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PROGRAM OF

Joint Meeting

The 137th regular Meeting of the Acoustical Society of America

2nd Convention of the European Acoustics Association: Forum Acusticum

integrating the 25th German Acoustics DAGA meeting

Berlin, Germany • 15–19 March 1999

NOTE: All Journal articles and Letters to the Editor are peer reviewed before publication. Program abstracts, however, are not reviewed before publication, since we are prohibited by time and schedule.

MONDAY AFTERNOON, 15 MARCH 1999   POSTER GALLERY, 12:00 TO 6:20 P.M.

Posters from various technical sessions remain on display in the Poster Gallery

Also, the following poster sessions are scheduled:
Poster Session 1pPAe
Poster Session 1pPPb
Poster Session 1pSCb

MONDAY AFTERNOON, 15 MARCH 1999   ROOM H3010, 1:55 TO 6:20 P.M.

Session 1pAA

Architectural Acoustics: Concert and Opera Halls: Case Studies of New Halls; Opera Houses

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

Jürgen Meyer, Cochair
Physikalisch-Technische Bundesanstalt, Bundesalle 100, 38116 Braunschweig, Germany

Chair's Introduction—1:55

Invited Papers

2:00

1pAA1. Acoustic design of the Tsuyama Music Cultural Hall based on the preference theory. Y. Ando, Y. Suzumura (Grad. School of Sci. and Technol., Kobe Univ., Rokkodai, Nada, Kobe, 657-8501 Japan, andoy@kobe-u.ac.jp), and I. Yamamoto (Arch. & Env. Res. Ltd., Kobe, Japan)

The city of Tsuyama is located about 100 kilometers west of Kobe. The fundamental shape of the plan and cross sections of the hall were designed by applying the theory of subjective preference [Ando, *Architectural Acoustics* (AIP Press/Springer-Verlag, New York, 1998)]. Special attempts made in the acoustic design of this hall are: (1) a number of columns distributed in front of the walls at the audience level and on the stage: This may keep a small value of the IACC and a certain initial time delay gap between the direct sound and the first reflection, making an increase of the subjective preference for both listeners and musicians; (2) a stage enclosure with the canopy of several triangular reflectors at adjustable height for musicians, to control the preferred delay time of the reflections in performing a certain type of music; and (3) the shape of the rear wall of the stage, to control the IACC for listeners.

2:20

1pAA2. The new Konzertsaal of the KKL Center, Lucerne, Switzerland. I. Acoustics design. Russell Johnson and Eckhard Kahle (Artec Consultants, Inc., 114 W. 26th St., 9th Fl., New York, NY 10001-6812, ek@artec-usa.com)

The 1840 seat concert hall is the first of three halls that will eventually make up the Culture and Convention Center in Lucerne. It was inaugurated on 19 August 1998 to widespread critical acclaim. The hall is the home of the Lucerne International Music Festival, one of the most prestigious international festivals dedicated to symphonic music. The room will first be presented and illustrated, and an outline will be given of the basic acoustics design features such as room shape, seat count, and balcony design. Then, focus will shift to more specific acoustic design features, such as the reverberance chamber surrounding the main audience chamber, the two-part moving canopy with both horizontal and vertical reflecting surfaces, and the acoustically diffusing surface finish on most of the side wall area of the hall.

2:40

1pAA3. The new Konzertsaal of the KKL Center, Lucerne, Switzerland. II. Preliminary acoustical measurements. Eckhard Kahle, Russell Johnson, and Brian Katz (Artec Consultants, Inc., 114 W. 26th St., 9th Fl., New York, NY 10001-6812, ek@artec-usa.com)

The newly opened Concert Hall in Lucerne is the home of the International Music Festival with its extremely varied programs. Furthermore, the remainder of the year requires this hall to provide an appropriate acoustic environment for an even wider range of performances, ranging from conferences and other speech events to symphony orchestra, amplified popular music, ethnic music, and organ recitals. In a town with a population of 60 000, it will also be necessary to accommodate a varying number of audience members. For all these reasons, several elements of acoustic variability have been included in the design of the hall. During the first months of the inaugural season, the settings of the variable acoustics features of the hall were documented for each individual performance in the form of a logbook. This paper will discuss the use of the acoustical elements for selected performances taken from the ongoing log. The results of preliminary acoustical measurements and subjective listening impressions for the selected performances will be given. Furthermore, objective measurements were recorded in the hall at various stages of construction.

3:00

1pAA4. The acoustics evolution of Chicago's Orchestra Hall: A case study. R. Lawrence Kirkegaard (Kirkegaard and Assoc., Inc., 4910 Main St., Downers Grove, IL 60515)

Originally opened in 1904, Chicago's Orchestra Hall was immediately recognized as acoustically flawed due to low reverberation times, poor bass response, a crowded stage, and poor on-stage hearing conditions. In 1991, the Orchestral Association of Chicago decided to renovate and expand Orchestra Hall into a comprehensive Symphony Center for music in Chicago. The core of the project was a commitment to improve the acoustics to the extent feasible within the constraints of the site and the historic beauty of the hall. The Association contracted directly with Kirkegaard and Associates as acoustical consultants to work in parallel with the architects, Skidmore Owings and Merrill. The renovated hall reopened in the fall of 1997, with completion of acoustics work delayed until summer of 1998. The renovated hall is beautiful architecturally, and its acoustics has been gratifyingly improved. This paper describes the history and results of the remodeling. Test data as well as subjective assessments will be discussed and important acoustics lessons shared.

3:20

1pAA5. Yokohama Minato Mirai Hall. Hideki Tachibana (Inst. of Industrial Sci., Tokyo Univ., Roppongi 7-22-1, Minato-ku, Tokyo, 106-8558 Japan, tachibana@iis.u-tokyo.ac.jp) and Hiroyuki Kimura (Nikken Sekkei Ltd., Kouraku 2-1-2, Bunkyou-ku, Tokyo, 112-0004 Japan)

The Yokohama Minato Mirai Hall was planned as a complex cultural facility on the waterfront of Yokohama City and was opened in June 1998. In this building, a large concert hall with 2020 seats (mainly for symphony concerts) and a small concert hall with 400 seats mainly for chamber music are included. For the design and construction of this building, an acoustic design and research group consisting of architects, acousticians, and building engineers was set up and such various acoustic problems as room acoustics, sound insulation between rooms and facades, prevention of structure-borne sound from subway by floating floor construction, and noise reduction of the air-conditioning system were investigated by cooperative work in the group. For the acoustic design of the two halls, computer simulation based on geometrical acoustics and a 1/10 scale model experiment were performed for checking room shape and auralization studies. During and after the construction, acoustic measurements were carried out under various conditions. Before the opening, test concerts were performed with an audience, and acoustic data under occupied conditions were obtained. In this paper, the outline of the design process and the acoustic characteristics of the two halls are introduced.

3:40–4:00 Break

4:00

1pAA6. Acoustical design of the Marienkirche concert hall, Neubrandenburg. Henrik E. Möller and Tapio Lahti (Akukon Oy Consulting Engineers, Kornetintie 4 A, FIN-00380 Helsinki, Finland, henrik.moller@akukon.fi)

St. Mary's medieval gothic cathedral in Neubrandenburg, Germany, was ruined at the end of World War II. Two decades ago a decision was made to turn it into a concert hall. The design started in 1996 with the architectural competition, won by Architects Pekka Salminen. The hall will be a combination of old brick walls and mullioned windows, joined with modern glass and concrete. The back wall and the ceiling are made of glass. An audience of 1200 maximum is divided into three blocks: stalls, a balcony, and a tier behind the stage. The initial properties were promising: rectangular, 21 m wide, 18 m high, and with inherent diffusion. The

acoustical design was challenged by the restriction of not to touch the old restored side walls. A full-audience reverberation time of 2.0 s was set as a goal. Other objectives were moderate reverberation rise in bass, strong spaciousness and lateral energy. The acoustics were designed with a computer using the Odeon hall design program. Main new elements of acoustical control are a cloud array above the stage, triangular sails near the side wall cornices, and glass reflectors beside the stage. The hall is expected to be inaugurated in autumn 1999.

<div style="text-align:center">4:20</div>

1pAA7. Festspielhaus Bayreuth—The unique acoustic situation. Karlheinz Müller (Müller-BBM, Robert-Koch-Str. 11, 82152 Planegg b. Munich, Germany)

The Festspielhaus in Bayreuth is the only opera house of the world which was founded and mostly designed by a composer. Richard Wagner himself founded the festival society and was one of the main designers of the opera house, the stage situation, and the pit. It is a very modern opera house, far away from the typical aristocratic or common theater of the times of 1876. The open house has a special acoustic situation. What are the reasons for this acoustic situation? Especially the sunken-pit design is subject to endless controversial discussions. All the maintenance work of the last years had the priority not to destroy the typical mysterious sound of the original Festspielhaus. Although all part of the whole house are now redecorated, the typical Wagnerian sound has been perfectly conserved.

<div style="text-align:center">4:40</div>

1pAA8. Acoustics of opera houses: A cultural heritage. Patrizio Fausti, Roberto Pompoli, and Nicola Prodi (Dip. di Ingegneria, Univ. di Ferrara, via Saragat 1, 44100 Ferrara, Italy, pompoli@ing.unife.it)

An important part of contemporary Italian musical culture has developed inside historical opera houses. An outstanding opera repertory was especially conceived to match these places, both from the architectural and the acoustical points of view. But, while extensive attention is devoted to architectural and historical aspects, the acoustical characteristics of an opera house, that is, the set of listening attributes which make such a place unique among all spaces for music, are still nowadays hardly recognized as being paramount. In fact they can be rightly considered a relevant cultural heritage in Italy and can thus be called acoustical heritage. This new kind of heritage calls for accurate study and attention, both from the scientific and social points of view. To preserve the heritage for future generations, it is necessary to know it precisely and to elaborate means of accurate description. A project with the participation of international specialists has been established to agree on a standardized procedure for acoustic measurements specific to Italian historical opera houses in order to investigate the pecularity of these places. The aim of the project is to give a state-of-the-art acoustic documentation fo some theaters with the possibility of later applying up-to-date acoustical analysis.

<div style="text-align:center">5:00</div>

1pAA9. New subjective and objective data on 20 opera houses of the world. Leo L. Beranek (975 Memorial Dr., Ste. 804, Cambridge, MA 02138-5755) and Takayuki Hidaka (Takenaka Res. and Development Inst., Chiba, 270-1395, Japan)

Acoustical measurements have been made in opera houses of Europe, Japan, the United States, and Argentina using the same equipment: dodecahedron loudspeaker, stretched impulses, and monaural and binaural sensing devices. The measuring positions were 3 on-stage and 2 in-pit for sources, and 8 to 17 in audience areas for sensors. Quantities measured as a function of frequency were reverberation time RT, early decay time EDT, clarity C80, deutlichkeit D, time gravity Tg, early to reverberant energy ratio as a function of cutoff time $C(t)$, strength G, interaural cross correlation IACC, IACCE, and IACCL, stage support ST1 and ST2, initial-time-delay gap ITDG, and reflectograms. International opera conductors were contacted by mail to rate 20 opera houses, with satisfactory responses. An attempt is made to correlate the objective data with the subjective ratings. Early results indicate that IACCE3 and ITDG are definite indicators. Others seem important in a certain type of opera house.

<div style="text-align:center">5:20</div>

1pAA10. Acoustic considerations in the redevelopment of the Royal Opera House, London. Rob Harris (Arup Acoustics, Parkin House, 8 St. Thomas St., Winchester, Hampshire SO23 9HE, UK)

The Royal Opera House, Covent Garden is currently closed for a major redevelopment. It is scheduled to re-open in December 1999 with a refurbished and extended main auditorium, a studio theater seating 400, major new ballet studios and facilities for the Royal Ballet, new and refurbished opera and chorus rehearsal rooms, and a completely new backstage, with full wagon facilities and a double height flytower. This paper discusses the acoustic philosophy behind the design and reports the design solutions and the progress on site to date. Matters of particular interest include the approach to the preservation/enhancement of the acoustic of the main house, the dynamics of the ballet studios, the exceptionally large and complex acoustic scenery doors and the acoustic variability incorporated into the studio theater, which is a box-in-box construction.

<div style="text-align:center">5:40</div>

1pAA11. Acoustic quality in the Theatre ''Palafenice,'' Venice. Lamberto Tronchin (DIENCA-CIARM, Viale Risorgimento 2, 40136 Bologna, Italy)

The burning of the Teatro La Fenice, in Venice, has also been a tragedy for the acoustical community. The Municipality of Venice decided to build a tense-structure (called PalaFenice) on the isle of Tronchetto, as a temporary auditorium for operas and concerts. The use of such a structure provokes many different acoustic problems, quite different from those already known to musicians at La Fenice. In this paper the acoustic quality of the theatre has been analyzed, and compared to the acoustic measurement already

performed in the former La Fenice theatre. Binaural measurements have been performed in the hall using the impulse response technique (including absorption of the ceiling), and a dummy head located at different listening positions. According to ISO 3382, many acoustic parameters have been evaluated, like listening level, ITDG, reverberation time, IACC, and others, and their values have been mapped. Also, Ando's quality maps of preferences, with reference to two different kinds of musical signals, was accomplished from experimental measurements. The measurements pointed out an unsuitable behavior of the structure at low frequency and low reverberation time and intelligibility in the hall. Finally, an hypothesis of the acoustic chamber has been formulated.

6:00

1pAA12. Orchestra shells: Acoustical and practical design considerations. Daniel E. Commins (commins acoustics workshop, 15, rue Laurence Savart, F-75020 Paris, France and 350 Fifth Ave., New York, NY 10118, comminsacoustics@compuserve.com)

A hall or an outdoor facility is rarely used for a single type of artistic program. Often, it is necessary to design an orchestra shell that transforms theoretically a theatre or an opera house into a concert hall. The acoustical criteria for opera or concerts are quite different; the design of an orchestra shell that would create proper acoustical conditions on stage for the musicians and in the hall for the audience can be quite complex. The larger the hall and the proscenium opening, the harder it is to come up with an adequate solution. The major problems are the following: architecture and aesthetics, acoustical efficiency, ensemble conditions, acoustic response, diffusion, size, weight, transformation time, and storage. Some progress has been made over the last few years, but the problem remains complex. This paper will attempt to explain the design and construction process and will provide recent examples of small and large shells: Ravinia, Lille concert hall, Gradignan, Porto Teatro S. João, Opera Garnier in Paris, Staatsoper in Munich, and Serralves auditorium in Porto.

Contributed Posters

The following posters will be on display in the Poster Gallery from Monday to Wednesday, 15–17 March. Contributors will be at their posters from 10:00 a.m. to 12:00 noon on Wednesday, 17 March.

1pAA13. Acoustical modifications of the interior design of a 50-ft chapel. Catherine Sémidor and Emmanuel Merida (ERIAC Ecole d'architecture et de paysage de Bordeaux Domaine de Raba, 33405 Talence Cedex, France, catherine.semidor@bordeaux.archi.fr)

In order to organize congresses, the University of Sciences in Bordeaux has bought a building with a chapel which will be modified to become both a lecture room and a place for music entertainment with a small orchestra. This paper deals with an interdisciplinary study from the acoustical inventory to the proposition of the new interior design, realized simultaneously by the acoustician team, the architect, and the university. The place, a church, is notable for its high reverberation time. Given this characteristic, a modified MLS software permitting measurements in such halls was tested. The numerical simulation was ajusted from the room-acoustics measured criteria. This reliable model became the base of the study to validate the architectural choices leading to the ''right acoustics'' for the ''right place.'' Among several acoustical solutions, the one presenting the more practical application was chosen. The design concept developed for this hall actually permits a variable acoustics to achieve the right level of reverberance for music and good speech clarity for lectures. The results of measurements carried out in the newly opened chapel will be compared with the computed criteria of the last model.

1pAA14. The acoustical properties of large studios. Bojan Ivancevic, Hrvoje Domitrovic, and Sinisa Fajt (Faculty of E. E. and Computing, Dept. of Electroacoustics, Unska 3, HR-10000 Zagreb, Croatia, bojan.ivancevic@fer.hr)

New studios for Croatian Radio-Television were planned with the starting user conditions. The background noise level must fulfill $N15$ or $N20$ criteria. Demanded reverberation time for a 400 m^2 radio studio is $Tr=0.6$ s, for TV studios of the same area, $Tr=0.7$ s, and for the 1000 m^2 TV studio, reverberation time is 1 s. Since all three studios with built-in control rooms were built in an urban area with excessive noise, it was necessary to obtain very good protection against noise, and at the same time, using the same elements, achieve the desired reverberation

time. Special attention was given to flutter echoes and achievement of diffuse sound fields. The inner shell construction was built up using special acoustical building blocks. Besides using double walls, the broad analysis was done in order to avoid any possible problem in sound insulation. The achieved coincident frequency for walls and floors is 7.6 Hz, and this is quite satisfactory. The computer simulation was used during the design of internal sound-absorbing and reflecting surfaces. Described solutions take care of all user demands, as well as the usage particularities of such spaces. The measurements of relevant acoustical parameters were done after the building was finished.

1pAA15. The sound quality of the Teatro Comunale in Treviso, Italy. Lamberto Tronchin (DIENCA-CIARM, Viale Risorgimento, 2 40136 Bologna, Italy)

The Teatro Comunale of Treviso was designed at the same time as the Teatro La Fenice of Venice, in the middle of the 19th century. Treviso being so close to Venice, the reputation of the Venetian Theatre, just rebuilt after the burning of 1836, influenced the municipality of this little town, which decided to design the Theatre with the same shape as la Fenice. After the burning of 1996, and following the disappointment of losing such unique architecture and sound quality, acoustical measurements were performed in many Italian theatres, especially in historic theatres where acoustical measurements were never before undertaken. In this paper, the results of the acoustic measurements are presented. According to ISO 3382, binaural measurements, with a dummy head, have been performed in 25 positions in the hall, with different positions of the sound source in the proscenium. Acoustical parameters, such as EDT, G, LE were calculated from the impulse responses and mapped. Also, Ando's quality maps of preferences, with reference to two different kinds of musical signals, were accomplished from experimental measurements. From the measurements, a good behavior of the theatre has been pointed out, even if some limitations in some positions have been discovered.

Session 1pAB

Animal Bioacoustics: Animal Electrophysiological Acoustics

Philip H.-S. Jen, Cochair

Division of Biological Sciences, University of Missouri, Columbia, Missouri 65211, USA

Alexander Ya. Supin, Cochair

Institute of Ecology and Evolution, Russian Academy of Science, 33 Leninsky Prospect, 117071, Moscow, Russia

Invited Papers

2:00

1pAB1. ''Internal spectra'' in the dolphin's auditory system obtained by a minimal-masking technique.　Alexander Ya. Supin and Vladimir V. Popov　(Inst. of Ecology and Evolution, Russian Acad. of Sci., 33 Leninsky Prosp., 117071 Moscow, Russia)

To understand signal processing in the auditory system, it is of importance to know the ''internal spectrum'' representations of test stimuli, i.e., profiles of excitation along the frequency-representation axis. Such profiles may be obtained using a ''minimal-masking'' technique: An excitation profile is reflected by a masking curve (masker level dependence on frequency) when a masking criterion is not the threshold response to the probe stimulus but the threshold masking effect—a very small standard decrease of the probe response. The dolphin's auditory system provides good opportunities for such measurements because of the large amplitude and high consistency of the auditory brain-stem-evoked potentials (ABR). Minimal-masking curves were obtained in bottlenosed dolphins for various probe stimuli: narrow-band, rectangular, and rippled spectrum stimuli. At low stimulus levels, the profiles reproduced the stimulus spectrum well, except for a little wider bandwidth; this widening corresponded to the bandwidth of peripheral auditory filters. Stimuli with rectangular spectra evoked excitation profiles with enhanced edges, thus indicating enhancement of spectral contrast by lateral effects. Stimuli with rippled spectra evoked profiles reproducing ripples; however, edge ripples were also markedly enhanced. At high stimulus levels, all stimulus types evoked excitation profiles markedly expanded to high frequencies, far beyond the stimulus bandwidth.

2:20

1pAB2. Effect of removing physiologically relevant sound-localization parameters on tuning properties of barn owl tectal neurons.　Hermann Wagner, Iris Poganiatz　(Inst. Biologie II, RWTH Aachen, Kopernikusstrasse 16, D-52074 Aachen, Germany), Israel Nelken, and Gilad Jacobson　(Hebrew Univ., Ein Karem, Jerusalem 91120, Israel)

Barn owls use the interaural time difference (ITD) for locating sounds in azimuth and the interaural level difference (ILD) for locating sounds in elevation. Neurons in the optic tectum have spatially restricted receptive fields. Recently, Keller *et al.* [Hearing Res. **118**, 13–34 (1998)] have demonstrated that the spatial patterns of responses were almost indistinguishable in response to virtual [stimuli filtered with head-related transfer functions (HRTFs)] and to free-field stimulation in neurons of the inferior colliculus. The inferior colliculus provides input to the optic tectum. The spatial restriction of tectal neurons is mainly due to their sensitivity to ITD and ILD. This paper deals with the effect of removing ITD and/or ILD from the virtual stimuli on the tuning of tectal neurons. HRTFs of two barn owls were recorded, and noise stimuli with ITDs, ILDs, filtered with HRTFs, filtered with HRTFs but containing no ITDs, and filtered with HRTFs but without ILDs were used in the experiments. Preliminary results suggest that removal of ITDs or ILDs caused a substantial loss of spatial tuning in tectal neurons. These results support earlier findings that suggested an essential role of the tectal neurons in sound-localization behavior. [Work supported by GIF.]

2:40

1pAB3. Temperature effects on the auditory system: An amphibian case history.　Peter M. Narins　(Dept. of Physiological Sci., Univ. of California, Los Angeles, 405 Hilgard Ave., Los Angeles, CA 90095)

The inner ear of anurans is unique in that it contains three organs specialized for sound reception. The amphibian papilla (AP), the basilar papilla (BP), and the saccule (S) are anatomically distinct, spatially separate organs each with its own complement of sensory hair cells and overlying tectorial structure. The BP and portions of the AP respond exclusively to airborne sounds, whereas other portions of the AP and the S exhibit sensitivity both to airborne sounds and to substrate-borne vibrations. Measurements from the auditory periphery of *Rana pipiens pipiens* suggest that in contrast to BP fibers, AP fiber tuning is highly temperature-dependent. Moreover, hair cells from the rostral AP and S exhibit a clear temperature dependence in their intracellular current step response, unlike caudally located AP and BP cells. Finally, spontaneous otoacoustic emissions, which presumably reflect hair cell motility, are highly temperature-dependent despite the fact that the upper portion of the emission frequency range (600–1600 Hz) corresponds to the domain of the BP. A model is presented that attempts to reconcile these disparate results. [Work supported by NIDCD Grant No. DC-00222.]

