, 658 Japan), Kengo Ohgushi (Kyoto City Univ. of Arts, Kyoto, 610-11 Japan), and Tomoko Sueoka (Osaka College of Music, Toyonaka, 561 Japan)

Expressive performance of music is realized with local and global variations in dynamics, timing, intonation, timbre, etc., where particular profiles of these ''expressive elements'' depend on the players' performance practices based on their musical thoughts. This paper presents a method to study the correspondence between the physical nature and the subjective effect of expressive elements in piano performances. Let $f(s)$ be a function in musical time, where s represents the temporal profile of physical correlates with either dynamics, agogics, or pedaling in a piano performance, and let n be the number of these performances for the same music score. The dissimilarity d_{ij} between two performances i and j is defined as the root mean square of $f_i(s)-f_j(s)$ over the prescribed time span S of the concerned music passage. The discussion focuses on: (1) the construction of $f(s)$ for each of the above-mentioned expressive elements, and (2) the mapping of n performances in a geometrical space with the dissimilarity matrix $D=[d_{ij}]$, which is then related to the rating data of subjective impression obtained from a suitably designed listening session. As an example, the maps of dynamics, agogics, and pedaling are given for 24 performances for Chopin's Waltz No. 9 rendered by eight pianists with three kinds of intended musical expressions, and the rating data from a listening session are associated with the maps.

2:30

4pMU3. Communicating performer intent through musical performance. I. Mapping intent to perception. Roger A. Kendall and Edward C. Carterette (Dept. of Ethnomusicology and Psych., Univ. of California, Los Angeles, CA 90095)

Although several recent studies have involved experimental approaches to aspects of musical performance, most have focused on ''emotive'' communication or analyses of timing profiles in isolation. In this work, several frames of reference are integrated: notational (musical style), performer-generated messages (performer-intent), generator class (different instrument types), the link to the listener and listener knowledge (musician and nonmusician), and, finally, the mappings among frames. In this talk, methodological issues in the study of expressive musical performances are discussed, focusing on convergent operational definitions of variables. It is demonstrated, by experimental results, how different listener operations explicate different aspects of expressive communication. A unique feature of this work is the use of different musical style categories (baroque, classical, romantic, and early 20th century) which initiate the chain of musical communication, providing different contexts for the performer–listener mapping.

2:45

4pMU4. Communicating performer intent through musical performance. II. Mapping performer intent through acoustical analyses. Edward C. Carterette and Roger A. Kendall (Dept. of Psych. and Ethnomusicology, Univ. of California, Los Angeles, CA 90095)

In this phase of this research, the acoustical signals generated in expressive performance by orchestral instruments are parsed. Most studies of musical expression and performance have concentrated on the profiles that arise from MIDI timing data. Here, the signals of ordinary orchestral instruments are employed from these studies of expressive communication, and analyses are provided that map timing, amplitude, and spectral measures to performer messages. Different instruments have different combinations of drivers, resonators, and couplings, and the resultant acoustical signal contains information about these. Recently, in the laboratory, Hajda has explored spectral centroid trajectories as a means of identifying attack characteristics, and this notion is applied to the issue of legato transitions in expressive musical performance. Timing and amplitude trajectory data from these experiments suggest that, while lesser musicians have a tendency to repeat similar gestures over the range of expression, highly trained performers at a virtuoso level are capable of distinct timing profiles under stylistically appropriate conditions which communicate their intent.

3:00

4pMU5. Acoustic correlates of the emotional expression in vocal performance. Kengo Ohgushi and Manami Hattori (Faculty of Music, Kyoto City Univ. of Arts, 13-6 Kutsukake, Ohe, Nishikyo-ku, Kyoto, 610-11 Japan)

Listening experiments and physical analyses of vocal performances were made to investigate how the emotional expressions were transmitted to listeners and to find acoustic correlates of each emotional expression. Three female singers were instructed to perform a short piece of music so as to communicate five basic emotions to listeners. The five basic emotions were ''happiness,'' ''sadness,'' ''anger,'' ''fear,'' and ''without expression.'' Some listening experiments revealed that, on the whole, the singer's intention and the listener's impression, although there were individual discrepancies, were in general agreement. The performances were analyzed regarding physical parameters such as tempo, dynamics, agogics, intonation, vibrato, and so on. The relationships between the expressive intention of the singers and measured physical parameters were revealed.

3:15

4pMU6. Acoustic correlates of emotionally expressive music. Alf Gabrielsson (Dept. of Psych., Uppsala Univ., Box 1854, S-751 48 Uppsala, Sweden)

Music is known for its ability to express as well as to evoke emotions. However, there is very little scientific research on what means musicians use in their performance to bring about different emotional expressions. In a series of ongoing experiments musicians are asked, using various instruments (violin, flute, electric guitar, synthesizer, singing voice), to perform short monophonic pieces of music to have them express ''happiness,'' ''sadness,'' ''anger,'' ''fear,'' ''solemnity,'' ''tenderness,'' and ''no expression.'' Listening experiments are conducted to confirm that listeners recognize the intended emotions. Approved performances are analyzed using sampling systems. The intended expressions affect practically all physical variables—tempo, timing of various units, articulation, sound level, amplitude envelopes, intonation, spectrum, onsets and decays, vibrato—in ways that differ among expressions and instruments used. There is considerable interindividual variation among performers in their ability to render the intended expressions as well as regarding the means that they use to achieve the expressions. On the other hand, listeners differ in their ability to recognize the intended expressions. Some emotions seem easier to express as well as to recognize than others.

Contributed Papers

3:30

4pMU7. Rhythmic interpretation of polonaise in piano performances. Kimiko Ohta (Graduate School of Eng., Osaka Univ., 1-2 Yamadaoka, Suita, 565 Japan) and Tomoyasu Taguti (Konan Univ., Higashinada, 658 Japan)

A method for analysis of rhythmic interpretation of a characteristic rythmic pattern in polonaise is presented. The pattern concerned is one 8th note followed by two 16th notes. By physical measurement of timing for this pattern as it appeared in the first beat in several performances of Chopin's polonaise (Grande Polonaise Op. 22), the duration of this pattern was generally found to be lengthened considerably. Further, a listening experiment showed that this portion had a rhythmic stress (of lengthening accent). To characterize the expressive timing of the pattern, a pair of ratios, $P=[a,b]$ with $a=x:1$ and $b=1:y$, was introduced, where a denotes the ratio of durations of the 8th note to the sum of two 16th notes, and b the ratio of the two 16th notes. The P's from the performances by six pianists were plotted in the x-y plane. The result revealed a clear distinction among individual rhythmic interpretations of the six pianists, e.g., $x<1$ and $y>1$ for the P's of three pianists, which implies that they put a stress on the second 16th note.

3:45

4pMU8. The context effect on loudness in listening to music. Keiko Arakawa (Lab. of Musicology, Graduate School of Lit., Osaka Univ., 1-16 Matikaneyama cho, Toyonaka city, Osaka, 560 Japan), Tazu Mizunami (Osaka Univ.), Sonoko Kuwano (Osaka Univ.), Seiichiro Namba (Takaraduka Univ. of Art and Design), and Toru Kato (Otemon Gakuin Univ.)

As the first stage of the research into factors determining the optimum listening level of music performance, its methodology was examined. In experiment 1, a part of Symphony No. 6 in b-minor Op. 74, ''Pathétique,'' composed by Tchaikovsky, was used and the instantaneous impressions of loudness were judged using the method of continuous judgment by categories developed by Namba and Kuwano. A high correlation coefficient was found between LA_{eq}, 100 ms, and instantaneous impression of loudness sampled every 100 ms, which confirms the applicability of the method to the loudness of music performance. In experiment 2, musically significant portions whose durations were 5 to 20 s, were selected from the stimulus used in experiment 1 and rearranged in descending series and ascending series according to their sound levels with about 5-s silent intervals. The instantaneous impression of their loudness was judged as in experiment 1. It was found that each block in the ascending series was judged louder than the corresponding block in the descending series. It was suggested that the loudness of musical performance is significantly affected by the context.

4:00

4pMU9. Interaction between auditory and visual processing in impressional evaluation of a piano performance. Haruka Shimosako (Inst. of Psych., Univ. of Tsukuba, Tsukuba, Ibaraki, 305 Japan) and Kengo Ohgushi (Kyoto City Univ. of Arts, Kyoto, 610-11 Japan)

To investigate the interaction between auditory and visual processing in impressional evaluation of a piano performance, a pianist was asked to play the same musical excerpt in three different expressive levels and these were recorded on a video. Three kinds of sounds and three kinds of body movements were combined and these nine performances were presented to musically trained students and less trained students. They rated the expressivity of each performance on a five-point scale in three modes: sound and vision together, sound alone, and vision alone. When sound and vision were presented, they also rated their subjective acceptability on a five-point scale. The result was that both musically trained students and less trained students could discern among the levels of a pianist's expressive intent when sound alone or vision alone was presented. However, when sound and vision were presented, it was found that vision had a greater influence on the impressional evaluation. This tendency was more remarkable for less trained students than musically trained students. When sound and body movements were at different levels, performances were rated lower by musically trained students but not by less trained students.

4:15

4pMU10. Motion as music: Comparing final ritards with stopping runners. Anders Friberg and Johan Sundberg (Speech, Music and Hearing, Royal Inst. of Technol., Box 70014, S-100 44 Stockholm, Sweden)

Music and motion are generally assumed to be closely related. An attempt was made to analyze such relations with regard to the stopping of running and the termination of a piece of music. Velocity, step frequency, and step length of four professional dancers were measured while they were stopping from running. Six choreographers rated the esthetic quality of the decelerations from video recordings. The data curves from highly rated decelerations were more regular and smooth as compared to those decelerations rated lower. It was found that the average instant velocity of the runners corresponded well with the average of instant tempo in final ritards. A model of the ritard shape with varying curvature, including the observed deceleration shape, was applied to three music examples in a listening experiment. The observed deceleration shape was found to be within the range of the subjects' preferences.

4:30

4pMU11. Measurement of performance response differences using aluminum and brass marimba resonators. Ronald A. Roberts (Ctr. for Non-Destructive Evaluation, 138 AscII, Iowa State Univ., Ames, IA 50011) and Barry J. Larkin (Iowa State Univ., Ames, IA 50011)

Historically, percussionists have maintained that a perceptible difference exists in the responses of aluminum and brass marimba resonators. Brass resonators are thought to produce a fuller and richer sound, whereas aluminum resonators are considered to yield an inferior, thinner sound. Marimba performers will seek out and pay a premium price for instruments equipped with brass resonators, while lower cost aluminum resonators are found primarily on student model instruments. Experiments are being performed to determine if any such difference can be detected in laboratory measurements. This paper will outline experimental techniques used in recording airborne responses generated by repeatable, mechanized mallet strokes. Data analysis will be summarized, and results will be presented showing the nature of measured differences in both overtone structure and response evolution.

4:45

4pMU12. Basic study of acoustic display of John Milton's *Paradise Lost*. Mineo Moritani (Dept. of English, Bukkyo Univ., Kyoto, 603 Japan)

Knowing how emotions change with sound pressure is the present concern with a synthesized acoustical display of this poem. Emotional changes in the poem are felt like the music of Beethoven. The sound pressure alone cannot express the delicate nuances of the emotion; other elements are needed for the whole emotional expression. But the sound pressure is the basic medium by which emotion is communicated; when one is emotionally excited, one often raises his voice. So, in the epic, strong sound pressure is assumed to be an expression of strong emotion. Hence, it occurred to the author to represent the emotional changes in terms of those of sound pressure. It is clear that each book has its idiosyncratic pattern of change of sound pressure; hence, emotion is inseparably related to the contents. Most interestingly, the fall of Eve and Adam appears in Book IX with two changes of abnormally high increase of sound pressure to about 80 dB against the average 73 dB. The change of emotion can be heard with a sine-wave generator using techniques developed by the author.

5:00

4pMU13. The touch precursor of the piano tone. Daisuke Naganuma and Isao Nakamura (Teikyo Heisei Univ., 2289 Uruido, Ichihara, Chiba, 290-01 Japan)

Recently research has discovered that sound appears before the contact of the string with the hammer in the piano tone. This sound is called the "touch precursor" or "early noise" and is observed only in a staccato type of touch [A. Askenfelt, Proc. SMAC **93**, 297–301 (1993); G. W. Koornhof and A. J. van der Walt, *ibid.* **93**, 318–324 (1993)]. In this article, analysis of the touch precursor is reported for two cases. The first case analyzed was the piano tone at three different levels (mezzoforte, forte, and fortissimo) played by a mechanical actuator. The touch precursor was observed in each sound. The results show that the lower the touch level becomes, the longer the duration time of the touch precursor becomes. The frequency of the touch precursor is almost constant, but slightly lower in the low-frequency range. The amplitude of the touch precursor gradually increases from the onset until the beginning of the string vibration. The second case analyzed was the piano tone played by a pianist using two different staccato touches, who pressed down the key by a strained finger and a relaxed finger. The results show that the duration time of the touch precursor is slightly different.

5:15

4pMU14. On the duration of pre-Helmholtz triggering in bowed violin attacks. Knut Guettler (The Norwegian State Acad. of Music, P.O. Box 5190 Majorstua, N-0302 Oslo, Norway) and Anders Askenfelt (Royal Inst. of Technol., S-100 44 Stockholm, Sweden)

The attacks of most bowed notes on a violin show a nonperiodic initial part before Helmholtz triggering occurs. Depending on the particular combination of bowing parameters, this state is characterized either by periods which are *prolonged*, or by a division of the nominal period into two or several parts, *multiple flyback*. A "perfect" onset with only one slip per period from the very start is also possible. The present study includes: (a) computer simulations of a violin G-string and characterization of the attack quality as function of bowing parameters; (b) evaluation of acceptability of a series of (machine-bowed) violin attacks with respect to duration of the pre-Helmholtz triggering; the evaluation took place in a listening test that involved music students and professionals; (c) analyses of 1694 attacks in different musical contexts as performed by two professional violinists. The limits for acceptability established by the listening test were about 50 and 90 ms of prolonged and multiple-slip periods, respectively, for an open G-string (196 Hz). These demands were confirmed by the playing test, even for other strings and a larger group of bowing styles. [Work supported by The Nordic Research Council.]

4p THU. PM

Session 4pNS

Noise, Structural Acoustics and Vibration and Signal Processing in Acoustics: Active Control in Sound and Vibration

Brian H. Houston, Cochair
Naval Research Laboratory, Code 5136, Overlook Avenue, Washington, DC 20375

Tsuyoshi Usagawa, Cochair
Department of Computer Science, Faculty of Engineering, Kumamoto University, 2-39-1 Kumamoto, Kumamoto, 860 Japan

Contributed Papers

2:00

4pNS1. The configuration of an adaptive prediction-type active noise control system. Shinichi Honda and Hareo Hamada (Dept. of Information and Commun. Eng., Tokyo Denki Univ., 2-2, Nishiki-tyo, Kanda, Chiyoda-ku, Tokyo, 101 Japan)

In this paper, a new active noise control system is proposed using an adaptive prediction algorithm. The adaptive prediction-type active noise control system has many attractive advantages; for example, no prior knowledge of primary noise fields will be needed. However, difficulty in the control of broadband noise was encountered. For example, a new system configuration with an auxiliary sensing microphone is proposed. The computer simulation revealed that an auxiliary sensing microphone can improve the noise reduction remarkably in a wide frequency range if the effective sensing of primary noise is not expected by a single microphone. It is also demonstrated that the improvement of noise reduction is considerable depending on the arrangement of sensing microphones.

2:15

4pNS2. Passive–active broadband sound absorption. Jerome P. Smith and Ricardo A. Burdisso (Dept. of Mech. Eng., Virginia Tech., Blacksburg, VA 24061)

Researchers have proposed hybrid systems to achieve high sound absorption over a wide frequency range. The passive component of the system is comprised of a layer of sound absorbing material positioned at a distance from a movable wall, leaving an air space. The wall is the active component and is used to increase the absorption of the system at frequencies where the passive system is not effective. The control input that drives the active wall is determined in order to achieve a desired boundary condition at the back of the absorbing layer. Both pressure-release (i.e., minimizing the pressure at the back surface of the layer) and impedance-matching (i.e., minimizing the reflected wave in the cavity) conditions have been proposed to increase the absorption of the system. The performance of the hybrid system for these two boundary conditions is compared for broadband disturbances over a frequency range of 0–1000 Hz. The passive system showed absorption coefficients greater than 0.7 only above 500 Hz. The impedance-matching condition yielded absorption coefficients of 0.8–1.0 over the 100- to 1000-Hz range, which was significantly better than the absorption achieved with the pressure-release condition. Sensitivities of these two control approaches to system parameters are also investigated.