1pAB4. Lateral inhibition in frequency tuning of central auditory neurons. Philip H.-S. Jen (Div. of Biological Sci., Univ. of Missouri, Columbia, MO 65211, pjen@biosci.mbp.missouri.edu)

Previous studies have shown that frequency tuning curves (FTCs) of auditory neurons in the inferior colliculus (IC) are composed of an excitatory area that is either neighbored by an inhibitory area on one flank or is sandwiched by two inhibitory areas. Neurons that are responsible for excitatory and inhibitory areas are likely neighboring neurons. Whereas this neural lateral inhibition sharpens frequency tuning and provides a means to reduce ambiguity in encoding frequency at high stimulus intensities, neurons with two-flank lateral inhibitory areas have larger Q_n values and smaller excitatory areas than neurons with one-flank inhibitory areas. The closer the inhibitory area is to the excitatory area, the sharper the frequency tuning becomes. The Q_n values of IC neurons tend to increase with the ratio of inhibitory area to excitatory area. Application of bicuculline and/or strychnine broadens FTCs of many IC neurons and completely or partially abolishes the inhibitory areas. Corticofugal pathways sharpen the FTC by narrowing the excitotory area and broadening the lateral inhibitory areas. [Work supported by NSF.]

Contributed Papers

1pAB5. Paradoxical lateral suppression in the dolphin's auditory system. Vladimir V. Popov and Alexander Ya. Supin (Inst. of Ecology and Evolution, Russian Acad. of Sci., 33 Leninsky Prosp., 117071 Moscow, Russia)

A paradoxical phenomenon was found in the auditory system of dolphins: weak sounds suppressed the brain-evoked potential responses to much stronger sounds. This occurred when the brain-evoked potentials were elicited by rhythmically amplitude-modulated sounds at modulation rates from a few hundred to more than a thousand Hz. The rhythmic-evoked response (the so-called envelope-following response) was markedly suppressed by addition of another sound of higher frequency and down to 40 dB lower intensity than the amplitude-modulated stimulus. This phenomenon was called the paradoxical lateral suppression. Only the sustained rhythmic response was a subject of this suppression, while the transient on-response to the stimulus onset was not suppressed, thus indicating that the suppression influenced the ability of evoked potentials to follow rapid amplitude modulations. At certain conditions the paradoxical lateral suppression prevents weak sounds from being masked by stronger ones. When a complex sound consists of two carriers—a higher-level, lower-frequency and a lower-level, higher-frequency, the paradoxical lateral suppression results in that evoked responses follow modulation of the weaker carrier, not the stronger one. It may help a dolphin to perceive weaker echo signals in the background of stronger emitted pulses.

1pAB6. Analysis of the role of inhibition in shaping responses to sinusoidally amplitude-modulated signals in the auditory system. George D. Pollak (Dept. of Zoology, Univ. of Texas, Austin, TX 78712)

Neurons in the inferior colliculus (IC) typically respond with phase-locked discharges to low rates of sinusoidal amplitude modulated (SAM) signals and fail to phase-lock to higher SAM rates. The hypothesis that these properties are shaped by the integration of phase-locked excitation and inhibition, as they are in lower nuclei, was tested. Responses were recorded from IC neurons evoked by SAM signals before and during the iontophoretic application of bicuculline, a competitive antagonist for $GABA_A$ receptors, strychnine, a competitive antagonist for glycine receptors, and the $GABA_B$ receptor blocker, phaclofen. The hypothesis that inhibition shapes responses to SAM signals in the IC was not confirmed. In more than 90% of the ICc neurons tested, the range of SAM rates to which they phase-locked was unchanged after blocking inhibition with bicuculline, strychnine, or phaclofen, applied either individually or in combination. These results illustrate that the same response property, phase-locking restricted to low SAM rates, is formed in more than one way in the auditory brainstem. In lower nuclei, the mechanism is coincidence of phase-locked excitation and inhibition, whereas in IC the same response feature is formed by a different but unknown mechanism. [Work supported by NICD.]

1pAB7. Measurement of impulse noise-induced temporary threshold shift in endangered and protected animals—Two case studies. Ann E. Bowles, Larry Wolski, Erik Berg, and Pamela K. Yochem (Hubbs-Sea World Res. Inst., 2595 Ingraham St., San Diego, CA 92109, annb1@san.rr.com)

Sonic booms at levels typical for supersonic airlines (<3 psf) are considered harmless to hearing based on laboratory studies. However, data on laboratory animals are difficult to generalize to other species. Wildlife present a particular problem because experimental efforts to induce permanent threshold shift (PTS) cannot be conducted for ethical or management reasons. Instead, ''significant'' temporary threshold shift (TTS) has been proposed as a conservative damage risk criterion (DRC). With this in mind, two series of experiments were conducted using simulated N waves at levels up to 6 psf, rise time down to 0.4 ms, and 100- or 300-ms duration—one on seals and sea lions and a second on the desert tortoise. Level of the least-detectable auditory brainstem response (ABR) was used to estimate best sensitivity before and after exposure. Small immediate threshold shifts (~ 5 dB) were detected in pinnipeds and larger (~ 20 dB) shifts in desert tortoises. The greater impact on desert tortoises may be explained by their greater best sensitivity at low frequencies. No PTS was detected. Thresholds for TTS were therefore detected in both species and may be useful in establishing DRCs for animals. [Work supported by NASA, Contract No. NAS1-20101 and USAF, Contract No. F33615-89-D-4003.]

Session 1pEA

Engineering Acoustics: 3D Loudspeaker Reproduction Methods

Neil A. Shaw, Cochair

Menlo Scientific Acoustics, Inc., P.O. Box 1610, Topanga, California 90290-1610, USA

Diemer de Vries, Cochair

Laboratory of Acoustic Imaging and Sound Control, Delft University of Technology, P.O. Box 5046, 2600 GA Delft, The Netherlands

Chair's Introduction—1:55

Invited Papers

2:00

1pEA1. Factors in the performance of systems for the production of virtual acoustic environments. P. A. Nelson, O. Kirkeby, Y. Kahana (Univ. of Southampton, Southampton SO17 1BJ, England), and H. Hamada (Tokyo Denki Univ., Tokyo, Japan)

This paper will describe recent progress in the development of systems designed to present, at the ears of a listener, the signals necessary to produce the illusion in the listener of the existence of a ''virtual'' source of sound. Attention will be restricted to systems which use a relatively small number of loudspeakers whose input signals are determined by optimal processing of the signal to be associated with the virtual source. It will be demonstrated that the form of the sound field produced is crucial to the success of such systems. Emphasis will be given to a system that uses two very closely spaced loudspeakers to transmit a particular form of sound field which most easily produces the iteraural time delay associated with a given virtual source position. Results will be presented of subjective experiments and of numerical simulations of the sound field. Factors determining the system performance will be discussed, particularly with regard to front–back confusions and individual differences in head related transfer functions. It will also be shown that systems using four louspeakers can significantly reduce the degree of front–back confusion even when assuming a relatively simple model of the listener's HRTF.

2:20

1pEA2. Creating virtual surround using dipole and monopole pressure fields. John W. Norris (Harman Multimedia, Harman Intl., 8500 Balboa Blvd., Northridge, CA 91329)

The objective of virtual 3-D sound technology is to convey to the listener an accurate impression of an acoustic environment. This is accomplished by conveying a realistic directional impression of sounds. Most of the successful techniques for doing this are based on the head-related transfer function (HRTF). In this paper a new method that creates the desired pressure field using dipole and monopole pressure fields will be presented. The object is to recreate locally the pressure field an actual sound source would produce, in a neighborhood of the listener's ears. One of the drawbacks of using HRTF's is that they have the pinna characteristics of the ears used to make the measurement, and so introduce the associated notches and peaks into the transfer function. The listener then has to ''in effect'' listen through these ears. And as pinna characteristics vary widely between individuals, the introduced notches and peaks associated with the measured pinnas may not correlate with the listener's. Thus by approximating the pressure field in the neighborhood of the ears, it is no longer necessary to create artifical peaks and notches in the transfer functions.

2:40

1pEA3. Spatial sound reproduction with wave field synthesis. Marinus M. Boone and Diemer de Vries (TU Delft, Lab. of Acoust. Imaging and Sound Control, P.O. Box 5046, 2600 GA Delft, The Netherlands, rinus@akst.tn.tudelft.nl)

Wave field synthesis is a reproduction technique developed at TU Delft, that enables the generation of high-quality three-dimensional spatial sound fields. The benefit of the method is that spatial impressions are highly independent of the position of the listeners within a large listening area. In short, the method uses a limited number of audio channels that are reproduced by generating plane and spherical wave fields with arrays of loudspeakers that surround the listening place. Applications include spatial sound reproduction in the home and in cinemas, sound reinforcement in theaters, teleconferencing with large video screens, and variable acoustics.

Contributed Papers

3:00

1pEA4. Comparison of virtual sound source positioning with amplitude panning and Ambisonic reproduction. Holger Strauss and Jörg Buchholz (Inst. of Commun. Acoust., Ruhr-Univ. Bochum, D-44780 Bochum, Germany, strauss@ika.ruhr-uni-bochum.de)

Different kinds of panning algorithms can be applied to render virtual sound sources in pantophonic or periphonic loudspeaker setups. One popular method is constant energy amplitude panning, where only the loudspeakers nearest to the virtual sound source are used for sound reproduction. Another well established reproduction method is Ambisonics, which uses the spherical harmonics theory in combination with knowledge from psychoacoustics to render virtual sound sources by simultaneously using all loudspeakers in a setup. Technical details and background information on both algorithms are given, and technical implementation issues (e.g., required computational effort) are discussed. A series of psychoacoustic listening tests was performed to evaluate virtual sound source localization with both reproduction methods for a six-loudspeaker pantophonic and an eight-loudspeaker periphonic setup. Localization accuracy and the subjects' reaction time were measured. The results show that the localization blur does not depend on the virtual sound source position for Ambisonics, while amplitude panning causes a high localization blue for virtual sources between the loudspeakers and a low localization blur for images close to the loudspeakers. Similar results were found in tests carried out to ascertain how dependent the reaction time is on the virtual sound source position.

3:20

1pEA5. Modification of electrodynamic loudspeakers for 3-D spatialization. E. Grigori Evreinov and V. Alexander Agranovski (Lab. for DIIS, Specvuzavtomatika Design Bureau, 44/5-13 Lenin St., 344038 Rostov-on-Don, Russia, asni@ns.rnd.runnet.ru)

The modification of electrodynamic loudspeakers was developed on a separation base of management cues by spatial and acoustic sound attributes. A loudspeaker has a single diaphragm and segmented surround, because a diaphragm embodiment in a kind of several adjacent parts evokes the losses and distortions at a sound reproduction of low frequencies. Surround's segments have an electrical control by their mechanical properties (a density). It allows the regulation of a velocity of the diaphragm fluctuations in appropriate zones. Thus a redistribution of a sound pressure (or of an acoustic energy) in the near-field or in the immediate neighborhood of pinna occurs according to the changes of spatial coordinates of sound objects. Offered construction provides a spatial localization of sound objects in the near field, even with using a single loudspeaker.

3:40

1pEA6. Sound focusing in reverberating rooms: The time-reversal approach. Sylvain Yon, Christian Dorme, and Mathias Fink (Lab. Ondes et Acoust., Univ. Paris VII/E.S.P.C.I., U.M.R. C.N.R.S. 7587, 10 rue Vauquelin, 75231 Paris Cedex 05, France)

Time reversal is already known as an efficient ultrasonic method to focus through inhomogeneous or multidiffusive media. The presence of pointlike reflectors in these media allows applications of this technique in fields like nondestructive testing, medical imaging, or underwater acoustics. An extension of this technique in audible range acoustics is presented. An array of 70 microphone/loudspeaker couples is used to refocus sound inside of a reverberating room. At a desired focal point, directivity patterns are measured and compared with those obtained by focusing with a cylindrical beamforming technique through the same antenna. Time reversal is shown to strongly improve the focal spot pattern (beamwidth and sidelobes level). These results are related to the ability of time reversal to compensate for reverberation and scattering induced in the studied room. The process acts as a spatio-temporally matched filter to the propagation

transfer function of the desired focal point through the room. It makes this technique an auto-adaptive focusing system to a random 3-D cavity. Theory, experimental results, and future applications will be described.

4:00–4:20   Break

4:20

1pEA7. Algorithm for the design of broadband, constant-beamwidth, point-element linear arrays with constant sidelobe level. Joseph B. Gaalaas and Elmer L. Hixson (Dept. of Elec. and Comput. Eng., Univ. of Texas, Austin, TX 78712)

An algorithm has been designed that allows one to calculate the individual element weights, as a function of frequency, required to achieve specified values of sidelobe level and half-power beamwidth. The arrays considered had linear geometry and an odd number of evenly spaced, point elements. The input parameters are the sidelobe level, half-power beamwidth, and the number of elements. For different values of the input parameters, the bandwidth that produced realizable solutions was determined numerically. Upon choosing values for the input parameters that yield realizable solutions, the weighting functions for each element are determined by an explicit calculation. An array may be combined with a scaled version of itself to increase the bandwidth. The element weights for an octave bandwidth broadside array were calculated using this algorithm and this array was combined with a scaled version of itself to extend the bandwidth to two octaves. Microphone and loudspeaker arrays based on these calculated weights were implemented with a digital signal processor. Measurements of the directivity patterns in an anechoic room were compared to the desired directivity patterns to assess the design procedure. [Work supported by NSF Graduate Research Fellowship and Uniden Endowed Thrust 2000 Fellowship.]

4:40

1pEA8. A sound transducer with a flat, flexible diaphragm working with bending waves. Daniela L. Manger (Manger Products, Industriestr. 17, D-97638 Mellrichstadt, Germany)

From the time of the idea to the finally working sound transducer, a period of over 20 years was necessary. The time has been needed for development and research in manufacturing and materials. It took a long time until new synthetic and magnetic materials were available with their best fitting mechanical behaviors. Nowadays it is possible to present a wideband sound transducer working from 100 Hz up to 35 kHz. It follows time precise without any mechanical energy storage the incoming signal. The special structure of the flat and flexible diaphragm works concentrically only with bending waves. A simple mechanical model, theoretical equations and measurements will be presented in comparison to the omnipresent Rice–Kellogg piston loudspeaker. The advantages in perception and hearing will be shown with regard to physiological behaviors of the hearing sense.

5:00

1pEA9. FEM simulations of horn loudspeakers and their experimental verification. Armin Jost and Reiner Kressmann (Inst. for Electroacoustics, Merckstr. 25, D-64293 Darmstadt, Germany, kre@uet.tu-darmstadt.de)

Horn loudspeakers are in common use for sound reinforcement of large rooms because of their high efficiency. In the design of horn loudspeakers, special care has to be taken on the horn's geometry and its interaction with the properties of the driver. In the present work, the finite-element analysis of these coupled acoustical–mechanical systems is compared with classical analytical and finite-difference methods. All simula-

tions are compared with experimental data, obtained from two different horn loudspeakers: First, a folded bass cabinet transducer and, second, a midrange horn loudspeaker. The bass cabinet is $50 \times 54 \times 104$ cm in size; the midrange horn measures $40 \times 44 \times 44$ cm. Both cabinets are made from wood; therefore, the horn's geometry can only be approximated to the ideal exponential characteristics. In general, the effect of finite horn length, resulting in reflections at the mouth due to mismatch in acoustical impedance, has to be considered. The bass horn, especially, has to be simulated very precisely, since this loudspeaker shows higher impedance mismatch. The finite-element simulations carried out with ANSYS show excellent agreement with the measurements. The sound-pressure level generated was investigated as a function of frequency and angle of radiation.

5:20

1pEA10. New numerical simulation tool for the design of electromagnetic transducers. Michael Schinnerl, Hermann Landes, Manfred Kaltenbacher, Reinhard Lerch (Dept. of Elec. Measurement Technol., Univ. of Linz, Altenbergerstr. 69, A-4040 Linz, Austria, michael.schinnerl@jk.uni-linz.ac.at), and Joachim Schöberl (Univ. of Linz, A-4040 Linz, Austria)

Due to the high geometric complexity and the interaction of different physical field types, like mechanical displacement, acoustic pressure, and magnetic induction, the design of electromagnetic transducers is a challenge for the developers. Since the fabrication of prototypes and experimental-based design is a lengthy and costly process, needs for appropriate numerical simulation tools arise. In this paper, a simulation program is presented which is especially adapted to multifield problems like the one described above. First, the description of the underlying physical fields (acoustic, mechanical, and magnetic) with partial differential equations (PDE) and their coupling is reported. The solution of these PDEs is based on either finite-element methods (FEM), boundary-element methods (BEM), or coupled FE-BE methods, depending on the considered problem type. However, these numerical techniques yield to long computer time, especially in the 3-D case. Therefore, a multigrid solver has been developed which enables considerably faster solutions of the presented numerical tasks. Finally, two application examples, the calculation of an EMAT (electromechanical acoustic transducer) in transmitting and receiving mode and the prediction of the sound field of a vibrating machine part, show the reliability of the simulations.

5:40

1pEA11. Determination of structure of the NARMAX model for dynamic loudspeaker using the orthogonal algorithm. Piotr Pruchnicki and Andrzej Dobrucki (Wroclaw Univ. of Technol., Inst. of Telecommunications and Acoust., Wyb. S. Wyspianskiego 27, 50-370 Wroclaw, Poland)

NARMAX is a general method for modeling of nonlinear systems. In the paper, an application of this method for modeling of nonlinearities in dynamic loudspeakers is presented. In most cases, a polynomial representation of the NARMAX model causes an ill-conditioned equation system which indicates that the use of orthogonalization is necessary to solve the system. The high number of coefficients is the main disadvantage of the NARMAX method, and it results in an erroneous model, and an algorithm for identification of coefficients that is noneffective. A modification of the orthogonal algorithm (classical Gram–Schmidt) is presented. It is based on the premise that the coefficient providing the highest error reduction ratio is chosen in every step of the algorithm. The creation of the model stops when the given accuracy is achieved. This way leads to the choice of only the most significant coefficients. The work and effectiveness of this modified algorithm for systems with given nonlinearities at different levels of accuracy of the model are presented in the paper.

6:00

1pEA12. A proposal to adopt Maxwell distribution as a measure of acoustic field diffuseness in a reverberation room. Matias Han, W. Budhiantho (Dept. of Elec. and Computer Eng., Satya Wacana Christian Univ., Jl. Diponegoro 52-60, Salatiga 50711, Indonesia), and Elmer L. Hixson (Univ. of Texas, Austin, TX 78712)

The existence of an acoustic velocity vector resultant field which follows Maxwell distribution in a reverberation room indicates 3-D diffuseness of the field. Therefore, Maxwell distribution is a better measure for the sound-field diffuseness than Gaussian distribution given by the acoustic pressure in the same room that does not have a directional information. A Gaussian distributed pressure field does not always give a satisfactory diffuseness. The acoustic velocity vector resultant was measured using total energy density sensor. Theoretical and experimental investigations are given to support this proposal.

Session 1pED

Education in Acoustics: Acoustics Education 2000

Daniel R. Raichel, Cochair

Department of Mechanical Engineering, City College of New York, 140 Street and Convent Avenue, New York, New York 10031, USA

Roberto Pompoli, Cochair

Engineering Department, University of Ferrara, Via G. Saragat 1, 44100 Ferrara, Italy

Chair's Introduction—1:55

Invited Papers

2:00

1pED1. Acoustics education in France: Past, present and future. B. Hamonic　(bernard.hamonic@eurobretagne.fr)

Acoustics is the science of sound. The interdisciplinary nature of acoustics implies the broad scope of acoustics education and the necessity to teach it at various levels of the student education, from high school to Ph.D. In the 1960s acoustics was not an important part of the French education system. Progressively teaching acoustics at a higher education level became natural and, in the 1990s, France was the only country to give a diploma between six and eight years after the secondary studies leaving certificate (Bac in french). The quality of this diploma was directly related to the emergence in France of research teams specialized in acoustics. Teaching acoustics at a secondary level came into effect only in 1994 for senior high school students. This presentation will first summarize the current acoustics education system in France with its characteristic in secondary education (senior high school) and higher education, univerities (first, second, and third cycle), engineering schools (French typcial system), etc. The future of acoustic education seems directly related to computer-based courses, the emergence of videotapes and videodiscs demonstration experiments, internet communication, etc. Reflection toward teaching acoustics in France in the future will then be addressed.

2:20

1pED2. Survey of the university institutions teaching acoustics in Poland. Edward Hojan　(Inst. of Acoust., Adam Mickiewicz Univ., PL-60-769 Poznań, Matejki 48/49, Poland)

Higher education in Poland is either state owned or private. The total number of students in the institutions of higher education at the university level amounts to 630 000, which makes 15 students per 1000 citizens. As a rule, the education provided by state-owned schools is free. Among the state-owned educational institutions, presently in Poland, there are 13 universities, 18 technical universities, 8 musical academies, and 12 medical academies in which acoustics (or only psychoacoustics) is learned at selected branches of the studies The subject of acoustics is not provided in private schools. In this survey a directory of schools will be given with the details of studies where acoustics is available, including the numbers of students, hours, and the topics taught in the lectures and tutorials.

2:40

1pED3. Acoustics education in the Low Countries. Adrianus J. M. Houtsma　(IPO, P.O. Box 513, 5600 MB Eindhoven, The Netherlands)

This presentation is an overview of acoustics education programs found at universities and professional colleges in the Netherlands and Flanders. As has happened in many other academic areas, the traditional organization of educational programs along disciplinary lines has to a large extent made a place for an organization based on application areas. In university programs on acoustics, one finds clusters of specialized subjects that seem to be typical for schools or departments, such as audiology, audiometry, and speech therapy in medical schools, phonetics, music, and speech in humanities departments, and architectural acoustics, noise control, imaging, and fundamentals in applied sciences or engineering departments. Several professional colleges offer courses that are focused on measurement techniques, insulation properties of materials, and noise control in buildings and structures. A fundamental and broad treatment of acoustics, followed by a treatment of specialized applications in the areas of noise control engineering, architectural acoustics, or electroacoustics as elective options, is found in the Advanced Acoustics Course, organized by the Dutch and the Belgian acoustical societies and taught annually in Antwerp.