2:30

4pNS3. Modal decomposition results for active control of energy density. Scott D. Sommerfeldt (Dept. of Phys., Brigham Young Univ., 277 FB, Provo, UT 84602), John W. Parkins, and Jiri Tichy (Penn State, P.O. Box 30, State College, PA 16804)

In recent years, an alternative sensing approach has been developed for active control, based on minimizing the acoustic energy density at the error sensor location(s). This new approach has been tested both numerically and experimentally, with the results indicating that one can often achieve improved global attenuation of the field by minimizing the acoustic energy density, rather than the sum of the squared pressures. Previous results from minimizing the energy density at the error sensors have concentrated on investigating the control that can be achieved by looking at the global energy in the field before and after control, and also by looking at the attenuation that can be achieved as a function of frequency. However, it has also been found that additional insight can be gained by examining the acoustic field in terms of the acoustic modes contributing to the acoustic field. This paper will present some of the modal decomposition results obtained for different active control approaches. These results provide insight into the control mechanisms and provide indications as to why one can often achieve improved global attenuation by minimizing the acoustic energy density rather than the squared pressure.

2:45

4pNS4. An adaptive algorithm with error path estimation for active control of periodic noise. Nobuyuki Seguchi, Yasuyuki Shimada, Yoshitaka Nishimura, Tsuyoshi Usagawa, and Masanao Ebata (Dept. of Comput. Sci., Kumamoto Univ., 2-39-1 Kurokami, Kumamoto, 860 Japan)

When active noise control (ANC) systems are actually implemented, it is necessary to take account of the modeling error of an error path. The modeling error limits the performance of ANC and it makes the system unstable at worst. The characteristics of the error path cannot be assumed to be stable, so that on-line estimation is preferred for actual applications such as an active control system of the exhaust noise of an engine. An adaptive algorithm called the delayed-X harmonics synthesizer (DXHS) algorithm for periodic noise has been proposed. This algorithm controls amplitude and phase parameters for each harmonic component. The original algorithm controls parameters represented in polar coordinates with constant delay parameters. This paper presents an extended algorithm to estimate delay parameters of the error path on-line. Delay parameters, which are important for the stability, are introduced as a phase, while other parameters are represented in orthogonal coordinates. The performance of the algorithm is evaluated by computer simulations. When SNR at the control point is 0 dB, the estimate delay is nearly the same value as the actual delay, and the system converges very stably.

3:00

4pNS5. Three-dimensional sound field simulation using the boundary integral equation method. Shinichi Katsumata and Hareo Hamada (Dept. of Information and Commun. Eng., Tokyo Denki Univ., 2-2, Kanda-Nishiki-cho, Chiyoda-ku, Tokyo, 101 Japan)

Computer simulation is a useful technique for developing the active control system (referred to as the ANC system). In this paper, the technique is introduced of sound field analysis for an ANC system based on the boundary integral equation method (BIEM). Since the BIEM accounts for wave motion characteristics precisely, the dynamic behavior of the control sound field can be considered, especially in the low-frequency band. The objective of the study is to investigate a transient sound field by applying the BIEM to practical problems. In order to introduce the BIEM technique to actual situations, including boundary problems with the existing acoustic materials, a database was developed on the normal acoustic admittance ratio of the materials. The ordinary feedforward model of the ANC system was used to control the noise field in the computer simulations. The results, in which the transient sound fields controlled by the ANC system are shown, will be very useful in understanding how to build up the control sound field.

3:15

4pNS6. A study of visualization of a sound field. Takeshi Suzuki and Haero Hamada (Dept. of Information and Commun. Eng., Tokyo Denki Univ., 2-2, Nishiki-cho, Kanda, Chiyoda-ku, Tokyo, 101 Japan)

The generated noise caused by the fluid inside a duct is difficult to estimate accurately due to the difference of power between fluid and noise caused by the fluid. However, it would be required to estimate the distribution of sound pressure inside the duct to introduce the active noise control techniques effectively. The objective of this paper is to estimate the radiated acoustical energy and to visualize the distribution of the sound pressure inside an air-conditioning duct using the acoustical power balance method. This method is suitable for analyzing complicated random noise at a fairly high frequency. Also introduced was a new method of the boundary integral equation to analyze the dynamic behavior of sound fields. The analysis of sound fields controlled by an active noise control system will be shown with the dynamic time-domain visualization of the sound-pressure field.

3:30–3:45 Break

3:45

4pNS7. Sound diffraction effects by screens in a room. Dominique M. Habault (CNRS, Lab. de Mecanique et Acoustique, F-13402 Marseille Cedex 20, France) and Ulf R. Kristiansen (N. T. N. U., Acoust. Group, N-7034 Trondheim, Norway)

Several numerical methods are used to describe the effects of diffraction by two kinds of screens in a room: a perfectly rigid screen and a vibrating screen. This study has two main applications in acoustics: In room acoustics, the aim is to describe the influence of a vibrating screen on the acoustical characteristics of a room in which screens made of light materials are suspended from the ceiling; in structural acoustics, it is to describe the effect of a vibrating structure in a cavity. An application in active noise control is also presented. First, the sound pressure in the room is computed by using a boundary element method. Then, a perturbation method is used in the case of the vibrating screen. Comparisons are made between the two methods and a finite-element method for two typical geometries. The numerical examples presented correspond to a 2-D problem and a rectangular room. From the acoustics point of view, it is shown that the effect of a vibrating screen compared with a rigid screen is bigger at low frequencies and for light materials. This difference also depends on the acoustical properties of the room (i.e., the reflection coefficients on the walls).

4:00

4pNS8. Multivariable structural acoustic control with static compensation. Robert L. Clark (Dept. of Mech. Eng. and Mater. Sci., Duke Univ., P.O. Box 90300, Durham, NC 27708-0300) and David E. Cox (NASA Langley Res. Ctr., Hampton, VA 23681-0001)

The active control of turbulent boundary layer noise in commercial aircraft has gained significant interest in recent years, and, due to the stochastic nature of the disturbance, feedback control strategies are required. Linear quadratic Gaussian control has been proposed in the past [Baumann *et al.*, J. Acoust. Soc. Am. (1991)], but compensators resulting from such control approaches are typically quite involved. The thrust of the current effort is to demonstrate that acceptable performance in structural acoustic control can be achieved with static compensation implemented through a multiple input, multiple output (MIMO) array of colocated transducers. Since static compensation requires a matrix of constant gains, it is easily realized in analog or digital hardware. By augmenting the system dynamics to include radiation filters, the dual Levine–Athans algorithm was implemented to design the appropriate static compensator. Results for various MIMO control cases are presented.

4:15

4pNS9. Multivariable active structural acoustic feedback control using piezoelectric sensoriactuators. Jeffrey S. Vipperman and Robert L. Clark (Dept. of Mech. Eng., Duke Univ., Box 90302, Durham, NC 27708)

Analytical and experimental results from the implementation of a feedback, active structural acoustic control (ASAC) system are presented. The test bed is a rectangular, simply supported steel plate inserted in a semi-infinite baffle inside an anechoic chamber. Piezoceramic patches are wired as bending transducers and implemented as adaptive piezoelectric sensoriactuators or self-sensing actuators. The transfer function across a sensoriactuator is minimum-phase, thereby enhancing stability for feedback control applications. Radiation filters are implemented to design the performance cost used in the control synthesis. Several design parameters are examined for different square (equal number of inputs and outputs) control configurations. Both single- and multichannel control system implementations are considered. Sensor and actuator optimization is performed using nonlinear optimization techniques, and both dynamic and static compensation are compared.

4:30

4pNS10. Control of sound radiation from panels using multiple globally detuned vibration absorbers. C. R. Fuller, F. Charette, and J. P. Carneal (Vib. and Acoust. Labs., Dept. of Mech. Eng., Virginia Polytechnic Inst. and State Univ., Blacksburg, VA 24061)

In this paper, the results of an investigation on using multiple globally detuned vibration absorbers to minimize sound radiation from vibrating elastic panels will be presented. The vibration absorbers consist of lumped mass–spring systems whose characteristics (i.e., resonance frequency) can be electronically varied. A fully coupled multiple-channel control algorithm, implemented on a DSP, is used to adapt the characteristics of the multiple absorbers so as to minimize the radiated sound at selected positions in the far field. Both theoretical and experimental results are presented and are compared to those obtained using tuned (i.e., to the disturbance frequency) vibration absorbers. Improved sound attenuation is shown to be obtained using the globally detuned vibration absorbers. The results are very similar in characteristics to those obtained previously using active structural acoustic control implemented with electrodynamic shakers or piezoelectric actuators. The results thus suggest a combined active–passive method to minimize structural sound radiation which requires lighter, smaller actuators and much lower control electrical power. [Work supported by ONR and NASA LaRC.]

4:45

4pNS11. Active vibration isolation using vibrational power as a cost function. Carl Q. Howard and Colin H. Hansen (Dept. of Mech. Eng., Univ. of Adelaide, S.A. 5005, Australia)

Active isolation of a vibrating rigid mass from a flexible simply supported beam is investigated theoretically and experimentally. Results obtained using vibrational power transmission as a cost function are compared with those obtained using base acceleration and transmitted force as cost functions. One problem with using vibratory power transmission as a cost function is the lack of correlation between the reference and error signals due to the frequency doubling effect of the multiplication operation used to obtain power. This problem is addressed by heterodyning the frequency doubled-error signal with the reference signal to obtain an error signal with the same frequency components as the reference signal, but with the amplitude dependent on the frequency doubled-error signal amplitude. Theoretical results indicate that it is possible to achieve zero vibratory power transmission into the beam irrespective of whether the cost function used is vibratory power, beam acceleration at the base of the isolator, or transmitted force. Experimental results show that similar isolation efficiencies are obtained for all three types of cost function with the achievable efficiency dependent upon the accuracy of the digital control system.

5:00

4pNS12. Active control of vibration intensity flow in a beam in the far field. Yukio Iwaya, Masato Sakata (Dept. of Elec. and Electron. Eng., Akita Univ., 1-1 Tegata Gakuen cho, Akita, Japan), Yoiti Suzuki, and Toshio Sone (Tohoku Univ., 2-1-1 Katahira, Aoba-ku, Sendai, Japan)

From the point of view of physical impedance, vibration intensity carried by a bending wave propagating along a beam can be decomposed into two components: One is the component owing to a mechanical impedance Z, and the other is owing to a bending moment impedance Mz. To control intensity, both components must be taken into consideration, though it causes some difficulties in execution, such as a complex cost function, many sensors, etc. However, a one-dimensional bending wave consists of four waves. Two of them are decaying waves and these can be negligible in the far field, which means far from the edge of the beam. In the far field, Z becomes the reciprocal of Mz. Using this relationship, three simple ways were found to control intensity as follows: (1) control amplitude at one point; (2) control of amplitude gradient, and (3) direct intensity control in the frequency domain. Computer simulations have been performed in these cases. The results were that case (3) showed the slowest convergence of parameters, and cases (1) and (2) could give equally good results. Furthermore, two shakers have been mounted on a beam and an experiment of control with a DSP has been accomplished for case (2).

THURSDAY AFTERNOON, 5 DECEMBER 1996 WAIALUA ROOM, 2:00 TO 5:45 P.M.

Session 4pPA

Physical Acoustics: General Topics in Physical Acoustics

Thomas J. Matula, Cochair
Applied Physics Laboratory, University of Washington, 1013 N.E. 40th Street, Seattle, Washington 98105

Kenshiro Takagi, Cochair
Institute of Industrial Science, University of Tokyo, 7-22-1 Roppongi, Minato-ku, Tokyo, 106 Japan

Contributed Papers

2:00

4pPA1. Ultrasonic study of the main (α) relaxation in an epoxy resin. Mami Matsukawa (Dept. of Electron., Doshisha Univ., Tanabe, Kyoto, 610-03 Japan) and Norikazu Ohtori (Niigata Univ., Niigata, 950-21 Japan)

The isothermal polymerization process of a common epoxy system (prepolymer: diglycidyl ether of bisphenol A, curing agent: diamine) has been characterized in the MHz range by ultrasonic pulse spectroscopy. The longitudinal wave velocity increased during polymerization. The profile of longitudinal wave velocity and attenuation depended on the curing condition (temperature and mixing ratio of the curing agent). In addition, velocity and attenuation showed clear effects of the main (α) relaxation, which can be observed during glass transition of the resin. This polymerization process was also observed in the GHz range by Brillouin-scattering measurements. In this frequency range, the polymerization process was also observed as the increase of velocity. However, the velocity values were much higher than those observed in the MHz range from the beginning of polymerization. By considering the glass transition temperature of the prepolymer (about -20 °C) used, this velocity difference indicates the possibility that the main (α) relaxation of prepolymer used situated between MHz and GHz ranges. To confirm this idea, the temperature dependences of the velocities were observed. The velocity difference became smaller as the temperature increased, which showed a smaller effect of relaxation at higher temperatures.

2:15

4pPA2. A general energy conservation law in dielectrics. Donald F. Nelson (WPI, Worcester, MA 01609)

To study the interaction of acoustic waves with other excitations in a dielectric crystal it is necessary to have a consistent set of coupled dynamic equations for them. A particularly important consequence of those equations is the energy conservation law. A Lagrangian-based theory has been used to produce the most general energy conservation statement yet obtained in a dielectric. It includes acoustic waves, electromagnetic waves, spin waves, and optic modes of ionic, electronic, and excitonic origin. Further, the excitations can be arbitrarily nonlinear in any field or combination of fields, can interact with any multipole order of the bound charge density and its current density, can involve any derivative of a field, and can occur in crystals of any symmetry. It is applied to a soliton that is a mixed mode (a ''polariton'') of an acoustic wave and a ''soft'' optic mode at a temperature just above a ferroelastic phase transition. Such an interaction involves the acoustic coupling to the optic mode turning the quadratic potential energy of the latter negative and so needs a quartic energy to stabilize it. Allowed soliton velocities fill the gap of linear wave velocities produced by the polaritonic interaction. [Work supported by NSF.]

2:30

4pPA3. Thermal phonon resonance in a cylindrical cavity. Koichiro Hattori, Keiji Sakai, and Kenshiro Takagi (Inst. of Indust. Sci., Univ. of Tokyo, 7-22-1, Roppongi, Minato-ku, Tokyo, 106 Japan)

Resonance phenomena were observed of a thermally excited phonon confined in a very small space. The power spectrum changes due to the boundary condition of the cavity shape. A cylindrical cavity which has a 1450-μm inner diameter containing water was prepared as a sample. The resonance frequency in the cylindrical cavity is given by a solution of Bessel's function. An optical beating system is used which has a hyper-frequency resolution to detect this fine structure of the resonance spectrum. The spatial decay of phonons is very large in the liquid sample. The low-frequency measurement is required to observe the resonance phenomena, because the phonon can reflect many times in the cavity. Generally though, a broadening of the wave vector due to angular divergence of the laser light is very large in a small-angle light-scattering measurement, since the wave vectors of the resonance peaks are determined only by a boundary of the cavity. Several resonance peaks can be observed around the 8 MHz measurement and a typical peak width is 50 kHz. The strong peaks are observed at every 500 kHz. These peaks are the symmetric and lowest asymmetric tangential modes.

2:45

4pPA4. Acoustic cavity resonances as a probe of the interior surface geometry of nearly spherical closed shells. Thomas J. Asaki, James K. Hoffer, and John D. Sheliak (MS K764, Los Alamos Natl. Lab., Los Alamos, NM 87545)

Spherical shells are used for a variety of applications ranging from fundamental gas properties measurement in large resonators to studies involving millimeter-size prototype intertial-confinement-fusion (ICF) targets. Most applications demand that either or both shell surfaces have a high degree of sphericity. Fabrication from hemishells inevitably leads to geometric imperfections and leaves the interior surface unavailable to metrological inspection. However, many common asphericities are predicted to break the degeneracy of resonator modes to first order in a boundary perturbation expansion [J. B. Mehl, J. Acoust. Soc. Am. **79**, 278–285 (1985)]. Various centimeter size aluminum and beryllium shells were manufactured with artificially enhanced common fabrication errors. The resonant modes of a completely closed shell are excited and detected using a swept-sine heterodyne technique [A. Migliori *et al.*, Physica B **183**, 1–24 (1993)] with four pinducers in contact with the exterior at the tetrahedral angles. Both shell and interior gas resonances are distinguishable, are nearly uncoupled, and display measurable mode splitting. While various gases, pressures, and temperatures are examined, primary research involves materials and conditions relevant to the manufacture of ICF targets. Experimental results are compared with the perturbation theory, and the utility of the technique is examined.