3:00

1pED4. State and trends of acoustics education in Germany. Peter Költzsch　(TU Dresden, Dept. of Elec. Eng., Inst. of Tech. Acoust., Mommsenstraße 13, D-01062 Dresden, Germany, peter.koeltzsch@iname.com)

An overall view of the state of the education in the field of acoustics in Germany, Austria, and Switzerland is given in ''Guide to Acoustics Studies'' (2nd ed., July 1998, edited by DEGA). In Germany, acoustics is taught at 63 institutes (23 universities, 16 universities of technology, and 3 universities for music, art film, or television). In connection with education, the following problem

is of great interest: How is the discipline of acoustics integrated into the traditional faculties of German universities? The present state is: integration of acoustics in mechanical, electrical, environmental engineering *et al.* On the other hand, a characteristic problem also is the existence of a ''strong'' acoustics institute at a university with a considerable research and education potential and, on the other hand, smaller special acoustics research and education capacities in all those scientific disciplines in which acoustics plays an important role. Here the question emerges whether it is more expedient for the development of acoustics to ''keep'' so-called ''acoustics pets'' in several faculties or to found a central acoustics institute to satisfy all requirements at the institutes and at the university in all. However, into which faculty should this ''acoustics institute'' be integrated then?

3:20

1pED5. Acoustics education in Ukraine. Stanislav M. Mayevskyy and Leonid M. Gelman (Dept. of Nondestructive Testing, Natl. Tech. Univ. of Ukraine, 37, Peremogy pr., Kiev, 252056, Ukraine)

Acoustics education in Ukraine is considered. In more than 40 universities, students learn acoustics: in the acoustics department of the National Technical University in Ukraine (only one department in Ukraine) and many related departments such as Nondestructive Testing, Physics, Electrical Engineering, etc. Acoustical specializations of departments are presented. The most promising and developing acoustical specialization is biomedical acoustics.

3:40–4:00   Break

4:00

1pED6. Education in acoustics: The Italian experience. Roberto Pompoli (Eng. Dept., Univ. of Ferrara, via G. Saragat 1, 44100 Ferrara, Italy, rpompoli@ing.unife.it)

Education in acoustics is a topic which has been discussed for many years in different international conferences: ICA, Inter-Noise, Forum Acusticum, etc. Tor Kihlman, analyzing in two papers [Proc. ICA'95, 311–314; Proc. Inter-Noise'96, 87–90] the education programs for students in technical faculties, arrives at the conclusion that two kinds of curricula are needed: one for students who are not specializing in acoustics and another one for students who are specializing in this field. He proposes some strategies to be followed in order to obtain better results when teaching acoustics in the technical faculties. Starting from these considerations, and taking into account on one hand the experience of education in acoustics in Italy, and on the other the analysis of the data collected by the author (in charge of education for the EAA), some possible solutions are proposed for a better coordination of the programs betweens the institutions and for the establishment of a European network of experts and teachers who are willing to move to different universities for short periods. The project could be coordinated by the EAA and funded by the European Commission.

4:20

1pED7. Education on acoustics in Spain and Latin America. Manuel Recuero (Departamento de Ingeniería Mecánica y Fabricación, Universidad Politécnica de Madrid, Ctra. Valencia Km. 7, 28031 Madrid, Spain, mrecuero@diac.upm.es), Constantino Gil, and Antonio Minguez (Universidad Politécnica de Madrid, 28031 Madrid, Spain)

Education in acoustics was incorporated as a part in engineering education in 1969, and from then to today it has had several transformations. In this paper, the results obtained from this education during the last years and how it is today are presented. Also presented are the conclusions reached from the theoretical and experimental knowledge points of view in the last three decades. The present situation of this type of education in the Spanish countries in South America are also expounded: development, implementation, years of teaching, etc. The paper is a summary of the education in acoustics in the Spanish universities with their curricula, laboratories, facilities, and so on, and the premises for the future.

4:40

1pED8. UK education in acoustics. C. G. Rice (Inst. of Sound and Vib. Res., The University, Southampton SO17 1BJ, UK, cgr@soton.ac.uk)

To definitively report on the status of acoustics education in the UK must be regarded as a challenging task. The areas of interest are even wider than the normally accepted classification of acoustic subjects (e.g., PACS), and the variety of situations to which such knowledge can be applied in practice is daunting. However, whatever the level of expertise required of practitioners in acoustics, everyone ought to be educated within a framework of life-long learning and continuing professional development. In the UK such formal educational and training requirements needed to practice the profession of acoustics have been specified, and it is in this context that UK education in acoustics will be reviewed. Formal educational requirements leading to chartered engineer status for individuals (e.g., CEng and FEANI EurIng) have been laid down by the UK Engineering Council in their ''Standards and Routes to Registration (SARTOR)'' document. These procedures were compiled by members from the professional engineering institutions and include guidelines for the accreditation of university degree courses. The Institute of Acoustics plays an important role in the implementation of such procedures. Information relating to the wider role of the importance of acoustics education in nonengineering disciplines will also be presented.

5:00

1pED9. Education in acoustics: Fundamentals and peculiarities. Nikolai A. Dubrovsky (Andreyev Acoust. Inst., #4 Shvernika Ul., Moscow 117036, Russia), Albert A. Dogadov, and Alexandr S. Sigov (Moscow State Inst. of Radiotechnology, Electron. and Automation, Moscow 117454, Russia)

An historical background of university and technical high school education in acoustics in Russia is considered. Basic features of Soviet concepts of training in acoustics are discussed. Main advantages are deep and broad training in basic and applied disciplines, a system of the ''base'' research centers for early research activity of students under leadership of active scientists, and reporting by students of their results in science and technology at conferences and seminars. These advantages had been very positive for the Soviet planned economy. The main drawbacks are redundancy and rigidity of this education system. Transition to free market society forced the modification of basic concepts of education in acoustics: to introduce flexible education similar to European and American ones, to train students in new areas of acoustics intended to enhance the quality of life (ecology, medicine), to provide optional training in acoustics (common or individual), to meet requirements of private enterprises to education of selected students. A new approach to training of foreign students is developed: transition from special high schools for foreigners to joint education of foreigners with Russian citizens in many state educational centers. The role of the Russian Acoustical Society in exchange of education methodology between training centers in Russia is presented.

5:20

1pED10. Acoustics: A route to science literacy in the 21st Century. Thomas D. Rossing (Phys. Dept., Northern Illinois Univ., DeKalb, IL 60115)

Two trends have been apparent during the second half of the 20th Century: the disappearance of acoustics from physics curricula and the decline in the percentage of students who take physics, both in schools and universities. The relationship, if any, between these trends is well worth discussing. More intriguing, however, is the likelihood that acoustics, with its rather wide appeal to students and to the public, can serve as a vehicle to promote science literacy.

MONDAY AFTERNOON, 15 MARCH 1999　　　　　　　　　　　　　ROOM MA004, 2:00 TO 6:20 P.M.

Session 1pMU

Musical Acoustics: Sound Production of Wind Instruments

William J. Strong, Cochair
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D. Murray Campbell, Cochair
Department of Physics and Astronomy, University of Edinburgh, Edinburgh EH9 3JZ, UK

Invited Papers

2:00

1pMU1. A comparison of wind instrument time-variant spectra. James W. Beauchamp (School of Music and Dept. of Elec. and Comput. Eng., Univ. of Illinois at Urbana–Champaign, 2136 Music Bldg., 1114 W. Nevada, Urbana, IL 61801, j-beauch@uiuc.edu)

Musical instruments can be compared in terms of the sound spectra that they produce. First, they may be compared in terms of spectral envelope, which allows an observation of change of amplitude with respect to frequency. Moreover, the spectra change with respect to time, usually with a burst of low-amplitude, broadband spectrum initiating the attack phase, followed by a gradual increase of spectral centroid. The so-called ''steady state'' may actually include some interesting changes, depending on performance expressivity. The final decay phase is perhaps the least manipulable by the performer, and is generally characterized by a gradual lowering of the centroid as the amplitude decreases. A centroid-indexed family of spectral envelopes based on data for several tones with different fundamental frequencies gives a good average picture of the spectra for a particular instrument. However, this representation fails to take into account special details such as pitch-dependent and harmonic-dependent spectral features. While such special details are least important for the brass, they are very important for the woodwinds, and several features of the clarinet, oboe, saxophone, and flute will be illustrated and compared.

2:20

1pMU2. Sound production of flue organ pipes. Andras Miklos and Judit Angster (Fraunhofer-Institut Bauphysik, Nobelstr. 12, D-70569 Stuttgart, Germany, angster@ibp.fhg.de)

For understanding the sound production of flue organ pipes, the acoustic properties of the two main components, the acoustic resonator and the air jet generator at the labium (flue + upper lip), have been investigated separately. It was found that the formant structure of the stationary sound spectrum is mostly determined by the geometry of the pipe, including the location, size, and shape of the openings (mouth and open end), while the attack is strongly influenced by the air jet generator. An influence of the wall

vibrations on the stationary sound has been also found, but this effect could be neglected until the wall is not too thin. The spectra of the stationary sound radiated by the openings are markedly different. The analysis of their formant structures implies that the source of the radiation is the acoustic flow at the openings. Experiments performed on labium models and pipes with damped acoustic resonators show that an edge tone with characteristic spectrum appears first during the attack. Its sound field may excite the acoustic resonator purely acoustically, preceding the development of the coupling between jet and acoustic oscillations. This assumption has been supported by the experiments discussed in the paper.

2:40

1pMU3. Sound production in flue instruments: A review. Benoit Fabre, Marc-Pierre Verge (Laboratoire d'Acoustique Musicale, Univ. Paris VI, 11 rue de Lourmel, 75015 Paris, France, fabreb@ccr.jussieu.fr), and Avraham Hirschberg (TUE, 5600 MB Eindhoven, The Netherlands, A.Hirschberg@phys.tue.nl)

The introduction of computer implementation of numerical methods, 40 years ago, has opened the way towards quantitative modeling of the sound production in flue instruments. This has induced a first wave of new research initiated in the late 1960s by the precursor work of Bechert, Cremer, Ising and Coltman. The breakthrough of real-time sound synthesis in the last decade has allowed the judgment of models on the basis of the sound which they produce. With this breakthrough ''verbal'' modeling such as the controversy between Helmholtz and Rayleigh about the nature of the sound source at the labium (volume source or dipole) is now a historical curiosity. However, it has became clear that lumped models as used in sound synthesis will remain semi-empirical carica-tures. It has appeared that direct numerical simulation of the details of the flow including the effect of turbulence is still impossible. One observes, therefore, a shift back towards experimental studies such as that of jet response to acoustical excitation, the effect of nicks, the effect of chamfers, the influence of materials, Coanda effect induced by turbulence, etc. A review of these recent studies will be presented.

3:00

1pMU4. A Rayleigh wave model for the lip reed: Qualitative aspects. R. Dean Ayers (Dept. of Phys. and Astron., California State Univ., Long Beach, 1250 Bellflower Blvd., Long Beach, CA 90840, rdayers@csulb.edu), Richard P. Birkemeier, and Lowell J. Eliason (California State Univ., Long Beach, CA 90840)

The stroboscopic technique of Daniel W. Martin [J. Acoust. Soc. Am. **13**, 305–308 (1942)] has been updated by using a video camera. Improved magnification and resolution show that the upper lip's motion is complicated: a Rayleigh wave propagates down-stream in a thin, passive layer of flesh. Muscles of the embouchure are used to adjust the tension and dimensions of that layer. The lips of accomplished performers show a wide range of detailed behaviors, due in some cases to asymmetry in the lip structure. The fact that players must find their own particular techniques for efficient control of the Rayleigh wave is significant for brass pedagogy. Seeing their lips in motion may help beginning players to advance more rapidly. Results of a preliminary survey among accomplished performers and beginners will be presented. [Work supported in part by the Scholarly and Creative Activities Committee and the College of Natural Sciences and Mathematics at CSULB.]

3:20

1pMU5. Self-excitation in woodwinds: From a particular model to a generic model. Ana Barjau, Vincent Gibiat (Dept. of Mech. Eng., Polytech. Univ. of Catalunya, Diagonal 647, 08028 Barcelona, Spain), and Vincent Gibiat (ESPCI, 75231 Paris Cedex 05, France)

The process of sound production in a double-reed woodwind is based upon two mechanisms: the Bernoulli mechanism, which is associated only with the reed and air dynamics, and the coupling mechanism, which represents the interaction between the bore and the reed. A realistic model, close to the physical reality of the system, leads to satisfactory simulated results compared to experimental measures. However, the great number of parameters and variables appearing in the mathematical formulation makes it difficult to predict the final behavior of the system. In this paper, the realistic model is presented, and progressive simplifications are done to reduce the formulation to a single equation containing a few control parameters and a single variable. The progressive models are discussed and applied to simulate the internal pressure at the input section of the instrument. Different behaviors are then evident: from more traditional ones (periodic states) to quasiperiodic and chaotic signals.

3:40

1pMU6. Some comments on the artificial blowing of the reed woodwind instrument. Tohru Idogawa (1-14-39 Gakuen-cho, Higashi-kurume, Tokyo 2030021, Japan, RXE07320@nifty.ne.jp)

Important results were obtained by artificial blowings of the reed woodwind instruments: Many states of air column vibrations can be excited under a fixed fingering and a fixed embouchure; these states have their respective regions of blowing pressure in which frequency and waveform of the air column vibration change slightly and continuously as the blowing pressure is varied; a sudden transition from one state to another takes place irreversibly at a blowing pressure, the upper (or lower) boundary of the state, when

blowing pressure is increased (or decreased) gradually. It will be presented that the above-described results are not necessarily valid for a conical brass pipe (0.63 m long), which has the same apex angle as that of the bassoon, blown by the mouthpiece of the soprano saxophone. One of the vibratory states excited has changed to another state without transition. When a cylindrical brass pipe, which has about the same length as that of the clarinet, has been artificially blown with the clarinet mouthpiece, the reed has been easily broken, which has rarely been observed for the clarinet.

4:00–4:20   Break

4:20

1pMU7. Differences between cylindrical and conical reed instruments. Jean-Pierre Dalmont (Lab. d'Acoust. de l'Univ. du Maine, UMR CNRS 6613, 72085 Le Mans, France)

Most investigations on reed instruments have been carried out on clarinet probably because this is the most common reed instrument. However, the extrapolation of the results to conical instruments is risky. The fact that internal clarinet spectrum does not contain even harmonics is not the only difference. For example the transfer function between input and output pressure is very different: this makes a saxophone, for lower notes, much louder than a clarinet. On the other hand, application of linear analysis near the threshold of oscillation to conical reed instruments is limited: because of the inverse bifurcation, small oscillations do not exist for conical reed instruments. This is confirmed by the observation that it is clearly much easier to play pianissimo with a clarinet than with a saxophone or an oboe. The saturation mechanism seems also to be different for conical and cylindrical instruments. On a saxophone the saturation occurs at a lower mouth pressure because of the existence of more than one regime at the fundamental frequency. All these aspects will be discussed with reference to simplified physical models and illustrated by experiments using artificial blowing. [This work has been done in collaboration with C. J. Nederveen, J. Gilbert, and J. Kergomard.]

Contributed Papers

4:40

1pMU8. A complete period-doubling cascade on a cromorn. Vincent Gibiat (LOA/ESPCI CNRS UMR 7587, 10 rue Vauquelin, 75231 Paris Cedex 05, France, Vincent.Gibiat@espci.fr) and Michele Castellengo (Univ. Paris, Paris, Cedex 05, France)

The cromorn is a very simple double reed instrument of the Renaissance which shares with the recorder the valuable property that there is no contact between the musician's lips and the double reed, allowing one to study it without any human intervention and so to control the parameters of the sound production. This has been studied by varying the blowing pressure from small values to the pressure needed to obtain the normal sound of the instrument and then by decreasing it. The sound has been recorded and analyzed in time frequency domain. A complete cascade of period-doubling bifurcations has been obtained following the increase of the pressure. Two chaotic regions have been found, one just after the third period doubling and another after a period tripling. For the lowest pressure in the decreasing process the coupling of the reed and the bore becomes too weak and the reed squeaks. The instrument is simple enough to follow the schematic given by Maganza *et al.* for cylindrical instruments and it will be shown that the decrease of the pressure is equivalent to the sharpening of the nonlinear function which relates pressure and flow that gives the possibility of period doublings.

5:00

1pMU9. The motion of air-driven free reeds. James P. Cottingham, C. Joseph Lilly, and Christopher H. Reed (Phys. Dept., Coe College, Cedar Rapids, IA 52402, jcotting@coe.edu)

An earlier study on free reeds from reed organs [P. D. Koopman and J. P. Cottingham, Reed Organ Soc. Bull. **15**, No. 3–4, 17–23 (1997)] yielded information about the variation in amplitude of reed vibration with pressure, but few further details on the reed motion. A more thorough set of measurements of the motion of these reeds has been done, including the use of a variable impedance transducer to measure reed displacement and a laser vibrometer system to measure reed velocity. For reeds from reed organs, at low to moderate blowing pressures, the reed motion at each point is nearly sinusoidal and the reed profile at maximum amplitude closely approximates that calculated for a cantilevered beam vibrating in the first mode. At blowing pressures considerably higher than normal playing pressure, strong indications of the presence of the second beam mode are observed. The nearly sinusoidal reed motion at normal pressure contrasts with the sound pressure waveform near the reed which, as a result of the valve action of the reed, roughly resembles a pulse wave. Similar measurements on accordion reeds yield somewhat contrasting results.

5:20

1pMU10. A new algorithm to simulate nonlinear propagation in a tube of any shape. Application to a real-time physical model of trumpetlike instruments. Christophe Vergez and Xavier Rodet (IRCAM, Paris, 1 pl. Igor Stravinsky, 75014 Paris, France)

When air pressure is low enough, wave propagation in a pipe can be considered linear to a reasonable degree. When the pressure level increases, this simplification becomes more and more inaccurate since high pressure values travel faster than low pressure values. The speed of propagation is thus a function of the pressure level. A new time-domain approach is proposed to simulate nonlinear propagation of a sound wave in a tube of any shape. This constraint-based algorithm does not prevent the derivative of the signal from breaking, as it happens when a shock wave occurs. Moreover, the algorithm allows the carrying out of nonlinear propagation even once a shock has occured. Thus, as expected the distorsion of a sinusoidal wave into a N-wave of progressively decreasing amplitude was observed. Moreover, without any additional computational cost, this technique provides fractional delay variation as needed for continuous tube's variations in length. The nonlinear propagation algorithm is included in the real-time high-quality physical model of trumpetlike instruments. Simulation shows a great improvement in the sound quality since this algorithm allows subtle nuances in the timbre as well as hard brassy effects.

5:40

1pMU11. Blowing and breathing in the woodwinds: Generation and control of input parameters. Leonardo Fuks (UFRJ–Rio de Janeiro Federal Univ., School of Music, Rua do Passeio 98, Rio de Janeiro, 20021-290, Brazil, leonardo@speech.kth.se)

Wind instruments, being more than an extension to the performer's body, are strongly connected to the respiratory system. Such a relationship permits a multiple and fine control of input parameters, enabling wide ranges of tone variation and expressiveness, but also imposes constraints concerning muscular demands for aerodynamic energy supply, breathhold times, and lung capacity, among others. The present investigation combined physiological and acoustical approaches, focusing on reed woodwinds under naturalistic playing conditions. Airflow and blowing pressure used by professional players were shown to be systematic in rehearsed tasks and to depend on pitch and dynamic level, instrument type, and reed characteristics, suggesting that musical training involves the incorporation and adjustment of the mechanical properties of the system to the musician's technique. Blowing pressure sensation was assessed by a psychophysical method, showing a quasi-linear behavior. Respiratory movements, decomposed into rib cage and abdominal contributions, have been recorded and analyzed during performance by an inductive plethysmo-

graphic method, providing insights to the musicians' use of the respiratory system. The ever-changing pulmonary air composition was shown to affect fundamental frequency to some extent. [The research work presented in this paper was carried out at KTH, Stockholm, and was sponsored by a Brazilian CAPES scholarship.]

6:00

1pMU12. Swiss horns and their harmonicity. Rolphe F. Fehlmann (Swiss Software Eng. College, Dept. of CAD/CAM Eng., Morgartenstr. 2, CH-3012 Berne, Switzerland)

Earlier it has been shown that there exists an optimal curved waveguide for musical purposes exhibiting a sequence of harmonic resonance modes [R. F. Fehlmann, in Intl. Congress on Acoustics, III (1995), pp. 417–421]. The radiation of some ekmelic overtones even consists of elliptical waves [R. F. Fehlmann, J. Acoust. Soc. Am. **103**, 2774 (A) (1998)]. According to the hypothesis of Risset/Mathews [J.-C. Risset and M. V. Mathews, Phys. Today **22**(2), 23–30 (1969)], the ''warmth'' of a sound source is due to the inharmonicity of the musical tones. However, in this paper, it is shown that this is not the case for the well-known ''warm and melancholic'' sound of the Swiss horns and thus questions the reasoning of Risset/Mathews.

MONDAY AFTERNOON, 15 MARCH 1999 ROOM H107, 1:55 TO 6:00 P.M.

Session 1pNSa

Noise: Urban Noise and Soundscapes

Joseph Pope, Cochair
Pope Engineering Company, P.O. Box 236, Newton Centre, Massachusetts 02459, USA

Brigitte Schulte-Fortkamp, Cochair
Acoustics/Physics, University of Oldenburg, D-26111 Oldenburg, Germany

Chair's Introduction—1:55

Contributed Papers

2:00

1pNSa1. Trading level for number of aircraft immissions: A full-factorial laboratory design. Joachim Vogt (Univ. of Dortmund, Dept. 14, Organisational Psych., D-44221 Dortmund, Germany, VOGT@WAP-MAIL.FB14.UNI-DORTMUND.DE) and Karl Theodor Kalveram (Univ. of Duesseldorf, D-40221 Duesseldorf, Germany)

To what extent can the number of overflights be increased without enhancing annoyance, if airplanes become softer? An original B737 takeoff sound (A-weighted sound-pressure level: $L_{max} = 76$ dB) was either reduced or amplified by 4.8 dB. In a 3×3 factorial design, nine groups of ten subjects were exposed for 27 min to 3, 9, or 27 copies of one of these three sounds. Regarding energy equivalence defined by $L_{eqA} = $const, a 4.8-dB sound-pressure reduction compensated for three times the number of the higher level, achieving constant Leq levels in the diagonals of the design. After noise exposure, cardiovascular responses and the subject ratings of loudness, annoyance, and quality of an imagined living area with comparable noise load were analyzed statistically. Comparison of the equal-energy conditions revealed no difference regarding heart rate and blood pressure, but subjective loudness increased with both level and number, and 27 soft aircraft were rated extremely loud. However, annoyance decreased, when 3 loud overflights were converted to 9 medium events, but increased again under 27 soft events. Imagined living quality always decreased with increasing numbers of events, and could not be compensated for by energy-preserving reduction of single-event levels.