3:00

4pPA5. Acoustic pre-condensation of deuterium. James K. Hoffer, Thomas J. Asaki, and John D. Sheliak (MS K764, Los Alamos Natl. Lab., Los Alamos, NM 87545)

Millimeter to centimeter size light-metallic shells are of interest as investigative tools in the inertial-confinement-fusion community. Such shells are typically filled at room temperature with pure or mixed hydrogen isotopes at high pressure. At low temperatures, the hydrogens condense forming a solid or liquid in equilibrium with the vapor. The sound speed of gaseous deuterium (1% He-3) in a closed aluminum shell has been investigated using a cavity/shell resonance method. An anomalous decrease in the measured sound speed is observed as the saturation pressure P_s (about 15 atm at 38 K) is approached during a nearly constant-volume reduction in temperature. This effect is explained qualitatively in terms of a reduced acoustic admittance at the shell wall due to a precondensed liquid layer at pressures below P_s [J. B. Mehl and M. R. Moldover, J. Acoust. Soc. Am. **77**, 455–465 (1982)]. Comparison is made with precondensation phenomena observed by various authors for single-component gases in experiments performed at constant temperature.

3:15

4pPA6. An ultrasonic-electrostatic hybrid levitator for crystal growth from a solution. Sang K. Chung and Eugene H. Trinh (JPL/Caltech, MS 183-401, 4800 Oak Grove Dr., Pasadena, CA 91109)

A novel approach to crystal growth from a solution has been introduced by combining the electrostatic levitation technology with the sample rotation capabilities of a single-axis ultrasonic levitator. In the current implementation, a 22-kHz standing wave is used to position and to rotate an electrostatically levitated 0.4-cm-diam droplet along a horizontal axis. Current studies are investigating the specific capabilities for rotation control (rotation rate and rotation axis orientation) as well as the trajectory of an immersed liquid and solid sample within the levitated solution droplet. The physical mechanism for torque generation is currently believed to be the aerodynamic drag from acoustic streaming flows external to the levitated droplet. The results of internal and external flow visualization studies are presented. [Work funded by NASA.]

3:30

4pPA7. Relation for calculating the heat capacity at constant volume for multicomponent systems by using the values of sound velocity for the pure components and binary systems. Dmitrii A. Denisov (Dept. of the Colloid Chemistry, Mendeleev Chemical Technol. Univ. of Russia, Miusskaya pl. 9, Moscow 125047, Russia)

It is known that heat capacities at constant volume C_V can be calculated by using the relation connecting C_V with values of sound velocity for these substances [A. J. Matheson, *Molecular Acoustics* (Wiley, London, 1971)]. The relation for calculating C_V for mixtures following the lattice model of regular mixtures in the zero quasichemical approximation [I. Prigogine, *The Molecular Theory of Solutions* (North Holland, Amsterdam, Interscience, New York, 1957)] on the basis of some data including sound velocity referring to the pure components has been derived. The analogous problem for mixed solutions, namely for the solutions containing several solutes, has been solved. The relation for calculating C_V for a mixed solution consisting of the solvent and one of the solutes and having the same solvent chemical potential value (which the considered mixed solution has), follows the same conditions.

3:45

4pPA8. Observation of phase transition and critical behavior in Langmuir films by a ripplon light scattering technique. Naoto Sakamoto, Keiji Sakai, and Kenshiro Takagi (Inst. of Industrial Sci., Univ. of Tokyo, 7-22-1, Roppongi, Minato-ku, Tokyo, 106 Japan)

The light-scattering study of ripplon (thermally excited surface tension waves) was used to measure the surface elasticity of a myristic acid monomolecular film spread on a water surface in its coexisting state of two-dimensional liquid and vapor phases. The temperature dependence of the surface elasticity of both phases suggested that the critical behavior analogous to that of the usual three-dimensional fluids occurred in this two-dimensional system. The surface elasticity of the liquid phase decreased monotonously with temperature from 1.2×10^{-2} N m^{-1} at 20 °C to 2×10^{-3} N m^{-1} at 60 °C, while that of the vapor phase showed no appreciable difference from the value on a clean water surface. The Van der Waals equation of state specially reduced for the two-dimensional fluid was used to explain the result, and the Π-A isotherms were theoretically calculated so that the surface elasticity predicted from these curves agrees well with the observed values. The two parameters characterizing the equation were determined, $a=6.5\times10^{-39}$ N m^3 and $b=3.9\times10^{-19}$ m^2, which are associated with effects of the attractive force and the excluded area, respectively. These values gave the critical temperature $T_C=85$ °C.

4:00

4pPA9. Mechanical properties of a hyperswollen lyotropic smectic liquid crystals at low frequencies. Jun Yamamoto and Hajime Tanaka (Inst. of Industrial Sci., Univ. of Tokyo, Roppongi 7-22-1, Minato-ku, Tokyo, 106 Japan)

Smectic liquid crystals are unique condensed phases in that they exhibit properties of a one-dimensional solid and of a two-dimensional liquid. There exist various types of the hydrodynamic modes, propagating and diffusive modes, such as second sound or layer undulation modes, related to the existence of the internal degree of freedom in the complex fluid. Mechanical properties have been investigated of hyperswollen lyotropic smectic phases, with focus on its low dimensionality and large intrinsic fluctuations. This layer structure is supported by the microscopic long-range intermembrane interactions such as electrostatic or steric repulsion that play an important role in the hyperswollen regime. The layer compressibility measurements tell one that the layer structure is supported by entropic repulsion, which originated in the undulation fluctuation motion of the membranes. The light scattering experiments are also shown which provide information on the relation between the static structure and the dynamics of hydrodynamic fluctuation modes.

4:15

4pPA10. Brillouin scattering in the isotropic phase of liquid crystals. Taketo Ueno, Keiji Sakai, and Kenshiro Takagi (Inst. of Industrial Sci., Univ. of Tokyo, 7-22-1, Roppongi, Minato-ku, Tokyo, 106 Japan)

An optical beating technique was applied to the study of thermal fluctuations in isotropic phases of nematogens (those liquids changing into nematic liquid crystals in cooling) by Brillouin scattering. Wide-range and high-resolution spectroscopy was made. With this method, the power spectra of the light scattered in 6 CB (p-n-hexyl p'-Cyanobiphenil) was observed over the temperature range from 28.8 to 65.0 °C. Not only the fluctuation of density, which causes the Brillouin triplet, but also the fluctuation of orientation, a kind of the local order, appears in these liquids. The light-scattering spectrum reflects the reorientational relaxation. The model suggested by de Gennes theoretically predicts the spectra of both polarized (VV) and depolarized (VH) scattering. By fitting these equations to the experimental results, characteristic values were obtained which describe the dynamics of nematogens: Γ, the relaxation frequency of the orientational fluctuation, and C, the coupling constant between the orientational fluctuation and the local shear flow. The physical meanings of these parameters are discussed in the framework of critical phenomena.

4:30

4pPA11. Acoustic coherent backscattering in inhomogeneous media. K. Sakai, K. Yamamoto, and K. Takagi (Inst. of Industrial Sci., Univ. of Tokyo, 7-22-1, Roppongi, Minato-ku, Tokyo, 106 Japan)

Observed for the first time was strange behavior of an ultrasonic wave called "coherent backscattering" from inhomogeneous media. This effect is theoretically expected to occur as a result of a weak localization of waves propagating in a medium with many scatterers. It is a general nature of all kind of waves and has actually been confirmed for the optical waves. The ultrasonic wave incident to the dispersion system of small particles is scattered by the difference in the mechanical impedance in the system, whose intensity profile has a peak in the backscattering direction with respect to the incident wave. The peak height is expected to be twice as intensive as that of the diffuse scattering. The stroboscopic Schlieren image of the ultrasonic propagation scattered by the dispersion of polystyrene particles 100 μm in diameter was obtained and an intensive component going back the same way as the incident light was found. The quantitative measurement of the scattering profile was also carried out by a laser diffraction method and the result obtained shows good agreement with the theoretical prediction, giving a peak factor of 2. This phenomenon would be important when treating ultrasonic waves in inhomogeneous systems such as human organs.

4:45

4pPA12. Wave propagation at a water–chiral boundary. Shiuh-Kuang Yang and Shao-Yi Hsia (Dept. of Mech. Eng., Natl. Sun Yat-Sen Univ., Kaohsiung, Taiwan, Republic of China)

To model the dynamic response of a particles-mixture composite, the plane-wave propagation in an elastic matrix containing the structural chiral microstructures had been asymptotically investigated. In this so-called effective chiral (isotropic, noncentrosymmetric) composite, six independent wave numbers can be found from the dispersion equations. Two of the wave numbers represent the nondispersive longitudinal waves while the remaining four represent the dispersive circularly polarized shear waves. According to the dispersion equations, two cutoff frequencies divide the frequency response of the transverse wave numbers into three different groups and the four transverse modes can only be distinguished in a specified frequency range. In this study, the reflected and transmitted characteristics at a water–chiral interface are solved. To further illustrate the effects of the chirality, the reflected and transmitted fields at the water–isotropic and the water–micropolar interfaces are also solved. It is observed that, due to the mode conversion of the chirality and scattering resonances in the chiral medium, the chiral material should instigate a reducible reflected plane wave, and may be used as an anechoic coating to "absorb" sound underwater.

5:00

4pPA13. Theoretical and experimental investigation for LFB acoustic microscopy characterization of proton-exchanged $LiTaO_3$ optical waveguides. Masahito Miyashita and Jun-ichi Kushibiki (Dept. of Elec. Eng., Tohoku Univ., Sendai, 980-77 Japan)

Line-focus-beam (LFB) acoustic microscopy has been recognized as a new analytical technique of quantitative material characterization, by measuring the phase velocities of leaky surface acoustic waves (LSAWs) excited on the water-loaded specimen surface. Recently, application of this ultrasonic technique has been demonstrated successfully for characterization of proton-exchanged layers for $LiNbO_3$ and $LiTaO_3$ optical waveguides and for evaluation of the device fabrication processes. It is necessary and important to calculate LSAW propagation characteristics, viz., velocity and attenuation, for interpretation of experimental results and for evaluation of physical properties, such as optical and elastic properties, in proton-exchanged layers. In this paper, LSAW propagation characteristics for the structural model of water/piezoelectric layer/piezoelectric substrate are analyzed. The propagation characteristics are interpreted with calculations of spatial distributions of mechanical displacements and electric potential. Specimens of Z-cut $LiTaO_3$ substrates, with a layer thickness of 0.5 μm, were prepared. Experimental data of LSAW propagation characteristics, measured in the frequency range 100 to 300 MHz, are discussed in detail comparing with the calculated results.

5:15

4pPA14. Elastic properties of single- and multidomain crystals of $LiTaO_3$. Izumi Takanaga and Jun-ichi Kushibiki (Dept. of Elec. Eng., Tohoku Univ., Sendai, 980-77 Japan)

Ferroelectric single crystals of $LiNbO_3$ and $LiTaO_3$ are widely used as substrates for ultrasonic and optoelectronic devices. Obtaining homogeneity in elastic and optical properties of the crystals is one of the most important problems for wafer production. Recently, inhomogeneities of commercial wafers/crystals have been investigated with data of velocity changes obtained by line-focus-beam acoustic microscopy. The variations of the elastic properties are considered to be due primarily to chemical composition changes and secondarily to residual multidomains. In this paper, a fundamental study of the differences in elastic properties between single- and multidomain crystals of $LiTaO_3$ is conducted by measuring the longitudinal velocities for X-, Y-, and Z-cut specimens and the shear velocities for Z-cut specimens. Two sets of X-, Y-, and Z-cut specimens are prepared: One is for normal (single-domain) specimens and the other is for perfectly depoled (multidomain) specimens. For the longitudinal wave propagation directions along the Y and Z axes, the velocities for the single-domain specimens, due to the piezoelectricity, are 91.57 m/s (1.59%) and

92.70 m/s (1.50%) greater than those for the multidomain specimens, respectively. On the other hand, for the X-axis longitudinal and Z-axis shear wave propagation directions, the piezoelectricity is uncoupled, but very interesting velocity changes are found in that the longitudinal and shear velocities for the single-domain specimens are 48.45 m/s (0.87%) and 135.59 m/s (3.79%) lesser than those for the multidomain specimens, respectively.

5:30

4pPA15. Sintering temperature dependence of the photoacoustic spectra for ceramic CdS and Cd $S_{0.5}Se_{0.5}$. T. Toyoda, N. Aikawa, and K. Shinoyama (Dept. of Appl. Phys. and Chemistry, Univ. of Electro-Commun., 1-5-1 Chofugaoka, Chofu, Tokyo, 182 Japan)

CdS and CdSe are compound semiconductors which have bandgaps of 2.52 and 1.75 eV, respectively. The bandgap of mixed crystals such as $CdS_{1-x}Se_x$ $(0\leqslant x\leqslant 1)$ is proportional to x. Reliable characterizations of them have become important for applications in solar cell technologies. Although several investigations of the single crystals have been carried out, optical absorption measurements of ceramic CdS and mixed crystals have been a difficult problem. The purpose of this report is to show the results of photoacoustic spectra for ceramic CdS and $CdS_{0.5}Se_{0.5}$ for different sintering temperatures. Ceramic plates were obtained by sintering over the temperature range from 600 to 950 °C in N_2 gas. The body of the cell was an aluminum cylinder with a small channel in its periphery in which the microphone was inserted. Modulation frequencies for the measurements were 33, 133, and 257 Hz and the measurements were carried out in the wavelength range of 330–1200 nm. The signal intensity and phase spectra shift toward the high-energy regions and the slope of the absorption edge becomes steeper as the sintering temperature is increased. Moreover, the phase spectra show minima whose positions depend on sintering temperature, because of the increase of the recombination rate.

THURSDAY AFTERNOON, 5 DECEMBER 1996 MAUI ROOM, 2:00 TO 5:30 P.M.

Session 4pPP

Psychological and Physiological Acoustics: Physiological Acoustics

Richard R. Fay, Cochair
Psychology Department, Loyola University of Chicago, 6525 North Sheridan Road, Chicago, Illinois 60626

Hiroshi Wada, Cochair
Department of Mechanical Engineering, Tohoku University, Aoba Aramaki, Aoba-ku, Sendai, Miyagi, 980-77 Japan

Contributed Papers

2:00

4pPP1. All-pole representation of the cochlear input impedance, modal representation of the ear canal signal, and reflectance in the ear canal in the presence of spontaneous otoacoustic emissions. Carrick L. Talmadge, Arnold Tubis (Dept. of Phys., Purdue Univ., 1396 Phys. Bldg., West Lafayette, IN 47907), and Glenis R. Long (Purdue Univ., West Lafayette, IN 47907)

The cochlear input impedance associated with the linear active component of cochlear mechanics is represented as an all-pole function of frequency. This allows the dynamics of the ear canal signal to be expressed in terms of the behavior of coupled damped harmonic oscillators. Spontaneous otoacoustic emissions correspond to negative damping and, consequently, unstable oscillations. The introduction of a compressive nonlinearity, which stabilizes the spontaneous emissions, gives an interesting and computationally simple model for investigating the response of the auditory periphery in the case where there are one or more spontaneous emissions. The model is used, in particular, for studying the behavior of the reflectance in the ear canal. The magnitude of the reflectance will normally be less than one due to the entrainment of the emissions.

2:15

4pPP2. Dynamic behavior of the human middle ear: A finite-element method (FEM) analysis. Takuji Koike, Hiroshi Wada (Dept. of Mech. Eng., Tohoku Univ., Sendai, 980-77 Japan), Toshimitsu Kobayashi (Nagasaki Univ. School of Medicine, Nagasaki, Japan), and Tomonori Takasaka (Tohoku Univ. School of Medicine, Sendai, 980-77 Japan)

A few investigations of finite-element method (FEM) application to the analysis of the middle ear have already been reported [Funnell *et al.*, J. Acoust. Soc. Am. **81**, 1851–1859 (1987); K. R. Williams and T. H. J. Lesser, Br. J. Audiol. **24**, 319–327 (1990)]. However, in the models of these studies, ossicles and middle ear cavities were neglected, and their effects on the dynamic behavior of the middle ear have not been analyzed. In this study, first, applying the authors' own FEM program, a three-dimensional FEM model of a human auditory periphery was established. Then, vibration modes and transfer functions of the middle ear were obtained. The following conclusions can be drawn. (1) The vibration mode of the tympanic membrane and ossicles changes considerably due to the elastic ligament support of the ossicles. (2) This change enables the transfer function of the middle ear to keep flat. (3) The effect of the cavities on the dynamic behavior of the middle ear is large; the amplitude of the tympanic membrane vibration after opening the cavities is nearly two times larger than that before opening them. [Work supported by Research Fellowships of the Japan Society for the Promotion of Science for Young Scientists.]