Conclusion: Beyond 10 events per 30 min the number probably has an increasing over-energetic impact on subjective noise effects.

2:20

1pNSa2. Calculation of noise contours for Belgian airports. Elisabeth M. J. Peeters, Jan Thoen (Laboratorium voor Akoestiek en Thermische Fysica, K. U. Leuven, Celestijnenlaan 200D, 3001 Leuven, Belgium), and Ghislain Geusens (Regie der Luchtwegen, 1030 Brussels, Belgium)

Noise contours are an objective way to represent the long-term noise load from departing and arriving aircraft. Many major European airports have a long history of noise contour calculations and related land use planning. For Belgian airports, several noise contour calculations have been made since the 1970s, but have always been based on limited information. Recently, noise contours for the airports of Brussels, Antwerp, and Ostend have been calculated based on detailed flight information. Longterm noise measurements around Brussels airport validated the accuracy of the Brussels airport noise contours. Up to today, neither for the development of noise contours, nor for the land use planning based on airport noise contours, official guidelines or practices exist, and new houses can still be built around the airports. On the other hand, with international harmonization efforts in mind, Belgian airports are not, as some other European airports restrained by legislation, based on dated technology noise contours, but have taken the opportunity to choose noise descriptors and calculation methods based on current evolutions in the international ways of calculating noise contours, as well as on international guidelines.

2:40

1pNSa3. Normalized noise spectra of aircraft take-off and landing operations. J. Salvador Santiago and Jose Pons (Instituto de Acustica, C.S.I.C.-c/Serrano 144, 28006 Madrid, Spain, ssantiago@fresno.csic.es)

At Madrid-Barajas Airport, a new runway will enter into service in the year 1999, located to the NW of the two existing ones. This fact implies a new configuration of flight paths, both for take-off and landing operations, and a new definition of the areas adjacent to the airport as regards the noise impact on the population. An ad-hoc regulation states that the dwellings within the area covered by the isophones of $L_{eq\ day} = 65$ dB and $L_{eq\ night} = 55$ dB, for the future situation in the year 2010, have to be supplemented by the airport operator with an acoustic insulation to achieve an internal noise level of $L_{eq\ day} = 35$ dB and $L_{eq\ night} = 30$ dB. The noise spectra that will be used for measuring the actual insulation of the buildings have been determined along the isophones $L_{eq} = 80$ dB to $L_{eq} = 55$ dB, in steps of 5 dB, for take-off and landing operations, by means of *in situ* tape recordings of 938 aircraft operations, recorded simultaneously in six measuring stations. Out of 5245 valid recordings, 2749 with enough signal-to-noise ratio were analyzed in 1/3 octave bands, and the results presented as spectra normalized to 0 dB.

3:00

1pNSa4. Noise problems, savage approaches. From ''just forget it,'' to physical violence. Fernando J. Elizondo-Garza (Laboratorio de Acustica, UANL-FIME, P.O. Box 20 ''F,'' Cd. Universitaria, San Nicolas, 66450, N. L., Mexico, fjelizon@ccr.dsi.uanl.mx)

Different approaches and/or philosophies that can be used to face a noise problem are described and commented on. Both the classical noise control procedures and the savage approaches that go from simply ''forget it,'' to physical violence, are considered.

3:20

1pNSa5. Can population density be used to determine ambient noise levels? Catherine M. Stewart (Oak Ridge Inst. of Sci. and Education (ORISE), Postgrad. Internship Prog., U.S. Army Ctr. for Health Promotion and Preventive Medicine, Environ. Noise Prog., 5158 Blackhawk Rd., Aberdeen Proving Ground, MD 21010-5422), William A. Russell, Jr., and George A. Luz (U.S. Army Ctr. for Health Promotion and Preventive Medicine, Environ. Noise Prog., Aberdeen Proving Ground, MD 21010-5422)

In 1974, the U.S. Environmental Protection Agency (USEPA) published a report endorsing the day–night average sound level (DNL) for nationwide use. Included in this report was an equation predicting DNL from the number of people per square mile. The equation was based on monitoring from 100 sites throughout the U.S. To determine whether the 1974 equation still provided a realistic estimate 25 years later, monitoring was conducted at 50 sites in Baltimore County, Maryland. Population density ranged from 600 to 13 000 people per square mile. To explore seasonal differences and test–retest reliability, the monitoring was conducted for 24 h in the summer and 24 h in the winter. The results are compared to those of the USEPA.

3:40

1pNSa6. Development of a curve to predict community annoyance due to transportation noise exposure. Lawrence S. Finegold (U.S. Air Force Res. Lab., Wright–Patterson AFB, OH 45433, LFinegold@falcon.al.wpafb.af.mil)

This paper presents a brief review of the history of the development of a curve to predict community annoyance in response to transportation noise sources beginning with the original ''Schultz curve'' [J. Acoust. Soc. Am. **64**, 377 (1978)]. This seminal article provided the results of a large-scale database meta-analysis (i.e., a secondary analysis of previously published data) using data from the 11 published transportation-elated community annoyance social surveys. The Schultz curve, which relates general transportation noise exposure (in terms of day–night average

sound level, DNL) and the percent of a community ''highly annoyed'' (%HA) has been used for decades in environmental impact analyses in much of the world. The community annoyance database used by Schultz has been updated with additional social survey data, nearly tripling its original size in the initial Fidell *et al.* [J. Acoust. Soc. Am. **89**, 221–233 (1991)] analysis. A more recent version of this meta-analysis by Finegold *et al.* [Noise Control Eng. **42**(1), 25–30 (1994)] differs from the Fidell *et al.* analysis on several technical issues and includes 400 data points from 22 international social surveys. The results of the Finegold *et al.* analysis have recently been adopted as part of a U.S. standard on environmental noise (ANSI S12.9-1996/Part 4). This paper reviews the details of this meta-analysis.

4:00–4:20   Break

4:20

1pNSa7. A field study about the effects of low-frequency noise on man. Ana M. Verzini, Carlos A. Frassoni, and Aldo H. Ortiz (Centro de Investigaciones Acusticas y Luminotecnicas UNC, Cordoba, Argentina)

During the last few years, several shopping centers have been placed in residential neighborhoods of Cordoba City, and inhabitants' complaints about noise have been reported. The present paper is a field study about the effects produced by very low-frequency noise on people's health and quality of life. It was carried out in different zones of Cordoba City, where considerable levels of noise in the frequencies below 163 Hz and also in infrasounds had been detected. The generating sources were air conditioning systems, industrial plants, and traffic noise. One hundred people between 21 and 65 years old were surveyed, and the measurements of noise were done in front of each interviewed person's home, with a B&K analyzer, type 2144 and microphone B&K type 4193. Noise levels in dBLin, dBA, dBLin below 163 Hz and dBLin below 20 Hz were found, and due to the multivaried characteristics of the research and the categorical variables used, data from the questionnaire and noise levels were analyzed by means of the multiple correspondence factorial analysis. Several relationships were found among type of noise, sociodemographics, and moderator variables, and effects of noises. [This study was supported by a grant of CONICET.]

4:40

1pNSa8. How information about the source influences noise annoyance. Karl T. Kalveram, Juergen Dassow (Univ. of Duesseldorf, Inst. of General Psych., Universitaetsstr. 1, 40225 Duesseldorf, Germany), and Joachim Vogt (Univ. of Dortmund, Dortmund, Germany)

Annoyance following noise exposure can be considered to convey a ''possible loss of fitness signal'' (PLOF-signal), indicating that the individual's Darwinian fitness decreases if she or he continues to stay in that situation. Especially, nonfamiliar conspecifics appearing in the habitat diminish fitness of the inhabitants because they are going to use the same but restricted resources. Therefore, sounds carrying the information that they are manmade are likely to evoke more annoyance than other sounds of equal level and spectral density. In an experiment, subjects were exposed to recorded sounds of ocean surf and party murmur. Both sounds were carefully equalized regarding spectral energy and overall level [$L_{eq(A)} = 52$ dB]. In the ''manmade'' sound condition, subjects felt significantly more annoyed and were significantly more impaired in a free recall memory test. However, physiological stress indices (potassium/sodium measured in saliva) did not discriminate significantly between the conditions. The results support the hypothesis that considered biologically, the main function of noise annoyance is to warn a person that fitness may diminish, but not to induce actual stress. This explains the frequently reported finding that noise, although annoying, causes only little or even no physiological stress reactions.

5:00

1pNSa9. Use of immersive sound in a museum exhibition. Peter Coulter (Powerhouse Museum, P.O. Box K346, Haymarket, Sydney, Australia 2000, peterc@phm.gov.au)

Studies have shown [Schaffer, World Sound Project (1997)] that in our environment, which includes our community, traffic and motorized sounds are growing at a faster rate than nonmotorized human sounds, suggesting that because motorized sound is low-information, high-intensity, it tends to desensitize the populace and results in less social interaction. One role of museums is to preserve a balance of sounds, as a sanctuary from our environment, and explore the consequence of our motorized culture. At the Powerhouse Museum, an exhibition has opened, December 1998, entitled ''Cars and Culture'' which examines the consequence of cars. In the exhibition there is an immersive sound space that has been created using an ambisonic sound system. This soundscape is discussed. An experiment was devised where the degree of sound immersiveness was varied for a number of visitors and their reactions were measured.

5:20

1pNSa10. Loudness evaluation of urban soundscapes by a cross-modal matching method. Patrick Susini (Inst. de Recherche et de Coordination Acoust./Musique (IRCAM), 1 place Igor-Stravinsky, F-75004 Paris, France, susini@ircam.fr) and Valerie Maffiolo (Lab. d'Acoust. Musicale, Paris Cedex 05, France)

In a previous study, loudness of a time-varying pure tone was evaluated using a cross-modal matching method. In this procedure, the loudness was matched to an equivalent muscular force. Subjects had to judge the instantaneous and global loudness using a proprioceptive input device with force feedback. Using this technique, subjects could respond easily, rapidly, continuously, and with feedback concerning their evaluation. The present study examines the relation between instantaneous and global loudness of 16 urban soundscapes lasting about 20 s using the same method. First, the procedure is individually calibrated with 1-s soundscapes presented at different sound-pressure levels. The data obtained for each subject fit well with a linear relation on a log–log scale between the sound-pressure level in dB and the associated force in log Newtons. Then, the global judgments are performed under two experimental conditions: one with instantaneous loudness evaluation during stimulus presentation and one without. Good correlation was found between these two judgments. The global judgment and the average over the instantaneous matching contour are nearly identical. In addition, a fluction factor (FI) calculated on the loudness judgment contour reflects the degree of movement produced by the subject as well as the presence of discriminable sources.

5:40

1pNSa11. Is pleasantness for soundscapes dimensional or categorical? Valerie Maffiolo, Michele Castellengo (Lab. d'Acoust. Musicale, Univ. Paris 6, case 161, 4 place Jussieu, F-75252 Paris Cedex 05, France, maffiolo@ccr.jussieu.fr), and Daniele Dubois (CNRS, UPR 9017, INALF, ENS, Montrouge, France)

This paper focuses on methodological issues regarding the dimensional or categorical properties of pleasantness for soundscapes. Sixteen sequences of urban soundscapes were recorded and reproduced with their real sound intensity, using techniques previously validated to produce the illusion of a real environment. The sequences were processed using two methods: free categorization and pair comparisons. The categorization method consisted of asking subjects to sort out the sequences according to pleasantness and to verbally qualify their categories. In the pair comparison task, subjects had first to choose, for each pair, which sequence was the most pleasant, and then to rate the pleasantness dissimilarity between the two sequences on a nine-point scale. Categories and ratings were processed through a cluster analysis and interpreted in connection with a psycholinguistic analysis of the verbal comments. Binary choices were transformed onto a dimensional representation of pleasantness. The main results lead to the discussion of the influence of procedures and data representations of the interpretation of pleasantness for soundscapes, and suggest that the dimensional aspect of pleasantness is more an artifact of representation than an intrinsic property of soundscapes. Unpleasant soundscapes seem to be more adequately represented as a specific category of sounds rather than values ranging on a scale.

Contributed Poster

This poster will be on display in the Poster Gallery from Monday to Wednesday, 15–17 March. The author will be at the poster from 2:00 p.m. to 4:00 p.m. on Tuesday, 16 March.

1pNSa12. Assessment of low-frequency noise in dwellings. New Polish recommendations. Marianna Mirowska (Bldg. Res. Inst., Acoust. Dept., Ksawerow 21 02-656 Warsaw, Poland)

The paper presents the new Polish recommendations for evaluation of low-frequency noise penetrating into dwellings from appliances installed inside or outside of building. Based on measurement data of annoyance noise, epidemiological investigations of the influence of noise on the health of exposed inhabitants, laboratory tests of thresholds of narrow- and broadband noise perception, review of the present literature, the new assessment criteria was proposed. In order to assess the noise spectra measured in dwellings, the A10 characteristics as the rating curve have been accepted. Its L levels for one-third-octave bands are determined by relation: $L_{A10} = 10$ kA. The low-frequency noise occurs as annoying when the sound pressure levels of noise exceeds the levels of A10 curve and exceeds the background noise by more than: 10 dB for tone noise and 6 dB for broadband noise.

Session 1pNSb

Noise: Noise from Air-Conditioning, Ventilating and Industrial Fans

Jean Tourret, Cochair

CETIM, Industrial Acoustics Department, B.P. 80067, F-60304, Senlis, France

Robert Schlinker, Cochair

United Technologies Research Center, 411 Silver Lane, East Hartford, Connecticut 06108, USA

Invited Papers

2:00

1pNSb1. Fan noise—In-duct sound power—What are we measuring and why? W. T. W. Cory (Woods of Colchester Ltd., Tufnell Way, CO4 AR Colchester, Essex, UK, Bill.Cory@woods-fans.com)

The paper will review the history of the measurement of fan noise. Early standards will be described and related to the state of knowledge existing at the time. The need for reliable methods of measuring the noise in-duct was foreseen at an early stage. This required the development of microphone shields able to ensure accurate readings of noise in a moving airstream. The relationship of in-duct levels to those measured around the fan in reverberent or free-field conditions will be discussed. Finally, the forthcoming ISO standards will be introduced and the recognized need for a number of different levels to be recorded.

2:20

1pNSb2. In-duct fan sound power measurement. Part I. Review of general problem. Wolfgang Neise and Frank Arnold (DLR-Institut fuer Antriebstechnik, Abteilung Turbulenzforschung Berlin, Mueller-Breslau-Strasse 8, 10623 Berlin, Germany, Wolfgang.Neise@dlr.de)

Many technical fan installations involve a duct on the inlet side and/or outlet side of a fan. Therefore the need arises for a measurement method for determination of the sound power radiated by fans or other fluid handling sources into a duct. In the paper the problems associated with such a method are discussed: Axial standing waves due to sound reflection from the duct end, acoustic loading of the source, turbulent flow pressures superimposed on the sound field, flow noise generated by flow/microphone probe interaction (self noise), discrimination between sound pressures and turbulent flow pressures, measurement position in the duct in view of higher-order mode sound propagation and directional characteristic of the microphone probe used, and modal distribution of sound power. Early and recent work on the above topics is reviewed. A brief description of the standarized in-duct method ISO 5136 (1990) is given.

2:40

1pNSb3. In-duct fan sound power measurement. Part II. Revision of ISO 5136. Wolfgang Neise and Frank Arnold (DLR-Institut fuer Antriebstechnik, Abteilung Turbulenzforschung Berlin, Mueller-Breslau-Strasse 8, 10623 Berlin, Germany, Wolfgang.Neise@dlr.de)

In recent investigations problems with the present version of the in-duct method ISO 5136 (1990) became apparent which indicate that a revision of the standard is necessary. The two major problem areas are: (1) Swirl flow existing in the test duct may interfere with the microphone probe to give too high sound power levels. Farzami and Guedel [1992 Proc. Fan Noise, CETIM Senlis, France, pp. 375–380] showed that this problem can be solved by placing a flow straightener between the fan outlet and measurement plane. (2) By comparison with other methods of sound power determination [Holste and Neise, JSV **152**, 1–26 (1992)], the sound power levels obtained by using the in-duct method are too low by several decibels in the frequency region of higher-order mode sound propagation. It is shown that the discrepancy between the results of the in-duct method and other methods is caused by the so-called modal correction C4 which is to account for the directivity characteristic of the microphone fitted with a turbulence screen in view of the propagation angle of the higher-order duct modes. Experimental and theoretical results are presented for a new combined frequency correction C34 which includes the effects of the superimposed mean flow on the sensitivity and directivity of the microphone probe as well as on the modal sound propagation and characteristics in the duct.

3:00

1pNSb4. Numerical method and software package for prediction of pressure pulsation in centrifugal ventilators. S. Timouchev (InteRe Ltd., DG, 19 Apt. 11, Engels St. 141400 Khimki, Moscow Reg., Russia, irico@glas.apc.org)

A numerical method and software package was developed for designers and researchers in the field of vibration and noise problems in centrifugal ventilators. The numerical procedure is based on a representation of unsteady compressible liquid flow as a form of vortex (pseudo-sound) and acoustic mode superposition. It gives a possibility to determine absolute amplitudes of pressure pulsation in the ventilator casing induced by unsteady interaction between nonuniform flow outgoing from a centrifugal impeller volute tongue or diffuser vanes, so-called blade passing tones. Inupt data includes 2-D ventilator geometry and operation mode. Acoustic impedance boundary conditions can be defined with a circuit specification or by direct inlet and outlet specific complex

impedance definition. It is possible to take into account local wall-specific complex impedance to study a coating effect on reduction of pressure pulsation. Output data represents: unsteady pressure map in the ventilator casing; amplitude distribution map for any selected harmonic; pressure versus time curves with accompanied spectrum information at any point selected by user; total vibration load vectors acting on the ventilator volute casing; and static pressure, velocity, and vorticity distribution in the centrifugal impeller. Validation of the software was completed on the centrifugal pump air model with more than 400 measuring points for two different radial gaps. The mismatch with experimental data is mostly 1.5–2.0 dB of unsteady pressure amplitude. Effects of radial clearance, number and form of impeller blades, exit and wall impedance are considered as well. The successful validation of the numerical method shows a good prospect in optimizing centrifugal ventilator geometry with improving vibration and noise characteristics at the draft stage and reducing refinement costs.

3:20

1pNSb5. Modeling centrifugal fan blade trailing edge noise. Robert H. Schlinker, Bruce L. Morin, C. D. Coffen, and Roger L. Davis (United Technologies Res. Ctr., 411 Silver Ln., East Hartford, CT 06108, schlinrh@utrc.utc.com)

Centrifugal fan noise contains numerous aeroacoustic mechanisms, one of them being trailing edge noise due to turbulent boundary layer flows convecting over the blade trailing edge. A fundamental research program was conducted to develop a trailing edge noise database and assess the ability of existing aeroacoustic models to predict this mechanism. The database included unsteady surface pressure spectra, trailing edge wake characteristics, and acoustic sound power. To focus on trailing edge noise, a mixed flow radial fan was simulated with a circumferential array of nonrotating vanes located in a radial diffuser. Both loaded and unloaded vane geometries were tested to simulate varying pressure gradients in an actual fan blade passage. It was demonstrated that the trailing edge noise mechanism can be accurately predicted if the trailing edge wall pressure spectrum is known *a priori*. However, for radial fans the strong adverse pressure gradients appear to rule out the existence of a ''universal'' wall spectrum shape typically observed in flat plate zero pressure gradient boundary layer flows. Hence, futher trailing edge noise modeling needs to focus on predicting surface pressure spectrum shape, in addition to, amplitude and frequency content.

3:40–4:00 Break

Contributed Papers

4:00

1pNSb6. Aeroacoustic response of a six-bladed axial fan to installation effects. Jean-Guillaume Lalanne, Hans Boden, and Mats Abom (Dept. of Vehicle Eng./MWL S-10044 Stockholm, Sweden, jgl@fkt.kth.se)

First, the effects of the duct on the aeroacoustic response of an axial fan have been studied experimentally and theoretically. The loading effect was modeled by a 1D model in the plane-wave frequency domain. This model was validated experimentally by changing duct lengths and fan positions. Second, an experimental characterization of the inflow turbulence has been done. Single degree-of-freedom statistical parameters have been measured as autocorrelation and temporal autospectrum for the axial turbulence components in order to estimate the integral scale and turbulence intensity. In parallel (the fan being located between an anechoic room and a reverberant room), sound pressure and sound-power measurements have been carried out. Different grids and supercritical screens have been designed to change the inflow conditions. The next task is to determine the spectral pattern in both the temporal frequency and tangential wave number domains and correlate it to acoustic response. Only the tonal part of the spectrum is considered. [Work supported by EEC.]

4:20

1pNSb7. Experimental determination of the fluctuating pressure on a rotating fan blade as a source of acoustic noise. Michael Stremel and Thomas Carolus (Institut fuer Fluid und Thermodynamik, Fachgebiet: Stroemungsmaschinen, Universitaet-Gesamthochschule Siegen, 57068 Siegen, Germany)

The fluctuating pressure on the rotating blades is a significant quantity responsible for the noise emitted from axial fans. In order to compute the noise from flow field data, a model is required which describes the fluctuations as a function of flow parameters. The paper presents the fluctuat-

ing pressure, measured on the rotating blade of a fan under different operating conditions. The turbulence intensity and the spectral density of the turbulent inflow velocity fluctuations are varied and measured. From CFD calculations quantities, such as boundary layer thickness, shape parameter, etc., of the flow around the blades are deduced and used as parameters. Simultaneously, the noise emitted from the fan is measured and compared to the fluctuating blade pressure.

4:40

1pNSb8. Aeroacoustics of axial propeller: Applying to an acoustic wind tunnel and optimization of an axial propeller for cooling system. Y. Dupont (Sciences Industries Conseils, 14, avenue de Sceaux, 78000 Versailles, France)

A simplified model is more convenient to predict design parameter effects on operating characteristics and radiated noise of a propeller. Thus, the aeroacoustic design software, DECIVENT, is used for subsonic axial flow fan, mounted as well in reduced systems like cooling systems as in greater structures like test wind tunnel. For a given flow rate and with the blades specified in cylindrical sections, the total pressure and torque are provided by the following computations: (i) outlet velocity triangle by solving radial balance, (ii) lift coefficient by a conformal transformation, and (iii) lift and drag forces, evaluated with considerations of laminar or turbulent boundary layer stall. The outlet relative velocity and force fields allow one to compute the radiated noise whose components are (i) loading noise, computed by Lighthill's model, and (ii) broadband noise computed by Fukano's model. Correlations with tests show the validity of this methodology. This software, weak customer in computational duration, may be

connected with an optimization software which, after some hundred iterations, allows one to exhibit the best compromise. Two propellers design by these means will be presented: one for a preliminary plan of an acoustic wind tunnel, and one for a cooling system.

5:00

1pNSb9. Experimental investigation of noise sources in cross-flow fans. Michael Herrmann,[a] Wolfgang Neise (DLR-Institut für Antriebstechnik, Abteilung Turbulenzforschung Berlin, Mueller-Breslau-Strasse-8, 10623 Berlin, Germany, Wolfgang.Neise@dlr.de), and Robert H. Schlinker (United Technologies Research Ctr., East Hartford, CT 06108)

Cross-flow fans are unusual types of blowers which operate in a fundamentally different way from axial or centrifugal fans. The flow enters and leaves the impeller in a direction vertical to its axis and passes the blade row twice, first radially inward and then radially outward. Hence, cross-flow fans are basically two-stage fans. Aerodynamic and acoustic modeling of cross-flow fans is extremely difficult because of the complexity of the flow field in the rotating impeller channels with a reversal of the flow direction during each revolution and large regions of severe flow separations. To obtain experimental input data for modeling efforts, experiments were performed with a scaled up model fan. The fan was installed in an open-inlet/ducted outlet test rig. Measurements were made of the unsteady pressures on the impeller blades and on the fan casing, in particular the vortex wall. The flow fields inside the impeller and at its outer periphery were studied by using hot wire anemometry and a three-hole probe. The effect of different casing geometries was examined. A simple model for broadband noise radiation involving surface dipole sources as suggested in an early paper by Sharland [JSV **1**, 302–322 (1964)] predicts the spectrum measured in an anechoically terminated outlet duct reasonably well for a medium frequency range. [a] Now with Behr GmbH & Co., Mauserstrasse 3, 70469 Stuttgart, Germany

5:20

1pNSb10. Practical approach to ANC of fan noise with feedback compensation. Petr Koníček and Ondřej Jiříček (CTU in Prague, FEE, Dept. of Phys., Technicka 2, 166 27 Prague, Czech Republic, jiricek@fel.cvut.cz)

Attenuation of low-frequency noise is a difficult task using passive methods. This paper presents the results of a fan noise reduction in a duct with circular cross section and a small diameter using active noise control. The control algorithm is designed for the optimum least square control in the time domain and contains acoustic feedback compensation. This algorithm was used for the attenuation of noise generated by a small axial fan. The discrete components of noise were totally canceled and the broadband component was reduced. The results of our project will be used for betterment of the working environment.

5:40

1pNSb11. Noise from airborne air conditioning system. Oleksander I. Zaporozhets and Vadim I. Tokarev (Acoust. Lab., Kyiv Intl. Univ. of Civil Aviation, 1, Ave. Cosmonaut Komarov, Kyiv, 252058 Ukraine, zap@elan-ua.net)

A prediction model for noise produced by an air conditioning system in an aircraft pressurized compartment is described. It is based on analysis of the sound flow energy in air conditioning system channels, including both the generation attenuation effects of the acoustic oscillations within the system. Differences in calculations from those usually used in stationary air conditioning systems for industrial, public, and residential buildings are discussed. The mechanism of noise generation by compressors, cooling turbine, separated flows in regulator, and jet flows from the pipe nozzle are investigated. The calculation results are obtained and carried out for different operational airplane conditions. Optimum tasks for operation conditions of an air conditioning system for noise impact criterion are formulated and solved.

Contributed Posters

These posters will be on display in the Poster Gallery from Monday to Wednesday, 15–17 March. Authors will be at their posters from 10:00 a.m. to 12:00 noon on Tuesday, 16 March.

1pNSb12. Rotational noise generation in peripheral fans. Volkmar Weise (TU Bergakademie Freiberg Institut f. Fluidmechanik und Fluidenergiemaschinen Lampadiusstr. 2, D-09596 Freiberg, Germany)

Peripheral fans are used in technical applications where radial and axial fans cannot supply the required pressure ratio at a low mass flux rate. Characteristic for peripheral fans is their high noise emission in the range above 1000 Hz. In comparison to the theory of noise generation in axial fans the sound sources in peripheral fans are not only the rotational blade forces. From an aeroacoustic viewpoint the main difference between axial fans and peripheral fans is the stripper which is necessary to separate inlet and outlet. The sound generation at the outlet is related to the circulating flow between rotor and side channel. This circulation is the main reason for the momentum and energy transfer from the rotor to the gas. The stripper divides the flow in the side channel from the flow between the blades. The result is the modulation of the circulating flow with monopole sources of aerodynamic noise. At the inlet the relief of the blade chambers filled with higher pressure proved to be the main aeroacoustic source by time-dependent periodical mass flux. According to the theory of aerodynamic sound generation this expansion flow represents a monopole source of noise. [Work supported by DFG.]

1pNSb13. Experiences with attenuation of low-frequency noise in duct. Ondřej Jiříček and Petr Koníček (CTU in Prague, FEE, Dept. of Phys., 166 27 Prague, Czech Republic, jiricek@fel.cvut.cz)

Fan noise still represents a serious component of noise pollution in the working place. Fan noise contains both random and discrete frequency components. This paper presents results of experiments with the ANC system useful for low noise design of air-conditioning systems. Computer simulation and measurements were used to examine the performance of the tested feedforward system. The control algorithm is designed for the optimum least square control in the time domain. Experimental results of an investigation of active control of sound are presented.

Session 1pNSc

Noise: Railway Noise I

Paul J. Remington, Cochair
BBN Systems and Technologies, 10 Moulton Street, Cambridge, Massachusetts 02138, USA

Markus Hecht, Cochair
Institut für Strassen- und Schienenverkehr, Fachgebiet Schienenfahrzeuge, Sekr. SG-14, Salzufer 17-19, D-10587 Berlin, Germany

Chair's Introduction—2:15

Invited Papers

2:20

1pNSc1. The effects of rail support stiffness on railway rolling noise.　David J. Thompson, Chris J. C. Jones, and Guillaume de France　(Inst. of Sound and Vib. Res., Univ. of Southampton, Highfield, Southampton SO17 1BJ, UK, djt@isvr.soton.ac.uk)

Rolling noise from trains is radiated by both wheel and track vibrations, the track being dominant in many cases. The stiffness of the rail fastener system, especially the elastomeric rail pad usually inserted between the rails and the sleepers, has a significant influence on the noise emitted by the track. Railways are increasingly using softer pads to reduce potential damage to sleepers and ballast. Softer pads decouple the rail from the sleeper. This reduces the noise from the sleeper but also reduces the decay of vibration with distance along the rail and hence leads to an increase of the noise radiated by the rail. The paper describes experimental and theoretical work to investigate the influence of pad stiffness on the dynamic and acoustic behavior of track. The vibrational behavior has been measured on a dedicated 36-m section of railway track, with different types of rail pad installed. The results are compared with predictions using theoretical models of the track behavior. This allows the effective pad stiffness and damping to be determined, laboratory measurements also being available for comparison. The attenuation of vertical and lateral vibration along the track is measured in order to estimate the effect on radiated noise.

2:40

1pNSc2. Time domain simulation of railway noise.　Anders Nordborg　(ABB Corporate Research, Appl. Mech., SE-72178 Vasteras, Sweden)

The most difficult problem concerned with railway noise generation is the process occurring in the contact region during rolling. The following computation efficient time domain model determines the vertical rail deflection by use of Green's function of a periodically supported beam. Nonlinear Hertzian contact mechanics are used, less realistic, however, than a discretized contact region allowing for rough surfaces. The wheel is as yet a rigid mass. More advanced contact and wheel models can readily be included later. Calculation results include the following observations. The contact force peaks sharply precisely above the support points, followed by a short loss of rail–wheel contact, also for smooth rails. Vibrations of the rail pinned–pinned mode are impossible to eliminate by removing surface roughnesses, because the excitation is above all caused by forward wheel velocity together with discrete supports. The response amplitude increases with forward velocity. Optimum pad stiffness depends on corrugation amplitude. These new findings should be considered when trying to noise optimize track constructions.

3:00

1pNSc3. Short pitch corrugation on straight tracks—Theory and numerical simulation.　Steffen Müller and Klaus Knothe (ABB Res. Ctr., Speyerer Str. 4, 69115 Heidelberg, Germany and Tech. Univ. of Berlin, Marchstr. 12, 10587 Berlin, Germany, steffen.mueller@decrc.abb.de)

Simulation models (Frederick, Sinclair, Valdivia, Hempelmann) which have been developed during the last 20 years showed the formation of short pitch corrugation to be a frequency constant mechanism, where a track mode about 1000 Hz dominates the corrugation process. Based on this, however, it is difficult to explain the relatively small variation in wavelength with vehicle speed which is observed in practice. In this presentation a sophisticated linear wheel-rail model is introduced which could help in resolving this contradiction. The transient dynamics is described by a feedback loop: Fluctuations in the contact geometry and the creepages cause varying contact forces. This produces fluctuating relative displacements between wheel and rail which in turn change the contact geometry and the creepages again. The simulation of the long-term behavior shows how the rail surface is damaged after a high number of wheel passages. Based on numerical results it is demonstrated that, due to contact mechanical effects, short pitch corrugation is growing within a fixed wavelength band, which could explain why the wavelength varies only a little with vehicle speed. The presented model is therefore a good basis for future research on how short pitch corrugation can be suppressed.

1pNSc4. On the determination of dynamic properties of elastic rail fastenings. Rolf J. Diehl and Paul Hofmann (Müller–BBM, Robert-Koch-Strasse 11, D-82152 Planegg, Germany)

The dynamic properties of the resilient layers of rail-fastening systems play an important role for sound and vibration emission from railway lines. In order to achieve low sound radiation from the rails, a rather stiff fastening for the rail is advisable, whereas for vibration minimization rather soft fixtures would be preferred. The dynamic properties of fastening systems are determined from measurements. In order to have comparable results, a CEN standard is currently being written which is consistent with the method described in ISO 10846 for laboratory measurements of the dynamic stiffness. The method and examples for results will be discussed. For the optimization of the setup for a given situation [track, vehicles, speed(s), operating conditions, surroundings], theoretical models are useful tools. In addition, such models can be used to predict the future impact. Apart from the prediction model itself, the quality of a prediction depends on the reliability of the input data. Results of calculations with the rail/wheel impedance model RIM will be shown to demonstrate the effect of different vertical stiffnesses of the rail fastening and its frequency dependence on the emitted sound and vibration levels.

1pNSc5. Reduction of ground-borne noise from rail systems. George P. Wilson (Wilson, Ihrig & Assoc., Inc., 5776 Broadway, Oakland, CA 94618)

Over the last 35–40 years there has been an ever increasing awareness of ground-borne noise from rail system train operations and an increasing need for mitigation. In response there have been extensive and continuing efforts to develop rail fixation and support systems which will reduce or control the ground-borne noise which originates at the wheel/rail interface. The rail fixation systems have varied from simple resilient boots on sleepers or resilient pads under rail fixation plates to elaborate concrete floating slab trackforms. Intermediate mitigation designs have included ballast mats, booted duo-block sleepers, and many configurations of resilient rail fixation baseplate assemblies. A review of the various types of ground-borne noise mitigation trackform designs is presented along with indication of the effectiveness for each type or class of design. Further, characteristics or design parameters of the various classes of design which result in poor performance are discussed, particularly for floating slab trackform.

4:00–4:20   Break

Contributed Papers

1pNSc6. STV: Silencing goods traffic. Part 2: Demonstrating practical low noise solutions. J. Lub (NS Technisch Onderzoek, P.O. Box 8125, NL-3503, RC Utrecht, The Netherlands)

The Dutch national project, Quiet Rail Traffic (Stiller Treinverkeer STV), aims at an overall A-weighted noise reduction of at least 10 dB for day-to-day goods traffic, with a maximum achievable reduction of 16 dB. The project has as goals the development of validated design and analysis methods, and the demonstration thereof by means of the construction and testing of two prototype quiet goods wagons and a prototype quiet track construction. Validation tests will be performed in November 1998. Static and pass-by noise and vibration tests are conducted with two prototype container carrying wagons, loaded with containers and a reference vehicle. The goods wagons are equipped with laser-cladded wheels, composite brake blocks, drum brakes, magnetic brakes, and wheel skirts. They are operated on a continuously laid slab track construction with embedded rail. The rail type is of a newly designed type, optimized for noise reduction and application in track designs with continuous rail support. Mini-barriers and sound absorption material are applied to the track. A telemetry system is used for measuring wheel noise. The improvements achieved by the total prototype wagon/track combination and components are compared to the predicted noise reduction and data collected in the reference phase of the project.

1pNSc7. The real sound emission of ''Rasengleis.'' P. Fuerst (cdf Schallschutz Dresden, Wilhelm-Liebknecht-Straße, 6, D-01257 Dresden, Germany, cdf-Schallschutz-DD@t-online.de)

The ''Rasengleis'' (railway lines which are covered by meadow and bedded on conventional ballasted tracks) has indeed a lower noise emission level. Various investigations focusing on the acoustical behavior of the ''Rasengleis'' were carried out in the last years. The results were rather different. The first sequence of measurements took place **before**

changing the construction of the second part of the railway line. That was to counterbalance errors due to locality, time (the last measurement was carried out almost 2 years later than the first one), and other ambient conditions. The levels could be easily compared among each other using the results of the reference point. Additionally, the measurements were repeated after at least a year to take aging processes of the meadow into account. Two different types of trams were used. They differ among other things in the kind of absorption used for the wheels. One wheel type contains rubber parts stressed of shearing (Tatra T4D), the other wheel rubber parts are stressed of pressure (Bochum). ''Rasengleis'' reduces noise emission depending on the kind of wheel up to 3 dB (minimum 2 dB). There will be measured levels of drive past, standardized values, and sonograms.

1pNSc8. Structure-borne noise reduction in a railway tunnel in Cologne by means of a ballastless mass–spring-system-type Züblin with discrete Sylodyn® bearings and dynamically soft Sylodyn® ballast mats. Rüdiger G. Wettschureck (Getzner Werkstoffe GmbH, P.O. Box 162, D-82025 Grünwald/Munich, Germany, Wet@getzner-werkstoffe.de), Franz Breuer (Peutz Consult GmbH, D-40599 Düsseldorf, Germany), Markus Tecklenburg (Getzner Werkstoffe GmbH, Bürs/Bludenz, Germany), and Horst Widmann (Ed. Züblin AG, D-70567 Stuttgart, Germany)

In the vicinity of Chorweiler (a suburb of Cologne) there is a tunnel in the double track of German Railways between Cologne and Düsseldorf which was built for airborne noise protection reasons. On both sides of this tunnel there are very close housing areas built during the last years. The structure-borne noise of the rail traffic caused a considerable interference with the residents. In 1997 measures to reduce the railway vibrations were installed in both tracks of a 970-m-long tunnel section. In one part of the tunnel, 230 m long, a mass–spring system with a natural frequency of 11 Hz, using discrete bearings, was installed. The rest of the tunnel was equipped with ballast mats tuned to extremely low dynamic stiffening and

to main line axle loads (push–pull trains of commuter traffic Rhein/Ruhr) with a specific static stiffness of 0.03 N/mm^3 [according to DB-TL 918071, Edition June (1988)]. The efficiency of these reduction measures, that means their insertion loss, was estimated by means of lumped-parameter models, and measurements of the structure-borne noise during train passages were carried out on identical measuring points before and after the structural alterations in the tunnel and in adjacent buildings. The measurement results show a good correspondence with the calculations and completely confirm the high expectations set in the measures.

5:20

1pNSc9. Short pitch corrugation on straight tracks—Practical experiences. Thomas Klimpel and Klaus Knothe (Tech. Univ. of Berlin, Inst. of Aviation, Dept. of Design, Marchstr. 12, 10587 Berlin, Germany)

One possibility to reduce rolling noise is to reduce roughness excitation. The basis of the paper is measurements of profile irregularities on a test section of the track between Baarn and Amersfoort in the networks of Nederlandse Spoorwegen (NS). In one of two segments, periodic wear pattern (short pitch corrugation) appears with wavelength from 3.5 to 4 cm, whereas the second one is uncorrugated. The line is used only in one direction and mainly by passenger cars with medium speed. Starting from the measured track data, systematic calculations with the program of Müller, which is presented in a separate paper, have been undertaken. The aim is to estimate the roughness growth and to find hints about how the growth can be stopped or at least reduced. In Müller's paper it is explained in detail that roughness growth is restricted to a wavelength range between 2 and 10 cm. Numerical results will be presented in order to clarify how

different track components influence the RGG process. The model only takes into account wear as a long-term mechanism. It shall be discussed how the long-term behavior is influenced by plastic deformation, hardness variations, or different values of micro-roughness.

5:40

1pNSc10. A reciprocal method to evaluate low, close to track noise barriers. P. F. van Tol (NS Technisch Onderzoek, Postbus 8125, 3503 RC, Utrecht, The Netherlands) and H. A. Holties (M+P Raadgevende Ingenieurs, 1430 AH, Aalsmeer, The Netherlands)

In the framework of the Dutch program Quiet Train Traffic (Stiller Trein Verkeer) low, close to track barriers and absorptive layers on a slab track have been investigated. Following numerical calculations on low, close to track barriers and absorptive layers an experimental study has been made in order to evaluate the different solutions. A special test site has been constructed, which acoustically resembles a slab track. It enables a quick mounting and dismounting of different low, close to track barriers and absorptive layers on the slab. Using a reciprocal measurement method the acoustical performance (insertion loss) of different solutions has been determined. One of the advantages of a reciprocal method is that the noise reduction for different noise sources on a freight train, such as wheels, rails, and wagon superstructure, due to absorptive layers and low, close to track barriers is easily identified. Note that this stationary method is very efficient since no train passby is necessary. It is explained that a real train passby may be simulated by combining the measurements over multiple angles. Finally, a comparison will be made between the experimental and the numerical results.

MONDAY AFTERNOON, 15 MARCH 1999   ROOM H2032, 1:55 TO 4:00 P.M.

Session 1pNSd

Noise: Airframe Noise

Ulf Michel, Cochair

DLR, Institut für Antriebstechnik Abt. Turbulenzforschung, Müller-Breslau-Strasse 8, 10623 Berlin, Germany

Michele G. Macaraeg, Cochair

Aerodynamic and Acoustic Methods Branch, NASA Langley Research Center, MS 128, Hampton, Virginia 23186, USA

Chair's Introduction—1:55

Invited Papers

2:00

1pNSd1. The ''Owl'' as a challenge in airframe noise research. Geoffrey M. Lilley (ICASE, NASA Langley Res. Ctr., Hampton, VA 23681 and Dept. of Aeronautics and Astronautics, Univ. of Southampton, Southampton SO17 1BJ, UK)

The ''silent'' flight of the ''owl'' has captured the attention of the aeronautical community for over 50 years. The silent flight, resulting from changes in the flight feathers of the ''owl'' from all other birds, notably the leading edge comb, the trailing edge fringe, and the upper wing covering of the velvety down feathers, has been known and carefully documented by biologists, ornithologists, and aeronautical engineers. Many of the noise characteristics of an ''owl'' have similarities with airframe noise. By training owls to fly with the leading edge comb removed Kroeger *et al.* (1971) established that the leading edge comb was essential for stable flight at its high lift and low speed. However Kroeger's measurements showed that the ''owl'' had a broadband radiated noise spectrum indicating that the flow in the adverse pressure gradient on the rear upper surface was turbulent at the wing trailing edge. The present paper describes the effect of the trailing edge fringe and the down feathers on the noise spectrum generated by the owl and why the owl remains silent to its prey in spite of their excellent hearing above 2 kHz.

2:30

1pNSd2. Airbus airframe noise reduction—A European harmonized research effort. Werner Dobrzynski (DLR, Institut für Entwurfsaerodynamik, Abt. Technische Akustik, Lilienthalplatz 7, 38108 Braunschweig, Germany)

Very large aircraft—such as the planned Airbus A3XX—require correspondingly high lift forces and extended landing gear structures, thus causing potentially very high airframe noise levels during the landing approach. This problem was realized by the Airbus Consortium, and the first relevant research initiatives date back to the early 1990s. Soon after, initial research at DLR began under contract to Airbus Industrie, dedicated to landing gear airframe noise. Recognizing the general challenge of research for aircraft noise reduction at that time, a European Noise Reduction Initiative (ENRI) was proposed, dealing with both engine and airframe noise. Subsequent technical discussions, however, led to separate programs for engine and airframe noise research, the latter being initiated under the acronym RAIN (reduction of airframe and installation noise). RAIN encompasses analytical and experimental work to describe, predict, and reduce the noise both from landing gears and high lift devices. In parallel, national research programs began: In France wind tunnel and flyover experiments relate to the whole aircraft airframe noise characteristics, while corresponding research in Germany focuses on high lift devices noise. In Great Britain, noise modeling is performed for landing gear noise prediction. These national efforts are coordinated by the Airbus partner companies (3E-program).

3:00

1pNSd3. Directivity of flap side-edge noise as observed with a phased microphone array on landing aircraft. Ulf Michel and Weiyang Qiao[a] (DLR, Müller-Breslau-Str. 8, 10623 Berlin, Germany)

The sound emissions of a large number of aircraft landing at the airport of Frankfurt/Main, Germany, were studied with an 8- by 8-m phased microphone array consisting of 111 irregularly distributed microphones. The data reduction procedure takes account of the motion of the sound sources. The measurement procedure is described and results for the directivity of flap side-edge noise of various aircraft types are reported. Flap side-edge noise is an important airframe noise source on many aircraft and can be clearly identified with the phased array. The variation of this noise source between different aircraft types is considerable, which indicates a substantial noise reduction potential. [a] Permanent address: NW Polytechnical Univ., Xian, PROC.

3:30

1pNSd4. Simulation of acoustic scattering from a trailing edge. Bart A. Singer, Kenneth S. Brentner, David P. Lockard (NASA Langley Res. Ctr., Hampton, VA 23681-2199, b.a.singer@larc.nasa.gov), and Geoffrey M. Lilley (ICASE, Hampton, VA 23681-2199)

Unsteady two-dimensional flow calculations were performed about a thin NACA airfoil with a bluff-body vortex generator positioned at 98% chord. The bluff body produced unsteady vortex shedding, which simulated large coherent eddies in a boundary layer. The shed vortices interacted and occasionally paired as they convected past the sharp trailing edge of the airfoil. The CFD calculations clearly showed acoustic waves emanating from the airfoil trailing edge. The Ffowcs Williams and Hawkings equation [J. E. Ffowcs Williams and D. L. Hawkings, Philos. Trans. R. Soc. A **264**, 321–342 (1969)] was used to compute the acoustic field generated by the unsteady aerodynamic field. Directivity maps and Mach-number scalings have been obtained and compared with the theoretical predictions for trailing-edge noise. The noise below the airfoil displays characteristics typical of a sound field scattered from a sharp edge. The noise above the airfoil is more complicated; it contains significant contributions from both the scattered acoustic field and the bluff-body vortex generator. The effects of spanwise correlation length and the choice of integration surface will also be discussed.