2:30

4pPP3. Middle ear effect on otoacoustic emissions (OAEs): A finite-element method (FEM) analysis. Hiroshi Wada, Takuji Koike (Dept. of Mech. Eng., Tohoku Univ., Sendai, 980-77 Japan), Toshimitsu Kobayashi (Nagasaki Univ. School of Medicine, Nagasaki, Japan), Kenji Ohyama (Tohoku Rosai Hospital, Sendai, Japan), and Tomonori Takasaka (Tohoku Univ. School of Medicine, Sendai, 980-77 Japan)

Otoacoustic emissions (OAEs) are believed to originate from outer hair cell movement and are considered to be transmitted to the external auditory meatus through the middle ear in a retrograde fashion. Therefore, the effect of the middle ear on OAEs seems to be large. However, few investigations of middle ear effects on OAEs have been published. In this study, first, applying the authors' own finite-element method (FEM) program, a three-dimensional FEM model of a human auditory periphery, which included the external auditory meatus, the tympanic membrane, ossicles, and middle ear cavities, was established. Then, the retrograde and anterograde transmission factors of the normal and pathologic middle ear were examined. The following conclusions can be drawn. (1) As expected, the value of the retrograde transmission factor is lower than that of the anterograde one. (2)

This is mainly caused by the dynamic behavior of the tympanic membrane. (3) The levels of OAEs obtained from the patients such as atelectatic tympanic membrane, I–S joint separation, and otosclerosis are lower than the noise level of the OAE measurement system, because the value of the middle ear transmission factor of these patients is smaller than that of the normal subject.

2:45

4pPP4. The half-octave temporary threshold shift. David Alan Bies (Dept. of Mech. Eng., Univ. of Adelaide, South Australia 5005, Australia)

A model is proposed which provides a simple explanation for the noise-induced half-octave temporary threshold shift. The cochlear partition is modeled as a series of lightly coupled mechanical oscillators in which the outer hair cells are displacement responsive and the inner hair cells are velocity responsive. The outer hair cells are assumed to take an active role in providing negative damping in a region of stimulation of the cochlea at low-sound pressure levels. In the model, each oscillator is linear to very-high-sound-pressure levels except in a region of stimulation where the damping ratio varies with sound-pressure level. The model predicts that the passive viscous damping ratio of each oscillator is $1/\sqrt{2}$. At low-sound-pressure levels negative structural damping reduces the total damping ratio to about 0.055. At high-sound-pressure levels negative damping ceases and the response in a region of high-level stimulation is then passive. The variable damping ratio provides an explanation for all cases of nonlinear cochlear response which have been investigated.

3:00

4pPP5. Expression of basic fibroblast growth factor and transforming growth factor $\boldsymbol{\beta_1}$ in the inner ear of the human fetus. Xiangdong Chen, Mingmin Dong, Xuelui Dong, and Minsheng Dong (Dept. of Otolaryngol., Second Hospital of Henan Medical Univ., Zhengzhou, Henan, China, 450003)

The expression and localization of bFGF and TGF-β_1 in the inner ear of ten 10-29 week human fetuses were studied using the LSAB immunohistochemical technique referred to the method of Shi. The bFGF was strongly expressed in Corti's organs, vestibule macula and ampullar crista. The color of the upper cells of Corti's anlage was a little deeper than that of the basal cells. The hair cells and supporting cells in the matured Corti's organ continued to have distinct expression. Sites such as spiral ganglion (SG) neurons, stria vascularis (SV) had positive expression. Some cartilaginous cells and osseous cells in the developing osseous labyrinthus had light expression. TGF-β_1 was expressed in the epithelium of Corti's organ, vestibule macula, and ampullar crista. Hair cell color had a tendency to become weak with the maturity of the sensory epithelium, SG neurons, SV and spiral prominence (SP) had the TGF-β_1 expression too. The results indicated that bFGF and TGF-β_1 had participated in the course of inner ear development. The bFGF is a strong mitogen. However in this experiment it was found that it had the effect to promote differentiation. TGF-β_1 had double effect of promotion and inhibition on the proliferation and differentiation of cells. The expression of bFGF and TGF-β_1 in SG neurons indicated that both factors had participated in the development of the nervous system of the inner ear.

3:15

4pPP6. Eighth-nerve responses to sideband-attenuated and center frequency phase-shifted three-tone complexes. Michael F. Chronopoulos (Parmly Hearing Inst. and Dept. of Psych., Loyola Univ., 6525 N. Sheridan Rd., Chicago, IL 60626)

In a previous study [M. Chronopoulos and R. R. Fay, J. Acoust. Soc. Am. **99**, 2583–2584 (1996)], it was shown that goldfish (*Carassius auratus*) perception of three-tone complexes with either reduced modulation depths or a phase shift introduced at the center frequency (CF) is independent of both fine structure and power spectra of the signals. Behavioral responses were accounted for by a single function of the signals' normalized fourth moment of the envelope (NFME). The current experiment investigated the neurophysiological correlate of the goldfish's auditory nerve fiber's (saccular fiber's) responses to previously measured behavior. Signals included a 100% sinusiodally amplitude-modulated (SAM), a 350-Hz tone modulated at 30 Hz, and three-tone complexes with reduced modulation depths (MD). Reduced MD were achieved by either attenuating the sidebands (320 and 380 Hz) or phase shifting the CF component (350 Hz). Period histograms of single auditory fibers' responses were calculated. The histograms showed phase locking of the neural response to the envelope of the stimulus. The NFME of the histograms were measured. The NFME of the response histograms declined as a function of increasing sideband attenuation and increasing phase shift. Similar to behavioral measurements, NFME declined as a single function of the NFME of signals. [Work supported by a NIDCD Program Project Grant.]

3:30

4pPP7. Tuning in primary saccular afferents of the toadfish (*Opsanus tau*) revealed by REVCOR analysis. R. R. Fay (Parmly Hearing Inst., Chicago, IL and Marine Biological Lab., Woods Hole, MA 02543), P. Edds-Walton, and S. Highstein (Marine Biological Lab., Woods Hole, MA 02543)

Male toadfish vocalize (e.g., the ''boatwhistle'' -f_0 from 100 to 250 Hz) to attract females during mating season. This experiment investigates whether the frequency response of primary saccular afferents is adequate to encode these vocalizations. Whole-body oscillation (0.1–300 nm displacements) was delivered by a three-dimensional shaker system driven by noise with a flat spectrum between 50 and 1000 Hz. Stimuli were presented along three orthogonal axes (side–side, front–back, and up–down). Spikes were time stamped with 0.1-ms resolution. REVCOR functions were averaged accelerometer recordings of the noise triggered by spikes for 51.2 ms before and after each spike. This estimated the impulse response of the linear filtering that preceded spike generation. An FFT of the impulse responses estimated an afferent's filter shape. Impulse responses obtained were damped oscillations having a major peak indicating the magnitude and direction of excitatory acceleration. In general, filter shapes are low-pass within the frequency range investigated with corner frequencies varying between 120 and 350 Hz. These and other data [P. Edds-Walton and R. Fay, Biol. Bull. **189**, 211–212 (1995)] show that the saccule's response to acoustic particle motion can account for the reception of ''boatwhistle'' vocalizations. [Work supported by the NIH, NIDCD.]

3:45

4pPP8. Single unit recordings from directionally sensitive cells of the torus semicircularis of the goldfish (*Carassius auratus*). Diana Ma (Boston Univ. Marine Program, Woods Hole, MA 02543 and Parmly Hear. Inst., Chicago, IL) and Richard R. Fay (Boston Univ. Marine Program, Woods Hole, MA 02543 and Parmly Hearing Inst., and Loyola Univ., Chicago, IL 60626)

This experiment characterized the directional responses of cells of the auditory midbrain of goldfish. Single units in the midbrain were characterized by their response to whole-body vibratory stimuli of various directions and amplitudes. The animal's head was fixed in a rigid dish that was accelerated using two orthogonal pairs of shakers in the horizontal plane, and one shaker oriented vertically. Three orthogonally oriented accelerometers resolved the dish's motion in azimuth and elevation. Computer-synthesized sinusoids activated the shaker channels with relative phases and amplitudes required to produce linear motion along 12 axes equally spaced over 180° in each of the horizontal, midsagittal, and frontal planes. Spike rate was plotted as a function of stimulation axis angle in both Cartesian and polar coordinates. Most cells showed strongly directional response patterns with a primarily vertical orientation. Response patterns were classified with respect to the symmetry, sharpness, and the amplitude dependence of these directional features. Models were developed to give possible explanations for the observed responses. Many of the response shapes can be explained by combinations of simple peripheral effects (threshold and saturation) and inhibitory interaction. [Work supported by a Program Project Grant (NIH, NIDCD) to the Parmly Hearing Institute.]

4:00

4pPP9. Chemistry of the hamster cochlear nucleus. Donald A. Godfrey, Nikki L. Mikesell, Andrea B. Fulcomer, Timothy G. Godfrey (Dept. of Otolaryngol., Medical College of Ohio, P.O. Box 10008, Toledo, OH 43699), and James A. Kaltenbach (Wayne State Univ. School of Medicine, Detroit, MI 48201)

The chemistry of the hamster cochlear nucleus was studied by microdissection of freeze-dried sections combined with microassays. Tissue sample dry weights usually were between 0.1 and 0.5 μg. Amino acid concentrations were measured by high-performance liquid chromatography (HPLC) and activities of enzymes of acetylcholine metabolism by radiochemical procedures. Expressed per dry weight of sample, aspartate concentration was higher in the ventral than in the dorsal cochlear nucleus, whereas glutamate, glutamine, y-aminobutyrate (GABA), taurine, glycine, serine, and alanine concentrations were highest in the molecular layer of the dorsal cochlear nucleus. In general, amino acid concentrations were similar to those in cats and rats. Activity of choline acetyltransferase, the enzyme of synthesis for acetylcholine, was relatively high in granular regions and low in the molecular layer of the dorsal cochlear nucleus, a pattern similar to what has been found in rats and cats. The activity in the anteroventral cochlear nucleus was low compared to the value in rats, however. Activity of acetylcholinesterase, the enzyme of degradation for acetylcholine, was generally lower than in rats and cats, especially in granular regions and the molecular and fusiform soma layers of the dorsal cochlear nucleus. [Work supported by the American Tinnitus Association.]

4:15

4pPP10. Neuronal correlates of pitch in the inferior colliculus. Didier A. Depireux and Shihab A. Shamma (Inst. for Systems Res., Univ. of Maryland, College Park, MD 20742-3311)

Characteristics of single neuron responses underlying possible mechanisms for the extraction of the pitch of complex tones in the inferior colliculus (ICC) of anesthetized ferrets has been studied. Sounds such as amplitude-modulated tones, trains of clicks, and other spectrally complex sounds were used. It has been observed that the fine structure of a contralaterally presented stimulus can be found, perhaps even enhanced, in the firing patterns of collicular stellate cells. Ranges of latencies, best modulation frequencies, and center frequencies in the responses were found, as found before; more importantly perhaps, very short, stimulus-independent periodicities in the response patterns were found, as by Langner [G. Langner, Hear. Res. **60**, 115–142 (1992)]. The functional significance of these findings, and in particular of these ''intrinsic oscillations,'' with respect to the problem of pitch extraction by the auditory pathway, will be discussed in the context of a model. [Work supported by ONR, NIDCD, and NSF.]

4:30

4pPP11. On the sound environment for the right and left human hemispheric tasks. Kiminori Mouri (Graduate School of Sci. and Technol., Kobe Univ., Rokkodai, Nada, Kobe, 657 Japan and Central Res. and Development Headquarters, Itoki Crebio Corp., 4-12, Imafuku-higashi, 1-chome, Joto-ku, Osaka, 536 Japan) and Yoichi Ando (Kobe Univ., Kobe, 657 Japan)

Considering the human cerebral-hemispheric specialization, specific tasks can be performed under a certain sound environment without any interference effects. Under other conditions, strong interference effects occur between the task performance and the sound environment [Y. Ando, Acustica **64**, 110–116 (1987)]. For example, the left hemispheric tasks can be interrupted by the verbal sound, and the right hemispheric tasks can be interrupted by nonverbal sounds like noise. In order to identify hemispheric specialization for some important tasks, which are often performed in an office, the effective durations of the autocorrelation function of the α-brain wave range from both left and right hemispheres are analyzed here. Results may be utilized to design the recommended sound environment for each task in the office.

4:45

4pPP12. On the relationship between the autocorrelation function of continuous brain waves and the subjective preference of the sound field in change of the IACC. Kazuki Nishio and Yoichi Ando (Graduate School of Sci. and Technol., Kobe Univ., Rokkoudai, Nada, Kobe, 657 Japan)

This study analyzed the range of α waves (8–13 Hz) of continuous brain waves (CBW) and compared them with the scale value of the subjective preference of the sound field in a change of the interaural cross-correlation function (IACC) [Ando, *Concert Hall Acoustics* (Springer-Verlag, Heidelberg, 1985)]. The source signal was Sinfonietta, Opus 48; third movement, composed by Arnold. The scale value of the subjective preference and the CBWs to a paired stimuli, i.e., reference stimuli (IACC $=1.0$) and test stimuli (IACC$=0.3$ or 0.65), which were presented alternatively were compared. It is found that the value of the effective duration (τ_e) of the autocorrelation function (ACF) of the α waves from only the right hemisphere corresponds well to the subjective preference. It is considered, therefore, that τ_e is an objective measure reflecting the subjective preference [Chen and Ando (unpublished)].

5:00

4pPP13. Electrical stimulation of the cochlea improved word recognition under noises in tinnitus patients. Jun-Ichi Matsushima, Noboru Sakai (Dept. of Otolaryngol., School of Medicine, Hokkaido Univ., Kita 15, Nishi 7, Sapporo, 060 Japan), Shigeki Miyoshi, Norihiro Uemi, Masatsugu Sakajiri, Tohru Ifukube, and Tsutomu Kamada (Hokkaido Univ., Hokkaido, Japan)

Hearing impaired patients sometimes complain of poor understanding of speech under noise. It was shown that electrical stimulation of the cochlea in tinnitus patients improved their word recognition. The objective of this study is to show that electrical stimulation of the cochlea in tinnitus patients improved their word recognition under noises. Twenty four-segment sentences mixed with multiple talk recorded on a CD were delivered to tested ears at a comfortable level via a headphone. The signal-to-noise ratios were 0-, 5-, and 10-dB. Patients were requested to repeat what they heard. Improved score was calculated at each segment in 20 sentences. A sinusoidal wave of 10 kHz at the intensity of 200 μA was delivered to the cochlea. The stimulating Pt–Ir electrode was placed on the middle ear and the return electrode, which is made of a plate electrode for ECG, was placed on the skin behind the ear. Alternatively, an external electrical stimulation was applied to patients. Improved word recognition was shown in most patients with tinnitus relief following electrical stimulation of the cochlea, showing that electrical stimulation of the cochlea improved S/N in the auditory system.

5:15

4pPP14. The optimization of an implantable tinnitus suppressor using an extracochlear stimulation method. Masatsugu Sakajiri, Shigeki Miyoshi, Tohru Ifukube (Res. Inst. for Electron. Sci., Hokkaido Univ., Kita 12, Nishi 6, Kita-ku, Sapporo, 060 Japan), and Junichi Matsushima (Hokkaido Univ., Sapporo, 060 Japan)

Tinnitus patients have been treated using simple electrical stimulation to the promontory of the cochlea. With this treatment about 60% of the patients reported that the tinnitus was suppressed. This method, however, suppresses tinnitus for only a short period of time, so many tinnitus sufferers have long been waiting for an implant system which will be able to suppress tinnitus whenever it appears. For this reason, an implantable tinnitus suppressor has been developed using extracochlear stimulation technologies. The implantable tinnitus suppressor consists of a wave generator, a primary coil set behind the auricle, a secondary chip coil implanted in the mastoid cavity, a Pt–Ir electrode covered with a polyvinyl

alcoholic gel placed on the promontory, and a return electrode fixed to the subcutaneous tissue near the mastoid tip. The wave generated by the electric circuit of the wave generator is sent to the primary coil. The current that is induced in the secondary coil by using electromagnetic coupling is transmitted to the Pt–Ir electrode. The main purpose of this study is to optimize the function of this implantable tinnitus suppressor. [Work supported by the Ministry of Education, Science and Culture, Grant-in-Aid for Scientific Research (A)(2) No. 06402067.]

THURSDAY AFTERNOON, 5 DECEMBER 1996 LANAI ROOM, 1:30 TO 5:45 P.M.

Session 4pSC

Speech Communication: Robust Speech Recognition (Poster/Lecture Session)

Sadaoki Furui, Cochair
NTT Human Interface Laboratories, 3-9-11 Midori-cho, Musashino, Tokyo, 180 Japan

Abeer Alwan, Cochair
Department of Electrical Engineering, University of California, 405 Hilgard Avenue, Los Angeles, California 90095

All posters will be on display and all authors will be at their posters from 1:30 p.m. to 2:30 p.m. followed by lecture papers beginning at 2:30 p.m.

Contributed Poster Papers

4pSC1. Dynamic auditory representations and statistical speech recognition. Brian Strope and Abeer Alwan (Dept. of Elec. Eng., UCLA, 66-147E Eng. IV, 405 Hilgard Ave., Los Angeles, CA 90095)

The most common spectral estimation algorithm used for automatic speech recognition incorporates rough approximations of basic aspects of auditory modeling: frequency selectivity and magnitude compression. Attempting to improve the robustness and overall performance of ASR, researchers have proposed more sophisticated auditory models as the spectral estimation front end, with generally modest success at best. One common concern throughout these efforts is that the representation derived from an auditory model may not be a good match for typical statistical recognition algorithms. Recently, a dynamic auditory model that emphasizes changing local spectral peaks and has improved recognition robustness compared to other common front ends has been derived and implemented. The present work uses specific case examples to show how the perceptual representation leads to a softening of the resulting statistical models. The works also proposes a simple mechanism to adapt dynamic spectral features into a form more suitable for segmentally static statistical characterization. The mechanism is based on approximating the temporal derivative of the frequency position of local spectra peaks. The impact of our auditory model with this processing mechanism on robust recognition performance is discussed. [Work supported by NIH Grant No. 1 R29 DC 02033-01A1, and by NSF.]

4pSC2. A robust speech recognition algorithm. Kazuo Nakata and Khoji Matsumoto (Dept. of Electron., Chiba Inst. of Technol. 2-1-17 Tsudanuma, Narashino, Chiba, Japan)

The very important point of applications of speech recognition in a real field is its robustness against noise disturbances. A new robust algorithm is proposed, which has the following three features: (1) It uses only one microphone for input and noise-free speech as references; (2) it simulates a human hearing process in the following three functions: frequency analysis by the critical bandwidth, physical to sensory level transformation, and lateral inhibition; and (3) the speech is enhanced by passing a noisy one to a filter made of the spectrum envelope derived by the linear predictive analysis of the noisy speech itself. Outputs of the analyzing filter are vector-quantized and recognized by VQ and HMM composed of the noise-free reference speech processed in the same way as the noisy ones. The method can improve recognition scores from 15% to nearly 30% in the range of SNR of 15–10 dB for additive white random noise, compared to those of the noisy speech by HMM of noise-free speech.

4pSC3. Robust speech parameter based on a psychoacoustical rule-base model. Syunji Nishimura, Yoshifumi Chisaki, Tsuyoshi Usagawa, and Masanao Ebata (Dept. of Comput. Sci., Kumamoto Univ., 2-39-1 Kurokami Kumamoto, 860 Japan)

To realize a speech recognition system for our daily life, surrounding noise needs to be considered. Many studies have been done for parameters which are robust in a noisy environment. This paper presents a new parameter which is based on a psychoacoustical rule-base model. First the spectral local peaks of formants are calculated as a basic parameter. By performing this operation frame by frame, the parameter can be represented in a three-dimensional form like a spectrogram. The parameter is then modified by applying psychoacoustical rules to the sequence of peaks, corresponding to a so-called ''auditory stream'' in a speech. The parameter modification utilizes the characteristics of the parameter such as the synchronous onset or offset, the continuity of sequential peaks, and so on. The efficiency of the proposed parameter is evaluated on an isolated word speech recognition system with various SNR. As a result, the improvement due to the proposed parameter is at least 4 dB in SNR for white noise, in comparison with the performance of a LPC-based system.

4pSC4. Vowel identification from harmonic contours. John S. Antrobus (Dept. of Psych., City College of New York, New York, NY 10031) and Octavio Betancourt (City College of New York, New York, NY 10031)

A set of automatic algorithms expresses the acoustic vowel signal as the ratio series, $\log(f_j/F_0^{2/3})$, where f_j are the first 32 integer multiples of F_0, plus an additional 32 $\log(f_j/F_0^{2/3})$ delta terms that represent vowel trajectories. F_0 is measured on a window-by-window basis by an algorithm that eliminates all smearing due to conventional windowing algorithms. In order to reduce the dimensionality of this expression, the 64 terms are summarized by 11 + 11 terms from the cosine series. Using ten monosyllabic words spoken by 137 men, women, and children, vowel classification

is within one percent of human accuracy. Because the model uses none of the circular definitions of formant measurement and makes only one assumption that is unique to speech, namely, the $-\log F_0^{1/3}$ offset, the $\log(f_j/F_0^{2/3})$ contour is superior to a formant representation of voiced speech. Because a Euclidean classifier is as accurate as a quadratic discriminant function which uses more than ten times as many degrees of freedom, it is argued that the $\log(f_j/F_0^{2/3})$ transform may be accomplished by a genetically acquired neural mapping of the acoustic signal that facilitates the learning of vowel categories by infants.

4pSC5. Robust speech recognition using singular value decomposition based speech enhancement. B. T. Lilly and K. K. Paliwal (School of Microelectronic Eng., Griffith Univ., Brisbane, Queensland 4111, Australia)

Speech recognition systems work reasonably well in laboratory conditions, but their performance deteriorates drastically when they are deployed in practical situations where the speech is corrupted by additive noise distortion. One way to improve the performance of a speech recognition system in the presence of noise is to enhance the speech signal (and remove noise) prior to its recognition. Recently, a singular value decomposition based technique has been proposed for speech enhancement [Dendrinos *et al.*, Speech Commun. **10**, 45–57 (1991)]. In this technique, singular value decomposition was applied to an overdetermined, overextended data matrix formed from the noisy speech signal and a noise-free, low-rank approximation was obtained by retaining a specific number of singular values. This technique was applied here as a preprocessor for recognizing speech in the presence of noise and was found to improve the recognition performance significantly, especially at lower signal-to-noise ratios. [Work supported by the Australian Research Council.]

4pSC6. Effects of noise masking on the identification of fricative noise frequency. Diane L. Schoenburg and Virginia A. Mann (Dept. of Cognit. Sci., Univ. of California, Irvine, CA 92717)

This study examined how signal degradation affects the use of natural formant transitions and following vowel quality (rounded versus unrounded) in the identification of a fricative noise continuum varying from /s/ to /sh/. The stimuli were made by combining a synthetic /s/-/sh/ continuum with vocalic segments extracted from natural tokens of /sa/, /sha/, /su/, and /shu/. These were presented in a clear condition and with the vocalic segment masked by signal-correlated noise at 1.8 dB SNR. Listeners labeled the fricative consonant that began each stimulus as ''s'' or ''sh.'' In general, the results replicated previous findings that /s/ transitions and the presence of /u/ gave rise to more ''s'' responses than /sh/ transitions or the vowel /a/ [V. A. Mann and B. H. Repp, Percept. Psychophys. **28**, 213–228 (1980)]. It was further found that noise masking significantly decreased the effect of the transitions, but not the effect of vowel quality. Discussion of this interaction considers the effect of noise on (1) the salience of brief versus sustained speech cues and (2) the relative weighting of weak versus strong speech cues [Gordon *et al.*, Cognit. Psychol. **25**, 1–42 (1993)].

4pSC7. Word boundary detection under noise by using an adaptive noise canceler. M. Sakamoto and Y. Ohshima (Tokyo Res. Lab., IBM Japan Ltd., 1623-14 Shimotsuruma, Kanagawa, 242 Japan)

An adaptive noise canceling technique was applied to word boundary detection problems observed in a word recognition task under noise in a car. Many word boundary detectors previously proposed only made use of sample statistics of the short-term energy of incoming signal and noise, the operation of which is often found unreliable and prone to errors with the existence of radio talk and background music. In this study, an adaptive noise canceler was employed that was based on a projection algorithm, and it was integrated into the signal processing front end of a word recognizer as the pre-processing to its word boundary detection, in order to reduce deletion errors and to improve accuracy of detected boundary of the utterances by suppressing the effect of co-channel components. Also observations were made of the behavior of the integrated boundary detector in terms of the estimated profile of the short-term energy, and its effect on the recognition accuracy were analyzed.

4pSC8. Recognition of first words of isolated spoken word pairs with ambiguous word boundaries. Matthew J. Swiergosz (Dept. of Psych., Univ. of Nevada, Reno, NV 89557)

A series of experiments was conducted to test whether acoustic-phonetic information elucidates word boundaries. Word pairs were presented to subjects in gated format, where discrimination of the first word was marked by the first or last phoneme of the second word (e.g., RAN–PUMP or RAN–DUMP, respectively). Subjects were asked to provide either a written perceptual identification or use a numeric keypad to indicate the number of words at each gate (lexical enumeration). Results were consistent across response measures; perceptual identification of first words and lexical enumeration required less acoustic-phonetic information when first words were marked early in second words. In addition, the interval of silence between the word pairs was manipulated in two experiments to ascertain whether interword pause (30, 90, 150, or 210 ms) would result in differential perceptual identification and lexical enumeration. The interval of silence was not a reliable cue for first-word recognition or word counting and, thus is at odds with theories of auditory word recognition that incorporate pause as a factor influencing the activation levels of lexical candidates.

4pSC9. Phonetically balanced cepstrum mean normalization for telephone speech. Masatoshi Morishima, Toshihiro Isobe, and Nubuo Koizumi (NTT Data Commun. Systems Corp., 66-2 Horikawa-cho, Saiwai-ku, Kawasaki, 210 Japan)

Cepstrum mean normalization is an effective method for recognizing distorted telephone speech. This method compensates for the difference of bias on cepstrum coefficients (CC) between training data and test data by subtracting the mean value of the CC calculated from a certain amount of given speech data. Such adaptation data are not phonetically balanced, which makes it difficult to get an accurate mean value for the CC. In this paper, a new approach to resolve this problem is proposed. Before recognizing speech, not only the mean value of the adaptation data itself must be calculated, but also the mean value from Gaussian distribution of continuous density HMMs must be calculated, whose phonemes appear in the adaptation data. When recognition occurs, the difference of these two mean values is subtracted from speech data. This proposed method using various telephone speech data has been investigated, i.e., speech data from an ordinal analog telephone, a code-less handset telephone, and a digital cellular phone (Japanese full-rate digital phone based on VSELP), which was recorded through the public switched telephone network. These experimental results show an advantage of the new method.

4pSC10. Robust word recognition in an automobile by using cepstral normalization and an improved acoustic model. Y. Ohshima and M. Sakamoto (Tokyo Res. Lab., IBM Japan Ltd., 1623-14 Shimotsuruma, Kanagawa, 242 Japan)

Multiple approaches are needed to achieve robustness in speech recognition in an automobile due to various noise sources, relatively enclosed cabin acoustics, and the manner of speech of the driver under stress. A HMM-based word recognizer was developed for automobile applications, by integrating cepstral normalization into the front end and by training model parameters in multiple conditions according to a few driving scenarios mainly in city traffic. Speaker-dependent models were built and evaluated for a male and a female speaker using samples recorded in an automobile, and conducted a road test in city traffic. It was found that the system works well with several percentage points degradation in recognition accuracy compared with the laboratory experiments. The difference was attributed to the errors in word boundary detection.

4pSC11. Affine transformation-based feature normalization for speaker-independent speech recognition. Ping Luo and Kazuhiko Ozeki (Dept. of Comput. Sci. and Information Mathematics, Univ. of Electro-Commun., 1-5-1 Chofugaoka, Chofu, Tokyo, 182 Japan)

In statistical speech recognition, speaker-independent models are usually trained by using speech samples from a large number of speakers. Those models have a problem in that they have wider feature distributions and hence greater overlaps between different phones than adequately trained speaker-dependent models. In order to cope with the interspeaker variability, a method of speech feature normalization based on affine transformation has been presented [P. Luo and K. Ozeki, Tech. Rep. of IEICE, SP96-10 (1996)]. Prior to HMM training, feature vectors of each speaker are mapped to those of a reference speaker by an affine transformation estimated with a small amount of training data. The transformation, which is phone independent and speaker dependent, is also applied to feature vectors of unknown speakers in the recognition stage. It has been shown experimentally that this method is effective in reducing interspeaker variations in the cepstral domain. In this paper, further discussions about the performance and limitations of the method are given within the framework of continuous HMM. Practical issues related to the selection of an appropriate reference speaker will also be discussed.

4pSC12. Noisy speech recognition using HMM-based cepstral parameter generation and compensation. Takao Kobayashi, Takashi Masuko, Satoshi Imai (Precision and Intelligence Lab., Tokyo Inst. of Tech., 4259 Nagatsuta, Midori-ku, Yokohama, 226 Japan), and Keiichi Tokuda (Nagoya Inst. of Tech., Gokiso-cho, Shouwa-ku, Nagoya, 466 Japan)

This paper proposes a technique for adapting continuous density HMM speech recognition systems trained on clean speech data to make it robust to additive noise. The approach is based on cepstral parameter generation from speech and noise HMMs and parameter compensation using generated parameters of speech and noise. For clean speech and noise HMMs including cepstral and dynamic parameters, cepstral parameter vector sequences are generated in such a way that the probability of observing the parameter vector sequence from the given HMM is maximized using dynamic parameters. Then the generated cepstral vector sequences of speech and noise are combined to yield a noisy speech cepstral vector sequence. Compensated means of cepstral and dynamic parameters for noisy speech HMM are obtained from statistics of the noisy speech parameter sequences. Variances for noisy speech HMM are also estimated using the relationship between the clean and noisy speech parameter sequences. As a result, the technique provides cepstral and dynamic parameter compensation for noisy speech HMM. Moreover, it does not require the lognormal approximation used in cepstral parameter compensation based on the parallel model combination technique. Word recognition experiments based on phoneme HMMs show the effectiveness of the proposed technique.

4pSC13. Isolated word recognition in a noisy environment. Tadashi Suzuki, Yoshiharu Abe, and Kunio Nakajima (Mitsubishi Electric Corp., Information Technol. R&D Ctr., Human Media Technol. Dept., 5-1-1 Ofuna, Kamakura, Kanagawa, 247 Japan)

In noisy environment, performance of speech recognition systems trained in quiet environment is degraded. One of the reasons is acoustic phonetic modification caused by the Lombard effect, another is noise contamination of speech signal. This paper presents a new method for isolated word recognition in noisy environment. The method is based on two techniques. One of them is based on variability models for acoustic phonetic modification in Lombard speech, and another is to estimate additive noise spectrum frame by frame. The acoustic phonetic variability models represent the spectral difference between normal speech and Lombard speech. Each model is comprised of a nonlinear warping function on spectral domain and two spectral filters. The warping function represents formant shift. Two filters do the changes of formant bandwidth and of spectral tilt. The noise estimation is executed for each frame of noisy input with noise models and speech models made with clean speech data. These techniques were applied to speaker-dependent word recognition based on continuous density HMMs of subphoneme. Experimental evaluations were executed with the noisy Lombard speech data of 100 isolated words. From the experiments, the effectiveness of the proposed method has been confirmed.

4pSC14. Investigation of subband-cross-correlation analysis for speech recognition under noisy conditions. Shoji Kajita, Kazuya Takeda, and Fumitada Itakura (Graduate School of Eng., Nagoya Univ., Furo-cho 1, Chikusa-ku, Nagoya, 464-01 Japan)

This paper describes subband-cross-correlation (SBXCOR) analysis for robust speech recognition under noisy conditions. The SBXCOR analysis is an extended signal processing technique of subband-autocorrelation (SBCOR) analysis that extracts periodicities present in speech signals. In this paper, the performance of SBXCOR is investigated using a DTW word recognizer, under simulated acoustic conditions on a computer and a real environmental condition. Under the simulated condition, it is assumed that speech signals in each channel are perfectly synchronized while noises are not correlated. Consequently, the effective signal-to-noise ratio of the signal generated by simply summing the two signals is raised about 3 dB. In such an ideal case, it is shown that SBXCOR is less robust than SBCOR extracted from the two-channel-summed signal, but more robust than the conventional one-channel SBCOR. The resultant performance was much better than that of smoothed group delay spectrum and mel-frequency cepstral coefficient. In a real computer room, it is shown that SBXCOR is more robust than the two-channel-summed SBCOR. The microphone setup for SBXCOR will be also discussed.