Session 1pNSe

Noise: Jet Acoustics

Michael Fisher, Cochair

Institute of Sound and Vibration Research, The University, Southampton SO17 1BJ, UK

Philip J. Morris, Cochair

Department of Aerospace Engineering, Pennsylvania State University, 233P Hammond Building, University Park, Pennsylvania 16802, USA

Invited Papers

4:30

1pNSe1. The noise spectrum of isotropic turbulence. Geoffrey M. Lilley (ICASE, NASA Langley Res. Ctr., Hampton, VA 23681 and Dept. of Aeronautics and Astronautics, Univ. of Southampton SO17 1BJ, UK)

Proudman's solution for the noise radiated from isotropic turbulence at low Mach numbers and high Reynolds numbers was reviewed by Lilley (1994) following the simulation of Sarkar and Hussaini (1993). It was suggested (Lilley, 1996) that the Lighthill–Proudman theory could be used for the prediction of the radiated acoustic power from turbulent free shear flows in the absence of shock waves. In this approximation the Lighthill ''filter function'' was used with distributions of the turbulent kinetic energy, the temperature fluctuations, and the turbulent length scale. The results were compared with experimental measurements. But the method (Lilley, 1996) was only concerned with the total acoustic power. Modifications to this theory are required for the estimation of the spectrum and directivity of the radiated sound. The frequency resolution in time-dependent RANS calculation (TRANS) is limited by the grid used. In most, the large scale structure of the turbulence is resolved and consequently the energy containing scales captured. But the high-frequency part of the spectrum is poorly defined. Present work discusses two models for the high-frequency spectral decay and some of the properties of the high-frequency spectrum in turbulent shear layers.

5:00

1pNSe2. Jet noise of subsonic aircraft: An aeroengine manufacturer's perspective. A. J. Kempton (Installations Eng., Rolls-Royce plc, Derby, UK)

These papers trace the history of jet noise reduction from early pure jets to modern engines with high bypass ratios. Choice of engine cycle is important in reducing jet noise, but there is also the need to balance the requirements of cost, weight, and specific fuel consumption, in addition to minimizing other engine noise sources. Some techniques for reducing jet noise are illustrated. Forced mixers on long-cowl engines are effective in reducing low-frequency jet noise, but can cause an increase in high frequencies (especially in flight). For larger aircraft with more powerful engines, where the noise problem is most severe, installation issues dictate somewhat against long-cowl configurations. The current challenge is therefore to achieve similar noise reductions to forced mixers but on short-cowl engines. In the future, the requirement for further noise reductions might result in much more novel aeroplane and engine configurations. Jet noise experiments can easily give misleading results unless the facilities and rigs are designed to avoid contamination by excess noise and to provide correct simulation of boundary layers and other aerodynamic parameters (including flight simulation). Fortunately, testing at reduced scale is acceptable for noise. Advanced measurement techniques (like source location) can be very beneficial in aiding understanding.

5:30

1pNSe3. Industry perspective on jet noise—Prediction and reduction requirements. David H. Reed, T. R. S. Bhat, and W. T. Conley (Boeing Commercial Airplane Group, Seattle, WA)

Spectacular growth of airline passenger traffic, nearly tenfold since 1970, has been fueled by a steady trend toward lower cost airfares made possible by technological achievements in airframe and aircraft engine design. Of all the advancements in aircraft design, none has been more important than that of the high bypass ratio (HBPR) turbofan engine, first introduced in 1969 on the Boeing 747. Since then, HBPR turbomachinery noise technology has kept pace with other engine refinements, allowing newer aircraft to meet increasingly stringent community noise requirements. Even more advanced turbomachinery and nacelle noise technology is now at hand as a result of government/industry efforts in both Europe and the United States. Engines with substantially lower fan noise will be introduced. Jet noise will remain the major obstacle to further noise reduction. Since the majority of new commercial aircraft over the next 20 years will be powered by HBPR engines, research in jet noise must be a major priority. This paper examines

the needs for jet noise reduction across the range of commercial aircraft types, and suggests technology development efforts including CFD tools for jet characterization, active flow control, installation effects, and novel suppression devices. This paper also discusses challenges in jet noise technology for supersonic airliners.

Contributed Paper

6:00

1pNSe4. Intensive narrow-band noise of oscillating circular gas jet. Andrew G. Semenov and V. Andrew Rimsky-Korsakov (N. N. Andreev Acoust. Inst., Russian Acad. of Sci., 4 Shvernik St., 117036 Moscow, Russia, bvp@acoins.msk.ru)

Self-sustained oscillations of a jet issuing from a circular nozzle in an impedance plane into unbounded space or interacting with various obstacles, such as a parallel plane or a coaxial rigid circle, are analyzed. In the problem statement region of the circular jet inner boundary, alternative

deflection during self-interaction or interaction with obstacles is considered as an axisymmetric monopole (volume) source of axial sound waves inside the jet cylinder directed to the jet root. By means of the corresponding 1-D boundary value problem solution, Nyquist stability criteria are derived for oscillations inside the cylindrical volume bounded by jet boundaries. Conditions favorable for jet boundary axially symmetric self-sustained oscillations excitation are found and their relations to the problem basic parameters are predicted, both for subsonic and supersonic jet cases. The proposed method could be used in various practical jet-obstacle interaction noise-generation problem analyses.

MONDAY AFTERNOON, 15 MARCH 1999   ROOM MA043, 2:20 TO 5:40 P.M.

Session 1pPAa

Physical Acoustics: Scattering

Richard Stern, Cochair

Applied Research Laboratory, Pennsylvania State University, P.O. Box 30, State College, Pennsylvania 16801

Adriano Alippi, Cochair

Department of Energetics, University of Rome, ''La Sapienza,'' via Scarpa 14, 00161 Rome, Italy

Contributed Papers

2:20

1pPAa1. Studies of scattering from differently shaped objects using the TLM technique. Ulf Kristiansen (Acoust., Dept. of Telecommunication, Norwegian Univ. of Sci. and Technol., O. S. Bragstads plass 2B N-7034 Norway) and Nicolas Brachet (ESM2, Technopole de Chateau-Gombert, 13451 Marseille, France)

The transfer line method (TLM) can be regarded as a discretized version of Huygen's principle. Wave energy propagates along a system of lines connecting regularly spaced node points. An energy packet reaching a node along a certain line will at the next time step be split up and reradiated into all the lines connected to that node. Wave propagation is therefore regarded as a discretized process in space and time with local updating rules. Special rules are applied for reflecting, absorbing, and partially absorbing surfaces. The method has been used for some time in electromagnetic wave propagation, but not so much in acoustic studies. The present study is a 2-D investigation of the scattering of sound pulses from objects having different shapes: circular, triangular, and rectangular. Performing this in the time domain allows the following of the scattering process in detail.

2:40

1pPAa2. Space-wave number representation of transient waves. Experimental determination of the space location origin of Lamb waves on cylindrical shells. Loic Martinez, Jean Duclos, and Alain Tinel (LAUE UPRESA CNRS 6068, BP 4006, Pl. R. Schuman, 76610 Le Havre, France, loic.martinez@lemel.fr)

A new space-frequency-wave number representation is introduced. This representation is obtained by space-sliding window FFT of the space-frequency representation. This new method is used to experimentally define the space origin of the pulse-generated Lamb wave on an air-filled cylindrical shell immersed in water (relative thickness is equal to 0.03).

The space-wave number slices also allow the measurement of the Lamb-wave complex wave numbers and their propagation direction (reduced frequencies range from 20 to 1000). The reflected field is experimentally separated from multiple propagating Lamb waves. This efficient method is complementary to the surface wave analysis method (S.W.A.M.) [Martinez *et al.*, 16th ICA and 135th ASA Congress, II, 1359–1360 (1998)]. In particular, the space-wave number representations are not limited by space windowing. The experimental results confirm the theoretical computations performed on waves propagating on a plane plate.

3:00

1pPAa3. Scattering by cylindrical shells: The Scholte−Stoneley wave nature. Naum Veksler, Jean-Louis Izbicki, and Jean-Marc Conoir (LAUE, CNRS UPRESA 6068, Pl. R. Schuman, 76610 Le Havre, France)

A systematic study of the A wave revolving around circular cylindrical elastic shells immersed in water is performed. The resonance frequencies and halfwidths have been computed for a wide range of the b/a ratio (b is the inner radius and a the outer one). The A-wave evolution is studied from the case of the quasisolid cylinder to the case of the very thin shell (b/a equal to 0.99). The resonances are computed from the complex frequency plane; it is then possible to get the dispersion curves of the A wave. The phase velocities dispersion curves, plotted against the adimensional variable ktd (kt is the shear wave number and d the thickness of the shell), practically coincide in the low-frequency range. The resonance halfwidths are small in this frequency range. The A wave is a bending type wave. At higher frequencies, the A wave becomes analogous to a diffracted Franz wave.

1pPAa4. Sound scattering by Gauss beam from cylindrical objects. I. L. Oboznenko, E. I. Oboznenko (Dept. of Electron., NTUU Kiev Polytech. Inst., Kiev 252056, Ukraine), and Vladimir I. Genis (Drexel Univ., Philadelphia, PA 19104)

Practically, when diagnosing vessels by focusing array, the vessel is radiated not by the plane wave, but a time-limited beam. Here one can see the results of theoretical and experimental research of Gauss beam sound scattering from cylindrical objects with wave resistance different from the medium. Well-known results of sound scattering on cylinders with limited sizes are wrong in reality, because of limiting the vertical size of the cylinder. In fact, all the objects, including the cylinder, are radiated by a nonconcentrated beam, like a Gauss beam. Theoretical and experimental results of Gauss beam scattering in the band of parameter $\alpha = a/w$ (where a-cylinder radius; w-Gauss beam width on e^{-1} range) from 10^{-6} (quasi-plane wave) to 100 (super-narrow beam) show, that a scattered wave spectrum changes very substantially. This point needs caution, and immediately changes classic results for the plane wave. Also, the sound beam echo-signal spectrum in the reverse direction gives one the ability to classify and to differ objects as ''more soft'' or ''more hard'' with respect to medium (for wave resistance it differs up to 5%–10% from medium). This permits the classification of cylindrical foreign inclusions or defects of little vessels like post-, precapillaries.

3:40–4:00   Break

4:00

1pPAa5. Scattering from cylindrical structures with various end-cap shapes. Angie Sarkissian and Louis R. Dragonette (Naval Res. Lab., Code 7132, Washington, DC 20375-5350, angie@aqua.nrl.navy.mil)

The scattered field from a cylindrical structure near the forward direction is computed using the Kirchhoff approximation of scattering from a screen. The effects of varying the shape of the end-caps are examined by carrying out target strength computations for various shapes. The cylindrical region is modeled to be impenetrable while the effects of varying the reflectivity of the end-cap regions are examined. [Work supported by ONR.]

4:20

1pPAa6. Complex poles of Lamb waves propagating along anisotropic and absorbing fluid-loaded plates. Marc Deschamps and Olivier Poncelet (Laboratoire Mecanique Physique, Univ. Bordeaux 1, 351 Cours de la Liberation, 33405 Talence Cedex, France)

This work presents theoretical and experimental investigations on Lamb wave generation along anisotropic immersed plates. The structure of these guided waves is strongly dependent on the nature of the source (transient source, bounded beam . . .). An incident plane wave for which the signal is time-limited is considered, and leads to transient Lamb waves mathematically defined by complex frequencies. These frequencies are the poles of the reflection/transmission coefficients of the fluid-loaded plates. This point of view, recently developed for isotropic plates, is completely different from the classic approach in terms of leaky Lamb waves, which are harmonic waves. New interesting results are then obtained for the NDT of thin anisotropic plates: the dispersion curves calculated for complex frequencies are very different from those calculated for real frequencies (harmonic waves). Moreover, some modes have a negative imaginary part of the frequency that implies transient phenomena increasing with time. The waveforms of the reflected signals are obtained and explained by the calculation of a contour integral in the complex frequency plane. The roles of anisotropy and absorbing effects are discussed.

4:40

1pPAa7. Experimental determination of the angular width of plate modes. Guy Durinck, Willy Thys (IRC, Katholieke Univ. Leuven, Campus Kortrijk, B-8500 Kortrijk, Belgium), Pascal Rembert, and Jean-Louis Izbicki (Univ. du Havre, 76610 Le Havre, France)

The reflection and transmission coefficient of a plate can be expressed as a function of frequency at fixed angle of incidence or as a function of angle of incidence at fixed frequency. In the first situation, the transmission coefficient can be considered as a superposition of frequency resonances, while in the latter case it can be considered as a superposition of angular resonances [R. Fiorito, W. Madigosky, and H. Uberall, J. Acoust. Soc. Am. **66**, 1857–1866 (1979)]. Using this approximation the properties of frequency resonances of a plate can be determined experimentally [Derible *et al.*, Ultrasonics International 93 Conf. Proc. 483–486 (1993)]. In this work, this approximation is used to determine the properties of the angular resonances. The transmission coefficient as a function of frequency is determined for a large set of angles of incidence by insonifying the plate with an ultrasonic pulse. Then, at fixed frequency, the data are plotted as a function of angle of incidence. The properties of the angular resonances are extracted using the Argand representation of the transmission coefficient. A comparison with theoretical values is made.

5:00

1pPAa8. Wedge diffraction analyzed by localized response function of the boundary value problem. Mitsuhiro Ueda (Tokyo Inst. of Technol., Dept. of IDE, O-okayama, Meguro-ku, Tokyo, 152-8552 Japan, ueda@ide.titech.ac.jp)

A new diffraction principle called virtual discontinuity principle of diffraction, abbreviated by VDPD, which is formulated by considering the wave propagation in a space seen by an observer virtually, has been proposed by this group. The boundary value problem is formulated using VDPD and the Green's theorem. Then it becomes clear that the response function, which shows mutual dependence of the potential on the surface of the object, is composed of two components. One of them is localized near the edges of the object that can be seen by the observer. The other is not localized, just like the response function used in the BEM formulation of the boundary value problem. The observer in the free space can always see the edge of a wedge. Thus the boundary value problem for the wedge diffraction can be formulated using the localized response function only. As a result of the localization near the edge, sampling points of the potential can be limited near the edge. Thus in spite of the infinite size of the wedge, the numerical procedure to solve the problem is simplified remarkably.

5:20

1pPAa9. Modal scattering from orifices. Jane L. Horner, Björn A. T. Petersson (Dept. of Aeronautical and Automotive Eng., Loughborough Univ., Loughborough LE11 3TU, UK, j.l.horner@lboro.ac.uk.), and Richard Lyons (Loughborough Univ., Loughborough LE11 3TU, UK)

When considering a sound wave traveling through a circular orifice in a rigid baffle, both the reflected and scattered field have to be established on the incident side of the orifice. Previous investigations have used Hankel transforms to establish the amplitudes of these two waves. This study concentrates on the coupling of the modes in the scattered and reflected fields, with the object of the investigation being to determine if a modal approach may be taken to describe the reflected sound fields from such an orifice. Results are shown in the form of the modal contributions to the integrand which must be used to obtain the scattered field.

Session 1pPAb

Physical Acoustics: Ultrasonic Non-Destructive Evaluation and Time Reversal Techniques

R. Glynn Holt, Cochair

Department of Aerospace and Mechanical Engineering, Boston University, Boston, Massachusetts 02215, USA

Walter Arnold, Cochair

Fraunhofer Institute for Nondestructive Testing, University, Building 37, D-66123, Saarbrücken, Germany

Contributed Papers

2:00

1pPAb1. Reflection coefficient of a Stonely–Scholte wave: An experimental investigation. Edouard Mouton and Manell E. Zakharia (CPE Lyon, LASSSO (LISA, EP92 CNRS), 43 Bd. du 11 Novembre 1918, BP 2077, Bat. 308, F 69616, Villeurbanne Cedex, France)

The Stoneley–Scholte waves are becoming more and more popular in both sediment characterization and nondestructive testing (computation of material and sediment properties, flaw or obstacle detection). When dealing with a 3-D obstacle, the coupling of these evanescent waves with the elastic waves in the target and the continuity conditions may have several definitions. In theoretical papers, some authors choose to impose the continuity of each component of the wave, some others the continuity of the sum vector. A specific tank experiment has been designed and conducted. Its specific goal is to define the continuity conditions that have to be used in future works. A special-purpose mock-up has been built up that contains two different blocks molded in an epoxy resin. One block is ''mechanically hard'' (duraluminum) and the other one is more ''soft'' (PVC). The geometry of both blocks has been especially optimized in order to enhance the reflection and refraction phenomena. For each material, two interfaces were considered: a perpendicular one and sloping one (wih respect to the direction of propagation of surface waves). Frequency range is centered around 100 kHz with an investigation thickness of a few centimeters.

2:20

1pPAb2. Characterization of bonding quality of composites with the nonlinear modulation method. Dimitri Donskoy, Alexander Ekimov, and Alexander Sutin (Stevens Inst. of Technol., 711 Hudson St., Hoboken, NJ 07030)

The nonlinear ultrasonic nondestructive testing technique is emerging as a valuable tool for material characterization and diagnostics. The most widely used nonlinear method is based on measurements of harmonic distortion of the probing ultrasonic wave. Another approach, developed by the authors, utilizes effect of modulation of ultrasound by low-frequency vibration. This method is more sensitive to various material and structural flaws, such as cracks, debonding, delamination, etc. It is also much easier to implement in practice. The present work focuses on characterization of bonding quality of composite plates used in the aerospace industry. The test was performed on 14 samples with various (known) degrees of bonding. The ultrasonic frequencies in the range of 50 kHz to 1.5 MHz were examined to determine the most sensitive frequencies. The modulating vibrations were in much lower frequency range up to 1 kHz. It was found that the average (across the frequency range) normalized modulation level is proportional to the bonding quality and can be used for its quantitative assessment.

2:40

1pPAb3. Sound scattering and transmission through cylindrical layer with concentrated defects. Vladimir I. Genis (School of Biomed., Eng., Sci. and Health Systems, Drexel Univ., Philadelphia, PA 19104) and Elena I. Oboznenko (NTUU Kiev Polytech. Inst., Kiev 252056, Ukraine)

An experimental method of defect detection in a cylindrical shell is proposed. Defects are the cylindrical hollows (the projections) on the outside or the inside of the shell, filled up, and encircled by water. The defect diameter makes up 1–2 mm (0–0,95) h long, where h—is a shell thickness. The inverse sound scattering field and pass wave field on the shell axis were measured in the sound acoustic beam. Experimental investigations were compared with theoretical investigations by changing the shell wave dimension in the 2–20 range (the 3-D resilience theory). The presence of the shell resonance clears surges and its inside liquid pillar as lengthening of the defect gradually results in their being illegible. The long defects reduce discrimination of the resonance system shell water. The through defect and the Helmholtz resonator are similar. The measurement results of the shells with defects and without were compared, when the shell was radiated by narrow sound beam (the beam width 10–15 times as much as the defect diameter). The theoretical calculation and experimental measurements of the nondefective shell are practically identical. The proposed method serves for defect presence (or absence) definition in pipelines, blood vessels, and others.

3:00

1pPAb4. Modeling of ultrasonic wave interaction with alumine and silica inclusions in steel. Noelle Mercier (LG2mS Acoustique, Univ. de Compiegne, BP 20529, Compiegne, France, Noelle.Mercier@utc.fr) and Nazir Chebbo (Univ. Libanaise, IUT de Saida, BP 813, Saida, Liban)

To progress in the characterization or discrimination of flaws in ultrasonic nondestructive testing, it is essential to have a good knowledge of the interaction of incident ultrasonic waves with these flaws and to model the echographic signals. The flaws considered here are of volumetric type. They have been simulated, in 2-D geometry, by circular cylinder forms. The developed model calculates the stemming time signal of the longitudinal or transverse wave diffused by a cylinder made of a fluid medium, an empty cavity, an elastic, or a rigid solid. The used method is based on the usual modal theory. The time signal is obtained by the Fourier transform of the multiplied monochromatic components, for each frequency, by the spectral response of the transducer. The presented results concern the case of two elastic media which simulate inclusions in steel. The convergence of the modal series, for a quite small flaw, is relatively rapid even for high frequencies. A comparison is presented between experimental and numerical results obtained in retrodiffusion, using a transducer of 10-MHz nominal frequency, for alumine and silica inclusions of about 300 μm. Whereas the beam aspect is not taken into account, the wave curves are similar.

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1pPAb5. Depth profiling of residual stress in steel by laser ultrasonics.

Weimin Gao, Silvio Elton, X. Kruger,[a)] Christ Glorieux, Kris Van de Rostyne, Walter Lauriks, Jan Thoen, and Jacques Charlier[a)] (Laboratorium voor Akoestiek en Thermische Fysica, Katholieke Universiteit Leuven, Belgium)

Depth profiling of residual stress in steel is of great interest in many practical applications. Conventional techniques are either destructive (x-ray diffraction and hole drilling) or with obvious restriction (Barkhausen noise). This paper presents a noncontact and nondestructive laser ultrasonics method for determining the depth profile of the residual stress layer in steel samples. It is known that surface acoustic waves penetrate into solids to a depth proportional to the wavelength. They are expected to be dispersive in the presence of gradients in physical properties such as residual stress. Based on the dispersion theory, the depth dependence of an effective velocity $v(z)$, which is related to the stress profile $S(z)$, can be fitted from the frequency dependence of the phase velocity $v(w)$. Due to the advantage of wide bandwidth, laser-generated and -detected surface acoustic waves (SAW) offer a promising nondestructive stress depth profiling method. The profiles obtained by SAW are compared with the stress data obtained by other techniques. [a)] Also at Service de Métallurgie Physique, Université Libre de Bruxelles, Belgique.

3:40–4:00 Break

4:00

1pPAb6. Measurements of pulsed velocity field generated by circular and rectangular piezoelectric transducers directly coupled to a block of aluminum.

N. Ouarradi and B. Piwakowski (Ecole Centrale de Lille, P.B. 48, Villeneuve d'Ascq 59650, France)

A new computational tool Discrete Representation Acoustic Modelling/Solid (DREAM/S), has been developed for computing the displacement field generated by a pulse-excited transducer, directly coupled to an elastic half-space. It allows the separate determination of different components (shear, compressional, radial, axial) of displacement. This method is based on the spatial pulse response approach and gives directly the transient time-domain field. DREAM/S has already been validated mathematically in regard to analytic and exact methods. The main objective of this paper is to show the experiments performed in order to validate this software. First, a brief presentation of DREAM/S and elementary examples of computing are given. Afterward, the experimental results obtained in the case of circular and rectangular transducers directly coupled to a block of aluminum are presented and compared with the ones obtained by DREAM/S. This comparison displays a good agreement and confirms the high precision of this approach. In order to facilitate the interpretation of multichannel records, including the direct and edge waves and the reverberations from the borders of the model, both computed and recorded signals are displayed in specific format, usually applied in the geophysical imaging.