4pSC15. Speech recognition based on subword units. Takuya Noizumi, Mikio Mori, and Shuji Taniguchi (Dept. of Information Sci., Fukui Univ., 3-9-1 Bunkyo, Fukui, 910 Japan)

Large vocabulary, isolated word recognition requires a large amount of training data proportional to the vocabulary size to characterize each individual word model. A subword-unit-based approach is a more viable alternative than the word-based approach to overcome the problem of the training data size, since different words can share common segments in their representations in the former. This paper deals with a couple of isolated word recognition systems where the subword-unit-based approach is commonly employed, though their methods of segmentation are completely different. In one system a hidden Markov model is used to decompose a word into subword units (segments), and frequency spectra of those subword units are fed to a recurrent neural network to yield a subword code sequence for the word. This sequence is then recognized hopefully as the original word by a set of hidden Markov models for isolated words. In the other system subword boundaries within a word are detected by finding peaks of the delta cepstrum of the word, and the resulting sequence of subwords is deciphered into the original word by means of concatenated hidden Markov models of isolated words. Those systems attain average recognition accuracies over 92%–96%.

4pSC16. Comparison of the effect of homophones on speechreading between Japanese and English. Shizuo Hiki (School of Human Sci., Waseda Univ., 2-579-15 Mikajima, Tokorozawa, 359 Japan) and Emiko Kamikubo (Waseda Univ., Tokorozawa, 359 Japan)

The speech information conveyed by the shape of the mouth differs according to the characteristic phonetic system of each language. Besides the phonetic characteristic, a peculiar feature found in Japanese speech is

that there are many homophenous words (words having a different sound but the same mouth shape), because the Japanese vocabulary comprises a large number of homophones (words having a different sound but the same mouth shape), compared with English. This results in a decrease of information obtainable through speechreading. In order to ascertain this feature quantitatively, first, the mouth shapes of vowels and consonants of Japanese speech are described symbolically. Detailed symbols are assigned to as many kinds of mouth shapes as are readable in a clear utterance. Using these symbols, the changes in mouth shapes in Japanese speech are described based on coarticulation rules and a combination of preceding and following vowels. Then, the ratios of homophenous words and homophones found in a basic vocabulary are estimated by referring to statistical data of the syllabic structure of Japanese words. By assembling these ratios, the nature of the information conveyed through speechreading is analyzed, and the negative effect of homophones on the speechreading of Japanese is discussed.

2:30–2:40 Break

Chair's Introduction—2:40

Contributed Lecture Papers

2:45

4pSC17. In search of an invariant representation for speech: The modulation spectrogram. Steven Greenberg and Brian E. Kingsbury (Univ. of California, Berkeley and Intl. Comput. Sci. Inst., 1947 Center St., Berkeley, CA 94704)

The ability to understand speech spoken by diverse speakers under a wide range of acoustic environmental conditions presents a central challenge to current theories of speech perception. Reverberation, background noise, and speaker variation all contribute to the heterogeneous acoustic realization of phonetic information from which listeners routinely derive linguistic information. Representations based on the detailed spectro-temporal properties of speech (e.g., the sound spectrogram) are not sufficiently stable under such conditions as to provide a reasonable account of the processes by which the brain decodes spoken language. A new representational format for visualizing speech, based on the distribution of the modulation spectrum below 16 Hz (and which emphasizes modulation frequencies centered around 4 Hz) across critical-bandlike channels, provides a high degree of stability under reverberant and low signal-to-noise-ratio conditions. This modulation-based representation appears to capture many essential properties of neurons in the primary auditory cortex, and is also well matched to the temporal dynamics of speech production. As such, it may provide a more principled basis for understanding the mechanisms by which human listeners decode the speech signal under natural acoustic conditions than finer-grained spectro-temporal representations.

3:00

4pSC18. Phoneme recognition using reference patterns constructed with discriminative training and DP matching. Yoshiyuki Okimoto (Graduate School of Information Sci., Tohoku Univ., Sendai, 980-77 Japan) and Shozo Makino (Tohoku Univ., Sendai, Japan)

A modified LVQ2 algorithm (MLVQ2) for phoneme recognition is proposed. The length of a phoneme reference pattern made by the algorithm was fixed. It showed high phoneme recognition performances for isolated spoken words. However, it did not show high performances for continuous speech because the phoneme reference patterns with fixed length could not deal with a big change in phoneme duration. In this paper, a new construction algorithm of the reference patterns using discriminative training based on the MLVQ2 and DP matching is proposed. The initial multiple reference patterns are optimized as follows: At first, the correspondence between a frame of the input sample and a frame of the nearest reference pattern in the incorrect phoneme class is calculated by using the DP matching. Then, the corresponding frame vector of the reference pattern in the incorrect phoneme class is moved away from the frame vector of the input sample based on the MLVQ2. On the other hand, the corresponding frame vector of the reference pattern of the correct phoneme class is moved nearer to the frame vector of the input sample. The experimental result using the ASJ 503 sentences speech database showed a 81.3% phoneme recognition rate, increasing 1.7% compared to the previous method.

Invited Papers

3:15

4pSC19. Feature and model compensation for robust speech recognition. Chin-Hui Lee (Bell Labs., 600 Mountain Ave., Murray Hill, NJ 07974)

A mathematical framework based on *maximum likelihood stochastic matching* is proposed to perform *feature* and *model* compensation for robust speech recognition. Speech recognition is often formulated as a matching problem between the feature vectors extracted from a test utterance and a set of speech models or patterns obtained from some training corpra. It is well known that a speech recognizer often degrades in performance when the testing data are not *acoustically similar* to the training data. One way to improve is to find features that are *invariant* under all acoustic conditions and distortions. Some form of compensation is often required. The proposed stochastic matching approach assumes a structure or a form of the feature and/or model transformations. Together with a set of *nuisance parameters*, the transformations approximate the distortion in the test utterance. To decrease the *acoustic mismatch* between a test utterance and a given set of speech models, e.g., hidden Markov models, the stochastic matching algorithm estimates the nuisance parameters and then applies the feature/model transformations during speech recognition. Simple channel distortion can be approximated with linear transformations. For more complicated distortions, such as environmental, speaker, and combined mismatches, nonlinear compensating transformations are needed. These compensations give a significant performance improvement in speech recognition over the systems without them when utterances are affected by additive ambient noises and convolutional channel distortions.

3:30

4pSC20. Beyond a "short-term" analysis of speech. Hynek Hermansky (Oregon Graduate Inst. of Sci. and Technol., Beaverton, OR 97006)

A short-term analysis of speech, inherited from speech coding, describes a time-varying speech signal as a sequence of vectors, each vector reflecting properties of a relatively short (10–20 ms) segment of the signal. Each analysis vector provides a single sample from the underlying time-varying speech process. Typically, the whole feature vector is used as one entity for a subsequent processing. For use in ASR, the short-term analysis may have some deficiencies. (1) From a single vector alone, there is no way to differentiate between components with different rates of change. (2) When one or a few components of the vector get corrupted, the subsequent ASR result may be corrupted. This is inconsistent with human speech perception where (1) many auditory phenomena seem to span at least a length of a syllable, and that (2) decoding of a linguistic message does not have to be severely impaired by a partial degradation of the speech signal. Emerging ASR techniques are discussed which attempt to alleviate the above-mentioned deficiencies of the short-term analysis by (1) doing temporal processing on trajectories of short-term features of speech, and (2) selectively merging information from several subsets of the short-term representation of speech. [Work supported by DoD (MDA-904-94-C-6169) and NSF/ARPA (IRI-9314959).]

3:45

4pSC21. Evaluation of a rapid environment adaptation algorithm in adverse environments. Hiroaki Hattori, Keizaburo Takagi, and Takao Watanabe (Information Technol. Res. Labs., NEC Corp., 4-1-1, Miyazaki, Miyamae-ku, Kawasaki, Kanagawa, 216 Japan)

This paper reports results for a rapid environment adaptation algorithm in adverse car environments. It is known that the difference between training and testing environments degrades speech recognition performance. This degradation becomes serious especially in applications such as telephone speech recognition and speech recognition inside a running vehicle, where the testing environment may drastically change. To solve this problem, a rapid environmental adaptation method (hereafter referred to as REALISE) is proposed and its performance is measured in telephone speech recognition. REALISE estimates the differences in multiplicative and additional noise in the spectral domain between the training and testing environments and uses them to adapt acoustic features of reference patterns to the testing environment. Utterances in a car were also recorded to investigate the performance of REALISE under more severe conditions. The testing data were 100 city names uttered by three males and three females under three running conditions (idling, 50 kph, and 100 kph). Whereas the average recognition rate achieved by a conventional spectral subtraction technique was 75.6%, REALISE achieved 85.7%. This result proves the effectiveness of REALISE in a very noisy vehicle environment.

4:00

4pSC22. Compensation for speech recognition in degraded acoustical environments. Richard M. Stern, Pedro J. Moreno, and Bhiksha Raj (Dept. of Elec. and Comput. Eng. and School of Comput. Sci., Carnegie Mellon Univ., Pittsburgh, PA 15213)

The accuracy of speech recognition systems degrades when operated in adverse acoustical environments. This paper discusses two ways in which more detailed mathematical descriptions of the effects of environmental degradation can improve speech recognition accuracy using both "data-driven" and "model-based" compensation strategies. Data-driven methods learn environmental characteristics through direct comparisons of speech recorded in the noisy environment with the same speech recorded under optimal conditions. Model-based methods use a mathematical model of the environment and attempt to use samples of the degraded speech to estimate model parameters. Two approaches to data-driven compensation, RATZ and STAR, are described, as well as a new approach to model-based compensation, referred to as the vector Taylor series (VTS) algorithm. Compensation algorithms are evaluated in a series of experiments measuring recognition accuracy for speech from the ARPA Wall Street Journal database that is corrupted by artificially added noise at various signal-to-noise ratios (SNRs). For any particular SNR, the greatest recognition accuracy obtained using a practical compensation algorithm is observed when that system is trained using noisy data at that SNR. The RATZ, VTS, and STAR algorithms achieve this bound at global SNRs as low as 15, 10, and 5 dB, respectively. [Work supported by ARPA.]

4:15

4pSC23. Hands-free speech recognition by a microphone array and HMM composition. Satoshi Nakamura, Takesi Yamada, Tetsuya Takiguchi, and Kiyohiro Shikano (Graduate School of Information Sci., Nara Inst. of Sci. and Technol., 8916-5 Takayama Ikoma, Nara, 630-01 Japan)

Hands-free speech interface is one of the final goals of human–machine interface. This paper introduces two methods for distant talking speech recognition in noisy and reverberant rooms. The first method is speech recognition using a microphone array. The microphone array enables one to enhance a speech signal using spatial phase differences even in environments where unstationary noises exist. The proposed method is composed of two modules: (1) a high SNR signal retrieval by a delay-and-sum beamformer and (2) localization and trace of the speaker's direction by extracting a signal power and pitch harmonics. The second method is speech recognition based on the HMM composition. The proposed HMM composition is obtained by extending the HMM composition method of an additive noise to that of the convolutional acoustical transfer function. The HMMs are prepared beforehand for clean speech, noise, and acoustical transfer function. Then the HMM composition is conducted twice for a speech HMM and an acoustical transfer function HMM in the cepstrum domain and for the distorted speech HMMs and noise HMM in a linear spectral domain. The speaker-dependent/independent word recognition experiments using tied-mixture monophone HMMs are carried out and have clarified the effectiveness of the proposed methods. Furthermore, an effective coupling of these methods is also discussed.

4:30

4pSC24. Multisensor-based speech processing for robust speech recognition. Y. Zhao, K. Yen (Beckman Inst. and Dept. of ECE, Univ. of Illinois, 405 N. Mathews Ave., Urbana, IL 61801), and X. Zhuang (Univ. of Missouri, Columbia, MO 65211)

A multichannel signal-processing technique is integrated with automatic speech recognition for dealing with time-varying interference signals. Two microphones are employed for signal acquisition, each picking up a convolutive mixture of two source signals. A decorrelation adaptive filtering [Weinstein *et al.*, IEEE Trans. SAP 405–413 (1993)] is first performed to restore the source signals; a cross-spectral analysis is next performed on the restored signals to determine the presence regions of each source signal [K. Yen and Y. Zhao, Proc. ICSLP (1996)]. The restored speech signals within the presence regions are inputted to an HMM-based speaker-independent continuous speech recognition system [Y. Zhao, IEEE Trans. SAP, 345–361 (1993)]. Five test sets were constructed from the TIMIT database under the SNR conditions of 20, 10, 0, −10, and −20 dB, each consisting of 78 sentence pairs. The acoustic coupling channels were simulated by FIR filters. The recognition vocabulary size was 853 and the task perplexity was 105. It was found that the multichannel integrated recognition system significantly improved recognition performance; above −10 dB, the word accuracies were close to that of the interference-free condition (91% word accuracy). [Work supported by NSF IRI-95-02074.]

4:45

4pSC25. Model-based unsupervised instantaneous speaker adaptation. Sadaoki Furui and Tomoko Matsui (Furui Res. Lab., NTT Human Interface Labs., 3-9-11, Midori-cho, Musashino-shi, Tokyo, 180 Japan)

Unsupervised instantaneous adaptation, which uses the input utterance itself for adaptation, is the ideal speaker adaptation method for speech recognition, and is expected to be very useful for a wide range of applications. Since voice individuality is phoneme dependent, speaker adaptation must be performed model dependently. However, it is impossible to obtain a complete model sequence, that is, what is spoken, for each input utterance, especially for speakers who have many recognition errors when using speaker-independent models. Therefore, how to perform model-dependent adaptation without knowing the correct model sequence is a crucial issue. If all possible model sequences were hypothesized and used for adaptation, the amount of calculation would become enormous. This paper proposes a new adaptation method, in which N-best hypotheses are created by applying speaker-independent phone models to each input utterance, and speaker adaptation based on a constrained MAP estimation technique is then applied to each hypothesis. Using this method, the likelihood of a correct hypothesis existing in a low rank with speaker-independent models rises, and, as a result, recognition accuracy increases. Experimental results for several continuous speech recognition tasks show that recognition accuracy is increased by this method, even for speakers who have very low accuracy with speaker-independent models.

5:00

4pSC26. Fast speaker-independent acoustic modeling and speaker adaptation. Shoichi Matsunaga (NTT Human Interface Lab., 1-2356, Take, Yokosuka, Kanagawa, 238-03 Japan), Masahiro Tonomura (ATR-ITL, Kyoto, 619-02 Japan), and Tetsuo Kosaka (Canon Media Technol. Labs., Kanagawa, 211 Japan)

A fast acoustic modeling method for speaker-independent speech recognition and a speaker-adaptation method, which is effective even with only a small amount of speech data, is described. The speaker-independent phoneme models are generated by composing representative speaker-dependent phoneme models, which are selected from among all speaker-dependent models by clustering the models without Baum–Welch parameter re-estimation. This generation method greatly reduces the computational cost needed to create the speaker-independent HMMs to much less than that of the Baum–Welch method, i.e., by a factor between approximately 1/20 and 1/50. This speaker adaptation algorithm unifies two conventional techniques, i.e., a maximum *a posteriori* (MAP) estimation and transfer vector field smoothing. *A priori* knowledge from initial models is statistically combined with *a posteriori* knowledge derived from the adaptation data to complement the sparse adaptation data. Transfer vector smoothing is used to interpolate the untrained parameters. Furthermore, in order to obtain a suitable *a priori* knowledge concerning speaker characteristics, a speaker-clustering model, generated by using speech of a selected speaker cluster, is used as an initial model. The cluster selection is performed with a tree-structured speaker clustering technique that determines the number of speakers and the members in the cluster based on speaker similarity. [Work supported by ART-ITL.]

5:15–5:45 Discussion

Session 4pSP

Signal Processing in Acoustics: Acoustical Imaging

Hua Lee, Cochair
Department of Electrical Engineering, University of California, Santa Barbara, California 93106-2991

Sadayuki Ueha, Cochair
Precision and Intelligence Laboratory, Tokyo Institute of Technology,
4259 Nagatsuda, Midori-ku, Yokohama, Kanagawa, 226 Japan

Chair's Introduction—2:00

Invited Papers

2:05

4pSP1. Acoustical imaging from the perspective of the acoustical imaging symposia. John P. Powers (Naval Postgrad. School, Monterey, CA 93943)

The Acoustical Imaging Symposium series began in 1968. The 23rd meeting of the series will be held in Boston in April 1997. The symposium has focused on the engineering and physical-science aspects of acoustic imaging. This presentation will survey a selected number of topics presented at these symposia, including acoustical holography techniques, phase-only imaging, imaging with backward and forward propagation techniques, acoustic microscopy imaging, underwater imaging, seismic imaging, and other imaging techniques.

2:20

4pSP2. Acoustic imaging by backward propagation: An overview of algorithm structure and configurations. Hua Lee (Univ. of California, Santa Barbara, CA 93106-9560)

The backward propagation technique is known as one of the most widely used algorithms in acoustic imaging for its simplicity, stability, and resolving capability. This presentation provides an overview of the development of backward propagation based algorithms for various system configurations in a systematic manner, from the simple case of coherent systems with stationary sources to the more complex wideband pulse-echo systems for time-varying objects, in both monostatic and bistatic modes. With this treatment, it is now possible to establish an accurate assessment of systems' performance in terms of computation complexity, algorithm structure, sensitivity, and limitations.

2:35

4pSP3. Medical ultrasonic imaging: Present status and future prospects. Joie P. Jones (Univ. of California, Irvine, CA)

For over 30 years, ultrasonic imaging, based on the same pulse-echo principles as radar and sonar, has played an important role in diagnostic medicine. In many ways ultrasound is an ideal diagnostic tool—noninvasive, nontraumatic, capable of producing real-time images, and as all available data indicate, apparently safe at the acoustical intensities and duty cycles encountered in existing diagnostic equipment. Here the objective is to provide a snapshot of the present state of medical ultrasound and to assess the future prospects of this technology from ongoing research activities. Innovations such as Doppler analysis and cross-correlation techniques to measure flow, phased array transducers, contrast agents, tissue characterization, and, in particular, quantitative imaging methods based on unique reconstruction techniques will, it is believed, greatly increase the clinical utility of ultrasound as well as maintain a continued high rate of growth in the marketplace.

2:50

4pSP4. Tomographic reconstruction techniques for pulse-echo imaging. I. Akiyama (Dept. of Elect. Eng., Shonan Inst. of Technol., Fujisama, Japan)

Imaging is possible based on tomographic reconstruction techniques such as computed tomography, although ultrasonic pulse echo imaging is based on focusing techniques. It is possible to obtain three-dimensional imaging data by the tomographic method. When divergent waves are insonified toward a target by a part of a spherical surface, backscattered waves which are generated at the target are received by the transducer, and the echo signal on a delayed time is expressed as the integral of scattered waves at the surface of the radius proportional to the delay. If the transducer has both convex and concave parts of the spherical surface, both divergent waves and convergent waves can be generated. These waves are focused onto the arc on the spherical surface; thus the amplitude at the delay of the received echoes is expressed as the product of the backscatter coefficient on the arc and the diffraction term. Those values are equivalent to the projection data as the tomographic reconstruction technique. Therefore the distributions of the backscatter coefficient on the spherical surface are reconstructed from the projection of the echoes. In this study the effects of the angle dependency of the amplitude on the spherical surface due to the diffraction term are discussed and then the method for correction is described.

3:05

4pSP5. An ultrasonic ring transducer system for studies of scattering and imaging. Robert C. Waag, Dong-Lai Liu, T. Douglas Mast, Adrian I. Nachman (Univ. of Rochester, Rochester, NY 14627), Paul Jaeger (Peak Design, Inc., Arizona), and Tadashi Kojima (Nihon Dempa Kogyo Co., Japan)

A novel ultrasonic ring transducer and special control electronics have been developed for scattering and imaging studies. The transducer contains 2048 rectangular elements with a center frequency of 2.4 MHz and a -6-dB bandwidth of 70%. At the center frequency, the element size is 0.29 wavelength $\times 40$ wavelength and the spacing is 0.37 wavelength. A multiplexer provides access to any contiguous 128 elements for transmission and any contiguous 16 elements for simultaneous reception. The transmit electronics have independently programmable waveforms. The receive electronics have time-varied gain functions independently programmable over the range 15–55 dB. Each receive channel includes a 20-MHz, 12-bit A/D converter. The electronics permit synthesis of arbitrary transmit and receive apertures. A novel ultrasonic wavefront design method has been implemented to determine element excitations using backpropagation of a user-specified field pattern. Pulse-echo compound images using constant $f/1.0$ transmit and receive apertures have been obtained for model scattering objects and an anthropomorphic breast phantom. Scattering measurements have been analyzed to obtain frequency- and angle-dependent average differential scattering cross sections of random media. The system is a useful facility for measurements of ultrasonic scattering for characterization of tissue, development of adaptive beam-formation techniques, and implementation of quantitative image reconstruction methods.

3:20

4pSP6. Ultrasonic mapping of blood flow through the microvasculature. K. Ferrarra (Dept. of Mech. Eng., Univ. of Virginia, Charlottesville, VA 22903-2442)

An overview of the use of high-frequency ultrasound to map blood flow through the microvasculature will be presented. The correlation of scattered echoes from blood at frequencies from 20 to 100 MHz are shown to demonstrate a smaller probability of error than the correlation at a lower transducer frequency. Experimental three-dimensional (3-D) maps of the velocity profile within vessels with a diameter of 100 μm or less will be presented. Blood velocity within vessels as small as 40 μm can be mapped using a transducer with a nominal center frequency of 50 MHz. Estimation of flow within these small vessels at the appropriate low velocities is challenging due to the small signal levels, the effect of cardiac and respiratory motion, and the unknown rate of fluctuation of the received signal from the small number of red blood cells contained within these sample volumes. Reliable *in vivo* estimation of the velocity within the microvasculature requires realignment of the signal from a single line of sight, and this method is presented. Recognition of 3-D continuity of a vessel tree is required to eliminate spurious noise. The rate of fluctuation in the received signal from small blood vessels is evaluated as a function of the sample volume and vessel size. A minimum time window for reliable detection of flow through a small vessel is determined. Finally, the use of acoustic contrast agents to improve the visualization of flow within small vessels will be introduced.

3:35–3:50 Break

3:50

4pSP7. Broadband acoustic-field image enhancement in a noisy shallow ocean. James V. Candy (Lawrence Livermore Natl. Lab.) and E. J. Sullivan (Naval Undersea Warfare Ctr., Newport, RI 02841)

When a narrow-band source propagates in a shallow ocean environment, the ocean can be characterized by a waveguide resulting in a normal-mode solution. When the source is broadband, its spectrum can be decomposed into a set of distinct narrow-band lines, each supporting a distinct set of normal modes differing from frequency to frequency by both the number of modes and their respective wave numbers. Couple this propagation problem to a set of noisy hydrophone measurements obtained from a vertical array of sensors spanning the water column and a formidable problem in estimating the broadband acoustic field results. The field itself is essentially a space–frequency and once this space–frequency field can be estimated from the noisy measurements, then various postprocessors can be applied to estimate, detect, and localize the underlying source. Thus it is important, in the broadband case, to extract and enhance the broadband acoustic-field image. In this paper the development of a model-based image enhancement technique is discussed which employs the broadband normal-mode solutions using an optimal estimation scheme to provide the required enhancement. A by-product of the outputs of the processor, which is a space-varying Kalman filter, is the ability to not only enhance the image, but also to extract and enhance the corresponding broadband modal functions (one for each frequency) as well.

4:05

4pSP8. Development of transducers for a holographic 3-D imaging sonar. Chiaki Ishihara and Osamu Takano (Akishima Labs., Mitsui Zosen, Inc., 1-50, Tsutsujigaoka 1-chome Akishima, Tokyo, 196 Japan)

This report describes the processes of development and the performances of the transducers applied to a real-time 3-D acoustical holographic imaging system based on the encoded wavefront technique. In this system, the acoustical wave modulated its phases by a system of Walsh functions, emitted from the transducers. Thus the performance of the broad bandwidth was needed for transducers to make clear the points of the phase modulation on the waveforms. Piezorubber film was selected as a piezoelectric material and it was laminated to improve the sensitivity of the transmitters. The receivers have a copolymer film (PVDF) as the piezoelectric material.

The thin film and a preamplifier mounted close to the film give superior performances to the receiver. It is made clear that the directivity of the transducers agrees with the results of theoretical analysis of the acoustic field generated by circular plane vibration; hemispherical shapes of the transducers distorted the waveform in some conditions; the broad bandwidth will be realized to set the resonance frequency lower than the operational frequency.

Contributed Papers

4:20

4pSP9. A comparison of quantitative imaging techniques for ceramic materials. D. J. Chinn, D. J. Schneberk, N. A. Del Grande, and G. H. Thomas (Lawrence Livermore Natl. Lab., Livermore, CA 94550)

This work compares the effectiveness of five different imaging methods for porous preforms made from ceramic fibers. The methods are acoustic attenuation, x-ray tomography, radiography (x-ray attenuation), optical attenuation, and dual-band infrared imaging. The preform properties evaluated are fiber concentration, cracks, voids, surface defects, and contamination. Preforms with varying fiber concentration, preform thickness, and fiber type have been examined. All the methods allow density variations to be mapped. X-ray tomography has the highest contrast and sensitivity to defects but is slow if the entire preform is scanned. The attenuation methods are orders of magnitude faster and less expensive, but average over the thickness in the viewing direction and have less sensitivity to voids and cracks. Of the line-of-sight attenuation methods, radiography has the highest resolution and contrast. Acoustic imaging is most sensitive to surface aberrations. [Work performed under auspices of the U.S. Department of Energy by the Lawrence Livermore National Laboratory under contract No. W-7405-ENG-48.]

4:35

4pSP10. Quasi-three-dimensional inverse scattering computerized tomography for high-precision sound velocity measurement in tissues. Akira Yamada (Graduate School of Bio-Applications & Systems Eng., Tokyo Univ. of Agriculture and Technol., Koganei-shi, Tokyo, 184 Japan)

A quasi-3-D inverse scattering technique is proposed to reconstruct sound velocity slices of the weak scattering tissues by transmitting and receiving acoustic waves around a single rotational axis. The reconstructing procedure proposed is as follows. First, a backpropagation operation is introduced using the holographic formula for a homogeneous medium. This backpropagation preprocessing operation contributes dramatically to the reduction of the final quantitative reconstruction errors. Next, after the Rytov transform over the marginal backpropagated plane, data are then filtered in the perpendicular direction to discriminate the horizontally scattered wave. Consequently, quasi-3-D sound velocity images on the horizontal sliced plane of an object can be obtained, using the conventional 2-D inverse scattering reconstruction procedure. Precision of the proposed 3-D reconstructed sound velocity image is investigated using the calculated scattered data from the spherical object. As a result, it is demonstrated that quantitative precision of the reconstructed 3-D velocity image is verified to be within 1% over the range of the velocity inhomogeneity from −10% to 10%. The results obtained are promising for actual clinical applications such as the human breast cancer screening test.

4:50

4pSP11. Two-step holographic reconstruction of an acoustic field. Makoto Tabei and Mitsuhiro Ueda (Dept. of Intl. Development Eng., Tokyo Inst. of Technol. 2-12-1, O-okayama, Meguro-ku, Tokyo, 152 Japan)

A method for the holographic reconstruction of an acoustic field from a finite sound source is discussed in detail. It is often experienced that the holographic reconstruction based on the angular spectral method gives unexpectedly erroneous reconstruction. This is due to the fact that the singularity of the transfer function in the spatial frequency domain cannot be treated accurately by the discrete system. [Waag *et al.*, "Cross-sectional measurements and extrapolations of ultrasonic fields," IEEE Trans. Sonics Ultrason. **SU- 32**, 26–35 (1985)]. In this paper, a reconstruction is proposed that is based on the convolution of observed data and the transfer function rather than the direct manipulation of data in the frequency domain. It is shown that when the observed data are backpropagated to the plane that includes the finite sound source, the sampling interval of the data can be taken to be significantly larger than the half-wavelength. On the other hand, by taking the pixel interval of the reconstructed source image as less than or equal to the half-wavelength, the acoustic field at an arbitrary point in the space can be evaluated by the successive forward propagation of the source image. The effect of observing an aperture on the resolving power is also discussed.

5:05

4pSP12. Acoustic-optic hybrid wave filter development. Anthony D. Matthews (Coastal Systems Station, Dahlgren Div., Naval Surface Warfare Ctr., Panama City, FL 32407-7001)

Digital filters are developed for mitigation of wave-induced image degradation in laser vibrometer data taken from the water surface. The target is ensonified by a submerged acoustic projector operated near 10 kHz. The filters are evaluated for effectiveness in restoration of the image. Unbeamformed (raw) target data are acquired from a surface with waves of variable height. Images are formed with and without the filters to determine restoration performance. [Work sponsored by Coastal Systems Station under Internal Research funding.]

Session 4pUW

Underwater Acoustics: Scattering and Reverberation

Charles W. Holland, Chair
Planning Systems, Inc., 7923 Jones Branch Drive, McLean, Virginia 22102

Chair's Introduction—1:45

Contributed Papers

1:49

4pUW1. High-frequency scattering from a target in proximity to a randomly rough interface. Garner C. Bishop and Judy Smith (Naval Undersea Warfare Ctr. Div. Newport, Newport, RI 02840)

A T-matrix formalism for free-field target scattering and a Helmholtz–Kirchhoff integral equation in the Kirchhoff approximation for rough interface scattering are used to calculate high-frequency plane-wave scattering from a stationary target in proximity to an interface with random surface roughness. Scattering from targets located above and below the interface are considered: When the target is located above the interface, it is assumed that the scattered pressure field is simply the superposition of the target and rough interface scattered fields. When the target is located below the interface, it is assumed that target-interface multiple scattering can be ignored and that the scattered pressure field above the interface is given by the superposition of the field scattered from the interface and the field scattered from the target and transmitted through the interface when the field incident on the target is the transmitted plane-wave field. Numerical calculations indicate some of the effects of surface roughness and target physical and geometric parameters on the ability to "see" the target.

2:01

4pUW2. A composite model for rough bottom scattering. Yevgeniy Y. Dorfman and Ira Dyer (Dept. of Ocean Eng., MIT, Cambridge, MA 02139)

To be valid and accurate, any model for mid-frequency bistatic scattering must account for the different scales of roughness found on the bottom. A composite scattering model was first introduced by Kuryanov. An extension for this model, similar to the one by Kuryanov, was applied to data acquired during the 1993 ARSRP cruise. In this model, the scattering surface is represented by a random surface composed of facets. Each facet is parametrized by a mean slope to account for large scale roughness and a perturbation to account for small scale roughness. First, partial coherence of the field over the facet surface is used to determine the contribution of each individual facet. Then, the scattered field is calculated via incoherent addition of the contributions from all facets. The choice of a Gaussian distribution for the mean slope of the facets results in a good match between the large scale roughness spectrum and the empirical power law. A simplified model for scattering from the individual facets assumes that all scatter is directed into the specular direction. This allows a completely analytical solution and is in reasonable agreement with experimental data. [Work supported by ONR.]

2:13

4pUW3. Beam simulation error estimation using the scattering operator. John R. Dubberley (Naval Res. Lab., Code 7181, Stennis Space Center, MS 39529-5004)

Beam simulation has been used to estimate scattering from large surfaces by calculating the scattering from the individual subsurfaces and then summing over these subsurfaces. Previously physical intuition and comparison to method of moments (MoM) calculations were used to determine beam simulation's region of validity. Here the MoM operator is used for a direct comparison of beam simulation to MoM. The improved speed of comparison allows a more in-depth analysis of beam simulation at a lower computational resource cost. Scattering from one-dimensional random surfaces with Gaussian height distributions will be specifically examined to verify and clarify earlier beam simulation validations. Although one-dimensional surfaces are an oversimplification for most applications, they are well studied and are likely to give insight into higher dimensional applications.

2:25

4pUW4. Probability distribution functions for shallow water acoustic bottom scattering at high frequency. Robert W. Farwell (Neptune Sci., Inc., 150 Cleveland Ave., Slidell, LA 70458) and Roger W. Meredith (Naval Res. Lab., Stennis Space Center, MS 39529)

When designing signal processors to discriminate against underwater sound background noise and scattering, it is important to know the statistics of the unwanted signal disturbances. A study was undertaken for the Naval Research Laboratory, Stennis Space Center, Mississippi to review the literature reporting on a number of bottom scattering experiments that were conducted during the past 15–20 years. The scope of the investigation is limited to high frequencies (10 to 200 kHz) and shallow water of depths less than 100 m. Many of these tests were performed by the Naval Research Laboratory, Stennis Space Center; the Applied Research Laboratories, University of Texas at Austin; and the Applied Physics Laboratory, University of Washington. Other experiments reviewed were performed by Thomson-Sintra ASM, France and by the University of Wisconsin. Each experiment will be briefly described, and the statistical results discussed and summarized for various parameters, such as frequency, beamwidth, pulse length, grazing angle, range, and environmental factors. The statistic that was primarily examined was the probability of false alarm. The PFAs were found to follow a number of probability distribution functions including Gaussian, Rayleigh, log-normal, etc. The PFAs were quantified to systematically determine any trends with the parameters.