4:20

1pPAb7. Quality assessment of adherent and fusion-welded joints in silicon wafers by nonlinear ultrasound.

A. Wegner, A. Koka, S. Hirsekorn, and W. Arnold (Fraunhofer-IZFP, Univ., Bldg. 37, D-66123 Saarbruecken, Germany, wegner@izfp.fhg.de)

The interest in fusion-welded silicon wafers has increased recently because of their excellent electrical and mechanical properties. Typical applications are in acceleration and pressure sensors. The interface of room temperature bonded wafers contains a physically adsorbed water layer. Bond temperatures higher than 373 K cause a reduction of the adsorbed water by diffusion and thus an increase in bond strength. If electronical components are protected by surrounding silicon wafers, the joining temperature must not be higher than 673 K. The bond strength of the interface essentially defines the reliability of the components. Therefore, a nondestructive method to evaluate the bond quality is required. Binding forces are nonlinear and cause a nonlinear modulation of transmitted or reflected ultrasonic waves. Inhomogeneities in bond strength of the interface can be imaged by the local ultrasonic amplitudes of the insonified frequency and of its higher harmonics measured interferometrically in transmission. A quantitative evaluation of the image data yields information about the variations in bond strength and the size of delaminations and of weakly bonded areas. This paper presents experimental and theoretical investigations of the interface of bonded silicon wafers exploiting the nonlinear transmission of ultrasound. The data are compared to destructive tests.

4:40

1pPAb8. Ultrasonic weld testing using the laser vibrometer detection of a time-reversed A wave.

Bruno Morvan, Alain Tinel, and Jean Duclos (L.A.U.E, Univ. du Havre, Pl. R. Schuman, BP 4006, 76610 Le Havre, France, morvan@iut.univ-lehavre.fr)

This study presents a method of weld inspection by a surface wave called A wave. The A wave propagates without attenuation in an immersed plane plate, so it is suitable for a long-range inspection of large structures. The studied structure consists of a brass plate perpendicularly soldered on to another one. For convenience, the welding is discontinuous and a nonsoldered zone is chosen as reference. The A wave is very dispersive, so a time-reversed method is used to temporally focus it on the weld: this improves the signal-to-noise ratio. An immersed transducer transmits a reversed A-wave signal. A laser vibrometer collects, at several locations along a line parallel to the weld, the incident A-wave signal and its reflection on the weld. At each position, owing to the A-wave time-reversal excitation, the two detected signals have the same duration. The correlation of the two signals allows one to determine the time delay between the incident and the reflected signal. Once the group velocity is known, the topography of the weld can be built and the defects localized.

5:00

1pPAb9. One-bit time reversal through high-order multiple scattering media.

Arnaud Derode, Arnaud Tourin, and Mathias Fink (Laboratoire Ondes et Acoustique ESPCI—Universite Denis Diderot Paris VII 10, rue Vauquelin 75231 Paris Cedex 05, France, arnaud.derode@espci.fr)

A time-reversal mirror (TRM) is an adaptative device that can focus an ultrasonic wave through an inhomogeneous medium. In a typical experiment, a pointlike source transmits a pulse that propagates through the medium. The distorted wavefronts are recorded by a 128-tranducer array. The 128 signals are digitized over 8 bits, stored in a memory, time reversed, and retransmitted by the array through the medium: the resulting wave converges onto the source, despite the inhomogeneities. In the presence of high-order multiple scattering, the TRM still works and takes advantage of it to refocus a pulse with a finer spatial resolution than in a homogeneous medium. Moreover, it is shown that the TRM is still able to focus a wave even if the recorded scattered wave is encoded over 1 bit, only its sign being used. Reducing the number from 8 bits to 1 does not appear to change the lateral resolution; the signal-to-noise ratio is a little higher. Thus the 1-bit TRM appears to be a very robust and low-cost focusing device. Experiments were carried out in a water tank through a random collection of 2400 parallel steel rods.

5:20

1pPAb10. Characterization of time-reversed elastic waves in ergodic billiards.

Julien de Rosny, Carsten Draeger, and Mathias Fink (Lab. Ondes et Acoust., Univ. Paris 7, URA CNRS 1503, ESPCI, 10 rue Vauquelin, 75231 Paris Cedex 05, France, julien.derosny@espci.fr)

One year ago, this laboratory showed experimentally and numerically that an elastic wave can be time reversed in a highly reflecting cavity with a single pointlike source [Draeger and Fink, J. Acoust. Soc. Am. **101**, 3090(A) (1997)]. This is possible because the cavity is low dissipative and its shape is ergodic. Now the properties of this experiment are better understood. Clearly, two parameters are involved. The first one (T) is the beginning of the time reversal window; the second one is its duration ΔT. The lowest value of T which gives an isotropic focusing is the ''angular mixing time;'' it may be related to the ergodicity time. There is another characteristic time: beyond a certain limit, increasing ΔT does not im-

prove the quality of focusing any longer. This ''saturation time'' is very dependent on the initial pulse length; it is analogous to the Heisenberg time. Also, the difference beteween the perfect theoretical time reversal and the experimental one is explained.

5:40

1pPAb11. Backscattering time-reversal of acoustic waves in random media. Arnaud Tourin, Arnaud Derode, and Mathias Fink (Lab. Ondes et Acoust. Univ. Paris VII-ESPCI 10 rue Vauquelin, 75005 Paris, France)

Three years ago, the first experiments showing the reversibility of acoustic waves propagating through high-order multiple scattering media were reported [A. Derode, P. Roux, and M. Fink, Phys. Rev. Lett. **75**, 4206 (1995)]. These experiments were performed in a transmission mode by means of a time reversal mirror (TRM). Here, new time-dependent experimental results obtained in a backscattering mode are reported. The experiment can be described as follows: the TRM used is a linear array of 128 transducers. One of the elements transmits a pulsed wave into the sample which is a random set of 2400 steel rods. In the 128 recorded backscattered signals, short windows are selected which are time-reversed and retransmitted. Surprisingly, it is found that the synthesized time-reversed waves revive their past and converge onto the initial emitting element, despite disorder. Thus the image of the source is recreated, what is referred to as the mirror effect. The spatial refocusing is found to be improved when selecting windows farther and farther in the backscattered signals. The refocused spot time-evolution is well explained by a simple model including both single and multiple scattering contributions.

Contributed Poster

This poster will be on display in the Poster Gallery from Monday to Wednesday, 15–17 March. Author will be at the poster from 10:00 a.m. to 12:00 noon on Tuesday, 16 March.

1pPAb12. A correlation-measuring system for ultrasonic NDT using maximum length sequences. Torsten Niederdraenk (SIEMENS Audiologische Technik GmbH, D-91050 Erlangen, Germany) and Rainer Thaden (Institut fuer Technische Akustik, RWTH Aachen, D-52056 Aachen, Germany)

In the field of nondestructive material testing (NDT), the advantages of a correlation-measuring technique can be used. In particular, the better dynamic range and gain in the signal-to-noise ratio provide a good possibility of investigating strongly scattering or absorbing materials. The application of maximum length sequences as excitation signals permits the performing of the correlation procedure by using the Fast Hadamard transform (FHT), a very fast correlation algorithm. In nondestructive material testing, a number of special requirements has to be fulfilled. Apart from the driving conditions of the piezoelectric transducers, measurements in NDT often require high measuring frequencies. Based on former developments, a measuring system has been developed that provides a high testing rate; even when using MLS signals of order $n=16$, a rate of 20 Hz is obtained. A partially parallel calculation structure of the FHT gives a vivid looking presentation of the measuring results.

MONDAY AFTERNOON, 15 MARCH 1999   ROOM MA041, 2:00 TO 6:00 P.M.

Session 1pPAc

Physical Acoustics: Nonlinear Acoustics

Lawrence A. Crum, Chair
Applied Physics Laboratory, University of Washington, 1013 N.E. 40th Street, Seattle, Washington 98105, USA

Contributed Papers

2:00

1pPAc1. Higher order effects in finite-amplitude sound fields. Sigve Tjøtta (Dept. of Math., Univ. of Bergen, Johs. Brunsgt. 12, 5007 Bergen, Norway and Dept. of Phys. Univ. of Oslo, P.O. Box 1048 Blindern, NO316 Oslo, Norway)

Nonlinear effects associated with intense sound fields in fluids are considered theoretically. Special attention is directed to the study of higher-order effects that cannot be described within the standard propagation models of nonlinear acoustics (the KZK and Burgers equations). The analysis is based on the fundamental equations of motion for a homogeneous, thermoviscous fluid, for which thermal equations of state exist. Model equations are derived and used to analyze nonlinear sources for generation of flow and heat, and other changes in the ambient state of the fluid. Fluctuations in the coefficients of viscosity and thermal conductivity caused by the sound field are accounted for. Also considered are nonlinear effects induced in the fluid by surface waves in the transducer. The intensity and absorption of finite-amplitude sound waves are calculated, and related to the sources for generation of higher-order effects.

2:20

1pPAc2. Modeling and simulating finite-amplitude propagation through time-varying inhomogeneous absorbing media. Ibrahim M. Hallaj (Appl. Phys. Lab, Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, ibrahim@apl.washington.edu), Robin O. Cleveland (Boston Univ., Boston, MA 02215), and Steven G. Kargl (Univ. of Washington, Seattle, WA 98105)

A model equation is derived for three-dimensional wave propagation through an inhomogeneous, time-varying medium. The model equation accounts for arbitrary inhomogeneities, finite-amplitude distortion, and absorption. The effect of time dependence of the background medium parameters is included in the equation. An ordering scheme based on the characteristic time of the evolution of each parameter allows one to evaluate their relative importance. A two-dimensional version of the wave equation is solved in the time-domain using finite-difference methods. Explicit, implicit, and operator splitting techniques were used in the solution to overcome numerical instabilities. Results from the code are compared to known solutions for some simple geometries in homogeneous and

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step indexed media. Various modeling and numerical schemes for treating the absorption on the computational domain interior and at the boundaries of the domain will be presented. [Work supported by ONR and DARPA.]

2:40

1pPAc3. Second-harmonic component in the focused beam transmitted through a weakly dispersive liquid layer. Shigemi Saito (Faculty of Marine Sci. and Technol., Tokai Univ., 3-20-1 Orido, Shimizu, Shizuoka, 424-8610 Japan, ssaito@scc.u-tokai.ac.jp)

The characteristic of the nonlinearly generated second-harmonic component in the focused beam which transmits through a layer consisting of weakly dispersive liquid is theoretically and experimentally investigated. In a focused beam, a destructive interference takes place, within the post focal region, between two portions of second-harmonic components which are generated in the pre- and post-focal regions. This is due to the phase delay of the second-harmonic component passing through the focal region relative to the fundamental component. When a weakly dispersive layer is inserted into the focal region, a phase advance occurs for the second-harmonic component passing through the layer due to slightly higher sound-speed for second-harmonic frequency, together with the phase delay due to diffraction. Then the effect of the destructive interference is degraded to result in amplitude increase of the second-harmonic component within the post-focal region. The experiment for a layer of water–oil mixture employed as a dispersive liquid is compared with the nondispersive case and demonstrates the validity of theoretical prediction.

3:00

1pPAc4. Investigation of nonlinear wave distortion in a focal and post-focal region in water. Grazyna Grelowska, Eugeniusz Kozaczka (Naval Acad., ul. Smidowicza 71, 81-919 Gdynia, Poland), and Grazyna Lypacewicz (Polish Acad. of Sci., 00-049 Warsaw, Poland)

The subject of interest is sources of finite amplitude. Focusing sources belonging to both of the two main differentiated groups of such sources have been examined. The first one consists of sources with additional lens, and the second one of the single-element PZT sources. Previous investigations concerned mainly phenomena occurring close to the source—in a prefocal and a focal region. The phenomena appearing at the longer distances in the post-focal region, where the nonlinear wave distortion is obviously considerably greater than in a pre-focal region, is the main topic of this paper. Experiments are conducted in measurements of spatial pressure distribution in a beam radiated by: (1) a single PZT element focused source with a fundamental frequency of 1.5 MHz, (2) a PZT plane circular source coupled with the plano-concave lens with a frequency of 1.0 MHz. A PVdF hydrophone with diameter of 1 mm was used as a pressure probe. The results of the experiments are compared to the theoretically predicted ones. The sound field was modeled by the KZK equation with the equivalent boundary condition adequate to the actual boundary condition.

3:20

1pPAc5. The decay of pulses with complex structure according to Burgers' equation. Sergey N. Gurbatov, Galina Pasmanik (Dept. of Radiophys., Nizhny Novgorod State Univ., Nizhny Novgorod 603600, Russia), and Bengt O. Enflo (KTH, S-10044 Stockholm, Sweden)

Nonlinear plane acoustic waves propagating through a fluid are studied using Burgers' equation in the two cases: small viscosity and viscosity tending to zero. The evolution of initial pulses with monochromatic and noise carrier is considered. The initial pulses are characterized by two length scales. The length scale for substantial changes of the modulation function is much greater than the corresponding scale of the carrier. Some simple pulses with only one length scale are also studied, since their properties are important for the studies of pulses with two scales. With increasing time, the initial pulses are deformed and shocks appear. The shocks then merge, and finally a finite pulse ends up with an N wave and a periodic signal with a sawtooth wave. The decrease of the energy of the wave with time is investigated for pulses with both monochromatic and

noise carrier and for vanishing and finite but small viscosity. How the characteristics of the final waves depend on the characteristics of the initial waves is also investigated. Both numerical and analytical methods are used.

3:40

1pPAc6. Investigation of the process of self-demodulation of acoustic waves in the river sand. Veniamin E. Nazarov, Vladimir Yu. Zaitsev (Inst. of Appl. Phys., Russian Acad. of Sci., 46 Uljanov St., 603600 Nizhny Novgorod, Russia), and Andrey B. Kolpakov (Inst. of Civil Architecture, 603600 Nizhny Novgorod, Russia)

Experimental investigation and theoretical description of nonlinear self-demodulation of a pulsed high-frequency acoustic wave and propagation of the secondary low-frequency acoustic pulses in dry and water-saturated sand are carried out. The dependence of the propagation time of the video pulses upon the initial static pressure and the relation of the duration and the form of the secondary pulses with the form of the envelope of the initial high-frequency wave were studied. In order to describe the observed dependences, nonlinear equations of state for the considered grain media were suggested. Using the equations, analytical expressions for the demodulated pulse forms were derived. On the basis of matching the theoretical predictions and experimental data, linear and nonlinear acoustic parameters of the sand are determined. The obtained results may be applied to analysis of seismic signals and used for development of seismoacoustic sensing methods, primarily for diagnostic applications of nonlinear effects.

4:00–4:20   Break

4:20

1pPAc7. Effects of micro bubbles vibration for increase of acoustic streaming. Shinichi Sakamoto and Yoshiaki Watanabe (Dept. of Electron., Doshisha Univ., Kyotanabe, Kyoto 610-0321, Japan, tr0141@mail4.doshisha.ac.jp)

The effects of the existence of micro bubbles in water for the increasing of acoustic streaming velocity are experimentally discussed. Velocity of acoustic streaming is observed on the sound axis by LDV (laser Doppler velocimeter). Acoustic streaming is generated by continuous ultrasonic sound transmitted by some PZT transducers whose diameter is 15 mm and resonant frequencies are 2.98 or 3.46 MHz. Sound pressures of ultrasonic sound are set to 80, 160, and 240 kPa at the last peak position on the sound axis. Micro bubbles are generated by extracting water with a syringe mechanism, decreasing the pressure of water down to 6.4% below static pressure. Average radius of micro bubbles is approximately 220 μm. The experimental results clearly show the velocity of acoustic streaming increases; the velocity increases from 9 to 13.5 mm/s when the micro bubbles exist. The harmonics are observed when the micro bubbles are generated. The mechanism for the increase of acoustic streaming is discussed from the point of view of the nonlinear effect of the bubble vibration.

4:40

1pPAc8. Anomalously high elastic nonlinearity and frequency-independent Q-factor as complementary properties of microinhomogeneous solids. Vladimir Zaitsev[a] (Katholieke Univ., Leuven, Belgium)

During the last few decades, a wide class of media demonstrating anomalously high elastic nonlinearity was experimentally revealed (e.g., rocks, grainy materials, concretes). Their nonlinear parameters often exceeded those of homogeneous liquids and solids by several orders of magnitude, whereas linear elastic properties remained ''normal.'' Just the same, materials often possess an almost frequency-independent Q-factor, which also is quite different from homogeneous liquids. These acoustical peculiarities may be evidently attributed to the influence of structural microinhomogeneities (e.g., cracks, grains, etc.) typical of the mentioned media. Recently, a few physical models describing nonlinearity of grainy

and crack-containing materials were proposed, while the description of the dissipation (almost frequency-independent Q-factor) still is restricted to phenomenological models. The report presents a rather demonstrative model of microinhomogeneous solids, which allows for explanation of the mentioned facts. It follows naturally that the microinhomogeneities may not cause significant change of linear elasticity, whereas a sharp increase of elastic nonlinearity and occurrence of the frequency-independent Q-factor appear to be complementary manifestations of the same microstructure. The model readily allows one to relate medium structural features with its elastic and dissipative properties. [Work supported partially by RFBR and Academic Board of KUL.] [a] Permanent address: Inst. of Appl. Phys. RAS, 46 Uljanova St., Nizhny Novgorod, 603600, Russia.

5:00

1pPAc9. Dissipation in solids: Thermal oscillations of atoms. Adnan Akay and Cem Celik (Carnegie Mellon Univ., Pittsburgh, PA 15213)

Dissipation in a solid is related to the increase in the average kinetic energy of its atoms as a result of external excitation. This paper investigates the relationship between vibration field of a nonlinear lattice of atoms and its energy absorption properties. Numerical results show that a nonlinear lattice accepts energy when excited within its resonance bands. The width of the resonance bands increases with increased nonlinearity, eventually leading to their overlap.

5:20

1pPAc10. ''Acoustical modeling'' according to the theory of the common networks—Taking into account nonlinear effects. Joern Huebelt and Ennes Sarradj (TU Dresden, Inst. fuer Technische Akustik, Mommsenstr. 13, 01062 Dresden, Germany)

One method to model the electroacoustical transducer consists of describing its basic system components using lumped elements according to the theory of the common networks. In the past this technique has been widely used. However, the validity of the underlying simplifications has to

be proofed in particular cases. The modeling of the acoustic behavior of a signal horn shall demonstrate the efficiency of this method. In a first step, a linear model in the frequency domain was established. It turned out that for a more exact simulation, the consideration of nonlinear effects is necessary. Due to the extension to nonlinear components, the analysis of the system has to be carried out in the time domain. The model provides the possibility to include the results of the FEM analysis or the BEM analysis, for those system parts which cannot be reduced to lumped elements. Moreover, it enables a relatively fast and inexpensive examination of the system characteristics compared to the FEM or the BEM analysis and can be easily adapted to similar horn types.

5:40

1pPAc11. On ambiguity in the description of sources and the choice of the acoustic variable. Ricardo E. Musafir (PEM/COPPE and DHS/EE, Universidade Federal do Rio de Janeiro, C.P. 68503, RJ, 21945-970 Brazil, rem@serv.com.ufrj.br)

In a homogeneous medium at rest, a classical problem of source ambiguity is expressed by the fact that one cannot differentiate between the fields of a volume displacement point monopole and of a volume acceleration isotropic point quadrupole. In more complex situations, a similar identity concerning the fields of more elaborate combination of point sources can be shown to exist. This reflects the fact that, since two types of energy are involved in the propagation, one cannot tell, from the wave field alone, of which type was the action responsible for originating the wave, there existing, ideally, always a way of representing a ''mass'' source by a combination of ''momentum'' sources and vice versa. It is shown, however, that, depending on the choice of the acoustic variable—and thus on the operations needed in order to combine the continuity and momentum equations into a single equation, this elegant picture may not be clearly displayed in the wave equation. The particular case of the use of the stagnation enthalpy as dependent variable is examined as an example, the analysis being used to discuss its adequacy as a variable for aerodynamic noise problems.

Contributed Poster

This poster will be on display in the Poster Gallery from Monday to Wednesday, 15–17 March. Author will be at the poster from 10:00 a.m. to 12:00 noon on Tuesday, 16 March.

1pPAc12. Nonlinear effect of the inertia of the fluid on acoustic streaming in cylindrical guides. Ludovic Menguy and Jöoel Gilbert (Lab. d'Acoust. de l'Univ. du Maine UMR CNRS 6613, Ave. Olivier Messiaen, 72085 Le Mans Cedex 9, France)

Acoustic streaming is the mean flow created by a high sound level wave. Classical treatment [J. W. S. Rayleigh, Philos. Trans. R. Soc. London **175**, 1–21 (1884)] considers that in cylindrical guides a standing acoustic wave creates a symmetrical toroidal vortex (slow streaming) due to the action of Reynolds stress forcing resisted by viscosity. A perturbation method followed by a time averaging applied to the conservation equations leads to the equations describing the behavior of the mean flow

velocity in a cylindrical air-filled guide. According to a dimensional analysis, the effect of the fluid inertia cannot be neglected if the streaming becomes perceptible. This effect, previously studied for free or semi-infinite space, is controlled in the case of a waveguide by the dimensionless parameter $Re_{nl} = M^2/Sh^2$ (M and Sh are the acoustic Mach number and the Shear number). Numerical resolution of this nonlinear system of equations (Newton–Raphson method) is performed using nonslip conditions on the tube wall and mass flux conservation accross the section of the tube. Results indicate that inertia distorts perceptibly the streamlines, and renders vortex pattern unsymmetrical. A comparison of slow and nonlinear acoustic streaming is finally achieved.