2:37

4pUW5. Measurements and modeling of low-frequency bottom-scattering strengths from the Scotian Continental Rise. Roger C. Gauss (Naval Res. Lab., Washington, DC 20375-5350), Charles Holland, Joseph M. Fialkowski, and Peter Neumann (Planning Systems, Inc., McLean, VA 22102)

A series of low-frequency (190–310 Hz), bottom-backscattering strength measurements were made on the Scotian Continental Rise in August 1993 using combinations of short-duration cw and HFM signals. Grazing angles realized were between 5 and 30 deg. A characteristic of many sites were strong (on the order of 10 dB) oscillations with a grazing angle superimposed on generally Mackenzian backscatter. The oscillatory behavior is due to caustics within the sediment, i.e., a propagation effect.

4p THU. PM

Previous modeling of this general area [P. D. Mourad and D. R. Jackson, J. Acoust. Soc. Am. **94**, 344–358 (1993)] predicted such an oscillatory behavior but only for unrealistic sediment sound–speed profiles. An alternative bottom-scattering model is proposed with the goal of predicting the observed scatter using a more realistic geoacoustic model. Measurement/model comparisons will be presented for several representative sites. [Work supported by SPAWAR, NRL, and ONR.]

2:49

4pUW6. Scattering from rough pressure-release surfaces using the finite-difference time-domain method. Frank D. Hastings, John B. Schneider, and Shira L. Broschat (School of Elec. Eng. and Comput. Sci., Washington State Univ., Pullman, WA 99164-2752)

The finite-difference time-domain (FDTD) method provides a robust and accurate means of studying a wide range of propagation and scattering problems. However, the FDTD method, though simple to implement and readily parallelizable, is computationally expensive. In this paper, a benchmark sea surface scattering problem [Thorsos, Reverberation and Scattering Workshop, Gulfport, MS (1994)] is used to show how the computational requirements of an FDTD solution can be mitigated. A contour-path approach is used so that the grid conforms to the continuously varying surface. This permits accurate modeling of the rough surface without requiring an excessive number of points per wavelength. Errors due to numerical dispersion are reduced by adjustment of the phase speed. Although the test-case problem has both transmitting and receiving arrays in the near field of the scattering surface, simple near-field to near-field transformations are used so that the transmitting and receiving arrays need not be included in the computational grid. These transformations significantly reduce the size of the grid and, correspondingly, the computational resources needed to obtain a solution. The problem considered here has been studied previously using time-harmonic techniques; however, an FDTD solution allows results to be obtained over a broad spectrum. The modifications needed to obtain broadband results are discussed. [Work supported by ONR.]

3:01

4pUW7. A Monte Carlo finite-difference time-domain technique for scattering from rough elastic bottoms. Frank D. Hastings, John B. Schneider, and Shira L. Broschat (School of Elec. Eng. and Comp. Sci., Washington State Univ., Pullman, WA 99164-2752)

Scattering from elastic ocean bottoms has been modeled using approximate analytical methods such as perturbation theory and the Kirchhoff and small slope approximations. An alternative approach is to use a numerical technique. This paper describes one such technique, the finite-difference time-domain (FDTD) method, which has been used successively to model scattering from pressure release surfaces [Hastings *et al.*, IEEE Trans. Antennas Propagat. **43**, 1183–1191 (1995)]. Monte Carlo scattering strengths are obtained for a tapered incident beam insonifying rough granite and basalt surfaces with either a Gaussian spectrum or a modified power law spectrum. These results are compared with those of formally averaged scattering strengths obtained for several different approximate models. One benefit of using the FDTD approach is that it allows the addition of heterogeneities at little extra computational cost. However, the initial computational cost is great, and the presence of surface and subsurface waves may result in strong scattering at grazing angles despite the use of the tapered incident field. Additionally, grazing incident angles may require surface lengths that are too computationally expensive. These issues and how they affect the applicability of the FDTD method to elastic problems are discussed. [Work supported by ONR.]

3:13

4pUW8. Bottom scatter measurements and modeling from the California Continental Borderlands. Charles W. Holland, Peter Neumann, and Andy Rogers (Planning Systems, Inc., McLean, VA 22102)

Bottom scatter measurements were conducted in several littoral and deep-water regions using explosive charges (processes from 100 to 2000 Hz) and low-frequency (processed from 175 to 325 Hz) coherent sources. The measurements provide an opportunity to compare bottom scatter data using coherent and incoherent sources. The bottom scatter collected by coherent and incoherent sources appears to be identical within measurement uncertainties. This important result implies that survey measurement techniques (which use incoherent sources) may be employed for developing databases that support coherent systems. A variety of seafloor mechanisms controlled the scattering depending upon frequency, grazing angle, azimuth, and environment. However, modeling results indicate that the dominant scattering mechanisms arise from sub-bottom features and not the water-sediment interface. One important scattering mechanism appears to be a sub-bottom horizon. This kind of mechanism is generally not treated in current scattering models. A modeling approach is presented that captures the angular and frequency dependence of the sub-bottom horizon scattering. [Work supported by SPAWAR and ONR.]

3:25

4pUW9. A wide frequency range model for seabed scattering. Anatoliy N. Ivakin (Andreev Acoust. Inst., Shvernika 4, Moscow 117036, Russia)

A unified model of seabed scattering applicable for a wide low- to high-frequency range is presented. The first-order perturbation solution has been obtained for the scattering amplitude of a randomly inhomogeneous seabed consisting of an arbitrary number of fluid sediment layers having slightly rough boundaries and covering an elastic half-space. The model for the second statistical moments of a scattered field involves both auto- and crosscorrelations between roughnesses of different interfaces as well as between fluctuations of different bulk parameters. Frequency-angular dependencies of the scattering strength for both monostatic and bistatic cases are calculated and analyzed for various seabed types. These dependencies are shown to be largely governed by transitions from one effective seabed type to another. For example, at sufficiently high frequencies, strong absorption causes the influence of stratification to be negligible, while, at moderate frequencies, interference effects due to sediment stratification become important. At lower frequencies, sound penetration of the sediment increases, and the effects of elastic scattering and reflection from the basement supporting both compressional and shear waves and corresponding interference become important. All these effects are considered in the frame of a unified model and their sensitivity to different medium and signal parameters is analyzed. Possible applications are discussed. [Work supported by ONR.]

3:37

4pUW10. Acoustic radiation or scattering: A multifrequency analysis by boundary element method. Christian Vanhille and Antoine Lavie (Inst. d'Électronique et de Microélectronique du Nord, U.M.R. C.N.R.S. 9929, Dépt. I.S.E.N., Lab. d'Acoustique, 41 Boulevard Vauban, 59046 Lille Cédex, France)

In the field of underwater acoustics, the boundary element method is usually used to model acoustic radiation or scattering from an immersed structure impinged by an acoustic wave. The EQI boundary element code uses Helmholtz integral equation. The Sommerfeld radiation condition is implicitly included in the formulation, and the pressure field in the surrounding fluid is evaluated from the mesh restricted to the surface of the body. At some frequencies called irregular frequencies, this integral formulation presents an infinite number of solutions. This problem is solved with the help of the Jones method: The integral equations system is overdetermined by null-field equations. For a multifrequency analysis, integral equations matrix assembling is very computation time consuming. A new boundary element algorithm based upon a frequency interpolation is proposed. It is adapted to quadratic isoparametric discretization for any geometry. This technique is original for two reasons: the automation of its utilization between 0 Hz and the top frequency of the spectrum; its adaptation to axisymmetrical problems by decomposition of circular integration surfaces. This algorithm allows an important saving of computation time which increases with the size of the problem without loss of precision.

3:49

4pUW11. Scattering from an axisymmetric structure insonified at any incidence using a coupled finite-element/boundary element method. Antoine Lavie and Bertrand Dubus (Institut d'Électronique et de Microélectronique du Nord, U.M.R. C.N.R.S. 9929, Département I.S.E.N., Lab. d'Acoustique, 41 Boulevard Vauban, 59046 Lille Cédex, France)

In the field of the scattering of acoustic waves from structures immersed in an infinite fluid medium, a coupling between the EQI boundary element code and the ATILA finite-element code is available. The boundary element method used is the Helmholtz integral equation. At some frequencies, called irregular frequencies, this integral formulation presents an infinite number of solutions. This problem is solved with the help of the Jones method: The integral equations system is overdetermined by null-field equations. Many structures are axisymmetric. So, it is attractive to develop a modeling that is able to keep the advantages of the axis of symmetry and to take into account an incident wave of any direction. This numerical tool is performed using a harmonic cylindrical decomposition. In this case, the mesh is restricted to the line of the structure included in the meridian plane. Compared with three-dimensional modeling, this new possibility provides the following advantages: either a reduction of the computation time, the memory size, and the disk storage space and an improvement of the numerical behavior; or an increase of the maximum frequency. Comparison between numerical results and experimental data is presented.

4:01–4:15 **Break**

4:15

4pUW12. Volume reverberation in littoral waters. Richard H. Love, Charles H. Thompson, and Redwood W. Nero (Naval Res. Lab., Stennis Space Center, MS 39529)

Until recently, virtually no low- and mid-frequency volume reverberation data were available from littoral waters. However, in the past few years, the Naval Research Laboratory has conducted volume reverberation measurements in deep, slope, and shelf waters in several oceanic regions. Deep ocean volume reverberation is relatively uniform over broad ocean areas at both low- and mid-frequencies. As water depths become shallower, many fishes responsible for deep ocean volume reverberation eventually disappear and the character of the reverberation changes. Measurements in littoral waters demonstrate that low- and mid-frequency volume reverberation levels can be very high and can vary greatly over short distances and from day to night. Results also show that changes in volume reverberation from deep to shallow water can be significantly different in different regions. Variability is the principal feature of volume reverberation in littoral waters.

4:27

4pUW13. A numerical study of time-domain backscattering from one-dimensional, rigid, random surfaces. Vincent Lupien (Dept. of Ocean Eng., MIT, Cambridge, MA 02139)

In certain high-resolution target detection systems, impulsive scatter from ocean boundaries gives rise to sonar clutter. In this work, the main objective was to identify the stochastic characteristics of rigid, one-dimensional rough surfaces which create backscatter containing spikes in the time domain. The interarrival time and amplitude statistics of impulsive events were determined numerically through Monte Carlo realizations of different surface types. For each surface the far-field impulse response in backscatter was determined using an exact integral equation method adapted from E. I. Thorsos [J. Acoust. Soc. Am. **83**, 78 (1988)]. The impulse responses were convolved with the source spectrum to generate exact backscattered waveforms. Three sets of surfaces were investigated. In the first set, surfaces with a Gaussian correlation function were considered, which possess a single horizontal and vertical scale. In the second set, surfaces with a multiscale, power-law spectral density were investigated. The third set consisted of surfaces made up of adjacent straight line segments, or facets. The co-ordinates of the vertices of each facet were randomized through a discrete stochastic process. The results have provided insight into the physical mechanisms which generate spikes in the time domain and helped distinguish the contribution that localized scatterers make versus those that result from random interference of nonadjacent points on the surface. [Work sponsored by the Office of Naval Research.]

4:39

4pUW14. The effect of environmental parameters on surface scattering strength. Michael Nicholas (Naval Res. Lab., Washington, DC 20375)

Throughout the Critical Sea Test (CST) experiments, a large body of quality surface scattering data was accumulated which was collected over a broad range of sea states using both explosive (SUS) charges (impulsive sources) and transmitting arrays (waveform sources). Simultaneously, an extensive suite of environmental measurements was also made, providing a unique data set with which to examine how the surface scattering strength changes with various environmental parameters. Historically wind speed has been the environmental descriptor of choice, but while this is the dominating factor, wind speed alone was found to be inadequate to fully describe all the observations. Other important environmental descriptors will be discussed, and some of their effects on surface scattering at both very high and very low sea states will be demonstrated. The latest empirical surface scattering strength algorithm based on all the CST SUS measurements will also be described, and the effect of current findings on future modifications to this algorithm will be discussed.

4:51

4pUW15. Modeling low-frequency surface scatter intensity statistics. Eric I. Thorsos, Donald B. Percival, and Kate M. Bader (Appl. Phys. Lab., College of Ocean and Fishery Sci., Univ. of Washington, Seattle, WA 98105)

Low-frequency surface backscattering experiments (CST-7, ASREX) show that the surface scattered field statistics are non-Gaussian, which means the intensity statistics will be nonexponential. These data show the intensity distribution to be closer to log normal. In such cases the statistical distribution depends on the scattering area on the surface or, equivalently, on the number of scatterers. An analytical model has been developed for the intensity distribution that results when the scattering area contains multiple, spatially independent, identically distributed scatterers. The individual scatterers, e.g., bubble clouds, are assumed to yield a log-normal intensity distribution. The assumption that the scatterers are spatially independent leads to a Poisson distribution for the number of scatterers in a given surface area. The parameters to be specified are the standard deviation of the log-normal distribution and the average number of scatterers per unit area. Monte Carlo simulations yield estimated intensity distributions in very good agreement with the analytical model. Preliminary comparisons will be made with ASREX data. [Work supported by ONR.]

5:03

4pUW16. Rayleigh scattering from a plume of suspended particulates located near an ocean boundary. David R. Palmer (NOAA/AOML, 4301 Rickenbacker Cswy., Miami, FL 33149)

A number of experiments have been conducted to acoustically image plumes of particulates suspended in the ocean. In many cases the scattering is in the long-wavelength or Rayleigh region. A comprehensive framework was recently developed for calculating the intensity received by a monostatic sonar system due to backscattering in the Rayleigh region from a plume of suspended, nonspherical particulates [D. R. Palmer, J. Acoust.

Soc. Am. **99**, 1901–1912 (1996)]. This framework is extended to include the possibility that the plume is close to an ocean boundary. The distances of the particulates from the boundary are assumed to be small compared to the wavelength and the boundary surface is assumed to be planar. The scattering amplitude in the presence of a rigid or a pressure-release boundary surface is related to the amplitude in the absence of the surface. As a result the intensity can be averaged over the random orientations of the individual particulates and this averaged intensity can be bounded in terms of the intensity that would be measured if the particulates scattered the acoustic wave as if they were spherical.

5:15

4pUW17. Sound scattering by the complexly shaped objects within the sea waveguides. E. I. Oboznenko and I. L. Oboznenko (Acoust. and Acoust. Electron. Chair, Kiev Polytechnic Inst., Kiev, Ukraine)

An algorithm for the sea waveguide located by arbitrary shaped objects' echosignals computation is obtained. The problem is solved with the help of boundary integral equations for the potentials of a single (or double) layer on the surface of the object, that is located within the waveguide that has already been defined by a Green function for its object-free state. The Pekeris waveguide is taken as an example. The object's wave dimensions were equal to 0.5–15 (low-frequency range of echolocation). Up to 12 modes could be present in the waveguide. Scattering patterns' calculations have been made for the object's azimuthal plane at the different distances between the source, the object, and the receiver and their mutual location's different depths. Object scattering patterns (both in static and in dynamic object state) have been investigated, hydrodynamical effects have been ignored. The echosignal for the four-component receiver (pressure and particle velocity vector), as well as for the scattered wave intensity vector, has been determined. It is shown that velocity and intensity vector method usage when receiving the scattered wave in the waveguide increases substantially the echosignal informative content when compared to the free sound field measurements, and this is due to the modes interaction both in scalar and in vector waves. The scattering pattern for the azimuthal plane contains the information about the scattering within the two mutually orthogonal planes. The results of the scattered fields' hydrophysical modeling (scaling factor 1:20–400 and 1:1) are presented. The theoretical and model investigation results are in good agreement.

THURSDAY AFTERNOON, 5 DECEMBER 1996 AKAKA FALLS ROOM, 1:30 P.M.

Meeting of Accredited Standards Committee S1 on Acoustics

to be held jointly with the

U. S. Technical Advisory Group for ISO/TC 43 Acoustics and IEC/TC 29 Electroacoustics

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, National Research Council, Ottawa, Ontario K1A 0R6, Canada

P. D. Schomer, Chair, U. S. Technical Advisory Group (TAG) for ISO/TC 43, Acoustics
U. S. CERL, P.O. Box 4005, Champaign, Illinois 61820

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, 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, electro-acoustics, 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. The international activities in ISO/TC 43 Acoustics, and IEC/TC 29 Electro-acoustics, will also be discussed. The chairs of the respective U. S. Technical Advisory Groups for ISO/TC 43 (P. D. Schomer), and IEC/TC 29 (V. Nedzelnitsky), will report on current activities of these international technical committees.

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