Session 1pPAd

Physical Acoustics: Cavitation Physics and Sonoluminescence I

Werner Lauterborn, Cochair

Drittes Physikalisches Institut, Universität Göttingen, Burgerstrasse 42-44, D-37073 Göttingen, Germany

Ronald A. Roy, Cochair

Department of Aerospace and Mechanical Engineering, Boston University, Boston, Massachusetts 02215, USA

Invited Papers

2:00

1pPAd1. Sonoluminescence: Current status and future perspectives. Lawrence A. Crum and Thomas J. Matula (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105)

Considerable recent attention has been directed at the phenomenon of sonoluminescence, especially since the discovery of single-bubble sonoluminescence (SBSL), in which a single, stable, acoustically levitated gas bubble can be made to pulsate with a sufficiently large amplitude to emit light each acoustic cycle. It appears that exciting physics may be associated with this phenomenon. When a sufficiently strong acoustic field is propagated through a liquid containing microscopic cavitation nuclei, similar optical emissions can be observed, although the spectrum of this multiple bubble sonoluminescence (MBSL) often possesses characteristics of the host liquid and its constituents, unlike the spectra of SBSL, which has no discernible bands or lines. Nevertheless, MBSL spectra should be able to shed considerable light on the discipline of sonochemistry. Relevant aspects of the current status of SBSL and MBSL research will be reviewed, some speculations on future directions will be offered. [Work supported in part by the NSF.]

2:20

1pPAd2. The cavitation collapse on a sub-ns time scale. B. Gompf, R. Pecha, and W. Eisenmenger (1. Physikalisches Inst., Univ. Stuttgart, Pfaffenwaldring 57, D-70550 Stuttgart, Germany)

A single sonoluminescing air bubble trapped in the pressure maximum of a resonant sound field in water is an ideal model system to investigate the end phase of the cavitation collapse. The dynamics of these single bubbles can be characterized with Mie-scattering. In earlier experiments, the scattered light was detected with photomultiplier tubes (PMT) where the time resolution is limited by the response of the PMT and no spatial resolution is possible. Using a streak camera, the scattered light can be recorded with high spatial and temporal resolution. The streak images show that at the minimum radius the scattered light intensity is not only a function of $R(t)$ anymore, and the changes in the refractive indices inside the bubble and in the highly compressed water surrounding the bubble have to be considered. Together with the width and intensity of the emitted light pulses, the results represent a complete data set for the end phase of the bubble collapse.

2:40

1pPAd3. Measuring cavitation conditions by multibubble sonoluminescence. Kenneth S. Suslick, William B. McNamaraIII, and Yuri T. Didenko (Univ. of Illinois, 600 S. Mathews Ave., Urbana, IL 61801, ksuslick@uiuc.edu)

Multibubble sonoluminescence (MBSL) arising from ultrasonic irradiation of solutions containing volatile transition metal carbonyls have been collected and analyzed. The principal emission is the atomic line spectrum from excited states of the metal atoms. This metal atom MBSL has been used to determine the effective emission temperature of cavitation as a function of the dissolved gas, and the effects of changes in the ratio of the heat capacities and of the thermal conductivity of the dissolved gas have been examined. The results strongly support a simple near-adiabatic compression model of cavitation. The results also indicate that the continua in MBSL spectra are due to emission from small molecules, and not due to plasma emission. [Work supported by the NSF and in part by the DOE.]

3:00

1pPAd4. Simulation of bubble motion in acoustic cavitation. Robert Mettin, Stefan Luther, Claus-Dieter Ohl, and Werner Lauterborn (Drittes Physikal. Inst., Univ. Göttingen, Bürgerstrasse 42-44, D-37073 Göttingen, Germany, R.Mettin@physik3.gwdg.de)

The motion of cavitation bubbles in an acoustic field is investigated both experimentally and numerically. To simulate the experimental results, a particle approach is employed which calculates the trajectories of many individual bubbles. For this purpose, added mass, drag, and Bjerknes forces are considered, and nucleation and coalescence of bubbles are modeled. Though the particle model is based on simplifying assumptions, it can reproduce different cavitation structure types observed in experiments. [Work partially supported by "Graduiertenkolleg Strömungsinstabilität und Turbulenz."]

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Joint Meeting: ASA/EAA/DEGA   **959**

1pPAd5. Bubble dynamics and SBSL in a variable acceleration environment. R. Glynn Holt, Ronald A. Roy, and Sean C. Wyatt (Dept. of Aerosp. and Mech. Eng., Boston Univ., 110 Cummington St., Boston, MA 02215, rgholt@bu.edu)

In order to prepare for pending experiments in variable g environments, some of the multiple effects caused by the time-varying acceleration of acoustic resonators used in bubble levitation experiments were considered. The coupled effects of the induced changes in ambient pressure (due to a changing hydrostatic head) and bubble position (due to a change in the buoyant body force) were modeled. Changing the ambient pressure, while holding the acoustic pressure amplitude constant, causes changes in the radial bubble response and diffusive equilibrium requirements. Changing the bubble levitation position causes both the local acoustic pressure amplitude and gradient to change, which will again impact bubble response. If the bubble is required to remain in a stable diffusive equilibrium, both of these effects will force the bubble equilibrium radius to change. Finally, by using an empirical relation for emitted SL intensity versus bubble response, the variation of emitted light intensity as a function of the changing ambient acceleration is obtained. Where possible, these results are compared to experimental data. [Work supported by NASA.]

3:40–4:00 Break

Contributed Papers

4:00

1pPAd6. Single-bubble large-scale cavitation. Vadim A. Simonenko, Vladimir N. Nogin, Nikolay G. Karlykhanov, Gennadiy V. Kovalenko (Russian Federal Nuclear Ctr., ITP, P.O. Box 245, 456770 Snezhinsk, Chelyabinsk region, Russia, sva@sva.ch70.chel.su), and William C. Moss (Lawrence Livermore Natl. Lab., Livermore, CA 94550)

Recent successes in experimental and theoretical studies of single-bubble sonoluminescence (SBSL) suggest the possibility of achieving high energy densities in the collapsed bubble. Scaling analyses and efficient ways for cavitation enhancement are discussed, with particular attention to the geometrical size of the system components and ambient pressure level. The goal is to increase the energy density in the collapsed bubble and increase the size and mass of the heated region. Theoretical estimations and results of direct hydrodynamic numerical simulations are presented for a prospective system with a 0.01- to 0.1-mm-radius gas-filled cavity that is surrounded by a 0.3-m-radius liquid-filled shell. The liquid-filled shell is surrounded by two concentric gas-filled spherical shells. The inner shell contains pressurized hydrogen, whereas, the outer shell contains oxygen. The initial rarefaction phase, which caused bubble expansion, is caused by a sudden release of the hydrogen gas into the outer oxygen-filled shell. The subsequent compression phase results from the combustion of the hydrogen and oxygen gases. [Work was partly supported by CRDF].

4:20

1pPAd7. Single-bubble sonoluminescence: Investigation of the emitted pressure wave with a streak camera and a fiber-optic probe hydrophone. R. Pecha, Z. Q. Wang, B. Gompf, and W. Eisenmenger (1. Physikalisches Inst., Univ. Stuttgart, Pfaffenwaldring 57, D-70550 Stuttgart, Germany)

In the collapse end phase of single-bubble sonoluminescence (SBSL), in addition to a short light pulse, a spherical pressure wave is emitted by the bubble into the surrounding liquid. This pressure wave was investigated with two different methods: (1) If the bubble is illuminated by a laser beam, laser light is Mie-scattered by the bubble itself, but also by the outgoing pressure pulse. The scattered light of both was recorded with a streak camera with a spatial resolution of 8 μm and a temporal resolution of 500 ps. From the time-dependent radial distance of the pressure pulse $r(t)$ from the bubble, the velocity $v(t)$ can be determined. The speed of sound in the vicinity of the bubble is increased in comparison to normal conditions. This change in the sound velocity was used to estimate the amplitude of the pressure pulse. (2) At a distance of 2.5 mm from the bubble, a fiber-optic probe hydrophone with a spatial resolution of 100 μm and a rise time of 5 ns was used to measure the pressure wave. Measurements depending on the amplitude of the driving sound field, the gas concentration, and the temperature of the water will be presented.

4:40

1pPAd8. Experimental observations of the effect of ambient pressure on single-bubble sonoluminescence. M. Dan, J. D. N. Cheeke (Phys. Dept., Concordia Univ., 1455 de Maisonneuve Blvd. W., Montreal, QC H3G 1M8, Canada, manas@alcor.concordia.ca), and L. Kondic (Duke Univ., Durham, NC 27708)

The experimental observations of the influence of ambient pressure on single-bubble sonoluminescence will be presented. In the SBSL regime as the ambient pressure was decreased only about 10%, the SL intensity increased significantly as predicted [L. Kondic, C. Yuan, and C. K. Chan, Phys. Rev. E **57**(1), R32–R35 (1998)]. There was an increase of about five times in SL intensity as the ambient pressure was decreased by about 10%–12%, at constant driving acoustic pressure. This increment continued until a certain value of the ambient pressure, below which the bubble became unstable and vanished. The equilibrium radius and the maximum radius of the SBSL bubble both inceased with decrease of ambient pressure. This observation provides valuable insight into the mass flow mechanism between the bubble and the liquid, and gives the first direct confirmation of ''argon bubble hypothesis.'' [Work supported by NSERC and Concordia Seagram Fund for Innovative Research.]

5:00

1pPAd9. Parameter space of single-bubble sonoluminescence for various liquid temperatures. D. Felipe Gaitan (NCPA, Univ. of Mississippi, University, MS 38677) and R. Glynn Holt (Boston Univ., Boston, MA 02215)

In a previous paper by the authors [Phys. Rev. Lett. **77**, 3791 (1996)], the region of parameter space (acoustic pressure, ambient bubble radius) in which stable single-bubble sonoluminescence (SBSL) occurs in an air–water system driven at 20.6 kHz was described as a function of the dissolved gas concentration. In this paper, new data (ambient radius, maximum radius, expansion ratio, as well as the emission and extinction thresholds) will be presented for the SBSL parameter space at various liquid temperatures for different gases and liquids. [Work supported by the Office of Naval Research.]

5:20

1pPAd10. Molecular dynamics approach for a sonoluminescing bubble. Burkhard Metten, Thomas Kurz, and Werner Lauterborn (Drittes Physikalisches Institut, Univ. Goettingen, Burgerstr. 42-44, D-37073 Goettingen, Germany, metten@dpi.physik.uni-goettingen.de)

Up to now the numerical calculations for the converging shock or compression wave model of sonoluminescence have been based primarily on continuum methods. Here an alternative approach by molecular dynamics simulation is shown to be feasible with today's computer memory and speed. Both approaches are compared. Results of the simulations are presented for the scaling behavior with the number of particles, for different boundary conditions (spherical and ellipsoidal bubbles) and for one and

more species in the bubble. The dependence of the light intensity and pulse duration on the sphericity of the bubble is discussed. [Work supported by the Graduiertenkolleg Stroemungsinstabilitäten und Turbulenz.]

5:40

1pPAd11. Single-bubble sonoluminescence in water during transitions from hypergravity to low gravity on NASA's KC-135 aircraft and laser-beam extinction methods for studying bubble dynamics. David B. Thiessen, Jeremy E. Young, Mark J. Marr-Lyon, Ben R. Dzikowicz, and Philip L. Marston (Dept. of Phys., Washington State Univ., Pullman, WA 99164-2814)

The sound field which drives SBSL also provides the radiation force which counteracts the average buoyancy of the bubble. One consequence is that the location of the bubble should be altered by the effective acceleration of gravity g_e. Variations in g_e may alter the physical processes giving rise to luminescence. This group's previous experiments have confirmed that SBSL is not automatically quenched in the reduced and enhanced g_e conditions in an aircraft undergoing parabolic flight trajectories [D. B. Thiessen *et al.*, in *Proc. of the 4th Microgravity Fluid Phys. Conf.* (NASA, 1998), pp. 379–383]. The new experiments were carried out with the SBSL chamber in contact with a constant-pressure gas-filled chamber. During intervals of negligible drift in the SBSL intensity, there can be a rapid intensity rise (with a relaxation time of about 5 s) of about 4% as g_e is decreased from 1.8 to near 0 g, but the increase is not seen in all data sets. In related work, diagnostics based on monitoring laser-beam extinction with a photocell [J. S. Stroud and P. L. Marston, J. Acoust. Soc. Am. **94**, 2788–2792 (1993)] is applied to SBSL bubbles in the laboratory. [Work supported by NASA.]

1p MON. PM

MONDAY AFTERNOON, 15 MARCH 1999   POSTER GALLERY, 2:00 TO 4:00 P.M.

Session 1pPAe

Physical Acoustics: Acoustic Evaluation of Materials (Poster Session)

Torsten Niederdraenk, Cochair
Siemens Audiologiesche Technik GmbH, D-91050 Erlangen, Germany

Jean Francois Allard, Cochair
Institut d'Acoustique et de Mecanique de l'Universite du Maine, UMR CNRS 6613, av. Olivier Messiaen, F-72085 Le Mans, Cedex 9, France

Contributed Papers

All posters will be on display in the Poster Gallery from Monday to Wednesday, 15–17 March. Authors will be at their posters from 2:00 p.m. to 4:00 p.m. on Monday, 15 March.

1pPAe1. Acoustic emission signals from the sol−gel transition of tetraethoxysilan (TEOS) and sodium silicate solutions. Jadwiga Rzeszotarska, Feliks Rejmund (Inst. of Fundamental Technolog. Res., Polish Acad. of Sci., Swietokrzyska 21, 00-049 Warsaw, Poland, freymund@ippt.gov.pl), and Jerzy Ranachowski (Polish Acoust. Society, 00-049 Warsaw, Poland)

Sol–gel processes have been used extensively to prepare oxide materials (glasses and ceramics). The acoustic emission (AE) method with broadband piezoelectric transducer (up to 500 kHz) was used to control the evolution of slow gel formation from organic TEOS and fast formations from sodium silicate as the precursors of silica. The obtained dependencies of the AE count rate and rms values versus time characterize the dynamic of the cross-linking and desolvation processes under ambient conditions. The frequency distribution analysis of individual power spectra provides an activity estimation of integral parts of processes. In sodium silicate solutions the spectra of low frequency (17–70 kHz) were mainly observed. On the other hand, the most representative spectra from the gelation of TEOS display at three basic ranges; 16–20 kHz, 200–230 kHz,

and 400–470 kHz. The relative intensities of spectra in these ranges were changed considerably during the gel formation. [Work partly supported by KBN Grant No. T 07B 047 15.]

1pPAe2. Ultrasonic velocity and absorption in the binary mixtures of 1-butanol with isomeric butanediols at 298.15 K. Edward Zorebski and Michal Zorebski (Inst. of Chemistry, Silesian Univ., Szkolna 9, 40-006 Katowice, Poland, emz@tc3.ich.us.edu.pl)

From the early days of ultrasonic technology, ultrasonic velocity and ultrasonic absorption have been a rich source of information on the structure and condition of the materials trough in which the ultrasonic waves are propagated. Here an investigation into binary mixtures (in the whole concentration range) of 1-butanol with 1,2-butanediol and 2,3-butanediol at 298.15 K is reported. For the ultrasonic velocity measurements at 2.15 MHz, the pulse-echo-overlap method was used and the ultrasonic absorption in the frequency range 10–200 MHz was determined by the standard pulse technique, where the amplitude of the first transmitted pulse was registered as a function of distance. The results were compared with the previous data for 1-butanol with 1,3-butanediol and 1,4-butanediol and discussed in terms of molecular interactions in highly associated liquids.

1pPAe3. Ultrasonic study of ionic association in aqueous solutions of lanthanide salts. Birutė Voleišienė, Gintara Miglinienė, and Algirdas Voleišis (Ultrasound Res. Ctr., Kaunas Univ. of Technol., Studentųstr. 50, LT-3031 Kaunas, Lithuania, ulab@tef.ktu.lt)

The association reactions between trivalent ions of lanthanum, cerium, neodymium, gadolinium, erbium, ytterbium, and nitrate or sulphate ligands has been investigated from ultrasound velocity dispersion measurement data. It is assumed that ultrasound velocity dispersion in aqueous solutions of lanthanide nitrates and sulphates is caused by an association process. The ultrasound velocity was measured by the ultrasonic laser interferometer within the frequency range of 3–200 MHz. The rates of formation of inner sphere complexes $[LnNO_3]^{2+}_{aq}$ and $[LnSO_4]^{+}_{aq}$ were calculated and the interaction between individual ions Ln(III) and water molecules is discussed across the series. A comparison of relaxation parameters for the nitrate solutions with corresponding sulphates was made. It has been shown that a magnitude of an ultrasound dispersion may be the quantity of complexation. Ultrasound velocity dispersion for nitrate solutions is much smaller than that of more diluted lanthanide sulphate solutions. The lanthanide nitrates form predominantly outer sphere complexes with some inner sphere substitution. In order to produce an appreciable amount of inner complexes in nitrate solutions for a similar dispersion and relaxation, a tenfold higher concentration of the nitrates must be used.

1pPAe4. Effect of wall adsorption on the photoacoustic detection of ammonia with a 1.53-μm DFB diode laser. Andreas Hebach, Andras Miklos, and Peter Hess (Inst. of Physical Chemistry, Univ. of Heidelberg, Im Neuenheimer Feld 253, D-69120 Heidelberg, Germany, andreas.miklos@urz.uni-heidelberg.de)

The adsorption of ammonia molecules (and other polar molecules such as nitrogendioxide, water, etc.), on the walls of the photoacoustic (PA) detector influences the accuracy of the concentration measurement. This effect becomes larger for small concentrations and depends on the direction of the concentration change. This means that a hysteresis of the PA signal is observed between measurements carried out by decreasing and by increasing the concentration. The adsorption problem was reduced considerably by performing the measurements in a continuous flow. For measurements with high flow rate a new photoacoustic detector was developed. The differential design provides good flow and electronic noise suppression. The PA signal has been generated by tuning a 40-mW, 1.53-μm GEC-Marconi DFB laser to the absorption peak of ammonia at 1527 nm. The dependence of the PA signal on the flow rate of ammonia–nitrogen mixtures with different concentrations was investigated. In another experiment the hysteresis of the PA signal was investigated by increasing and decreasing the concentration in several steps, but maintaining the same total flow rate. It was found that a higher flow rate results in a smaller error of the concentration measurement and a shorter response time of the PA detector.

1pPAe5. The effects of ultrasonic waves on dye penetration with cellulose fabrics dyed by vinylsulphonic dyes. Vesna Ignjatovič, Mile Novakovič (Faculty of Technol., Leskovac Bulevar, oslobo/enja 124, 16000 Leskovac, vesnateh@tehfin.tehfak.ni.ac.yu), and Milena Miljkovič (Leskovac Bulevar, 16000 Leskovac)

Research has shown that ultrasound traveling in the form of waves through liquids causes the effect of cavitation. Cavitation is manifested in a continuous creation and disappearance of vacuums within the liquid. Therefore, interruptions are created in the liquid, that is, strong forcing waves are made. The cavitation strikes are used in solid matter destruction, emulsion creation, greasy surface cleansing, and the like. The ultrasonic energy has been applied in textile industry, for the most part in soaking processes, and textile washing and cleaning. There is research indicating that ultrasonic energy can be used in dyeing of cotton with direct and reactive dyes, of polyamid and wool with acid dyes, and of polyester and acetate fibers with disperse dyes. Based on previous research, this paper is an account of an experiment whose aim was to imple-

ment the latest achievements in vibration and ultrasonic theory in textile dyeing technologies. Thus cotton knitwear has been dyed with vinylsulphonic reactive dye, with and without the use of ultrasonic energy, and with the introduction of an air bubble in the dye tub with the application of ultrasonic energy. Previous experiments have shown that ultrasound contributes to greater dye absorption from the tub, so that equal distribution dyeing with the use of ultrasound requires less dye, shorter period of dyeing, less power consumption, and at the same time the pollution of waste waters is reduced. All these characteristics make this procedure economically and ecologically affordable. So as to intensify the process, this research has been directed towards the introduction of an air bubble in the dyeing tub which is under the effect of ultrasound. Dye absorption from the tub and dye penetration in the fabric have been monitored by means of spectrophotometry (absorption and reflection spectrophotometry). Previous research projects have not been able to explain completely the fundamental interaction of ultrasound and fibers, that is, reactive dyes in water, and with that, in the process of dyeing itself. Therefore, future work will deal with this problem. [This paper is a part of research done for the requirements of the project titled ''Modern Approaches in New Dyeing Procedure Development for Optimal Pollution of the Environment,'' and financed by Science and Technology Ministry of Serbia.]

1pPAe6. The effects of ultrasonic energy on reactive dye absorption in the presence of reduced quantities of salt. Mile Novakovič, Vesna Ignjatovič (Faculty of Technol., Leskovac Bulevar oslobo/enja 124, 16000 Leskovac, vesnateh@tehfin.tehfak.ni.ac.yu), and Milena Miljkovič (Leskovac Bulevar, 16000 Leskovac)

Ultrasonic energy has shown a wide range of application possibilities in many fields of human craftsmanship, and its application in textile industry has been a matter of research for a number of years. For the time being, the best results have been achieved in other research still being done in laboratories. This paper presents a part of the overall research done within the project ''Modern Approaches in New Dyeing Procedure Development for Optimal Pollution of the Environment,'' financed by Science and Technology Ministry of Serbia. One of the predetermined objectives of the project is to examine the possibility of ultrasonic energy application in technological processes of dyeing cellulose fabrics by vinylsulphonic dyes. Having that in mind, it is necessary to systematically investigate all of the aspects of ultrasonic energy effects on textile substrata, the dye, and the relation of dye solution and fiber. In the experiments, cellulose microfiber knitwear has been dyed with vinylsulphonic dyes according to the on-line procedure, at a stable temperature, with the addition of varying quantities of salt. Concerning the fact that salt has the ability of enhancing dye absorption from the tub, and that ultrasonic energy helps the process of absorption, the quantity of salt required may be reduced, and thus reduce the cost of dyeing and the pollution of waste waters. Dye absorption from the tub has been monitored by means of absorption spectrophotometry. With the method of reflection spectrophotometry, spectral remission curves have been measured, K/S values calculated, and respectively the relative intensity of dyeing on the fabric determined.

1pPAe7. Acoustic emission (AE) generated by thermomechanical stress in ceramics. Przemyslaw Ranachowski (Inst. of Fundamental Technolog. Res., Polish Acad. of Sci., Swietokrzyska 21, 00-049 Warsaw, Poland)

The thermomechanical long-term toughness of electrotechnical porcelain of long-rod insulators was investigated. The objects were stressed in thermostat to 63% of nominal strength. The cyclic temperature changes from −25 deg to 40 deg were applied. This treatment corresponds to real environmental conditions on insulators of high-voltage lines. The AE

method was used for monitoring the gradual destruction process during the test. The initiation and propagation of microcracks correlate with AE count rate. It was stated that the magnitudes of consecutive maxims of count rate during the cooling part of the cycle depend on the advancement of material destruction. The increase of these maxims means reduction of ceramics toughness related with multiplication of microcracks number and larger probability of critical cracks appearance. When the ceramics retains its strength, the maxims have approximately the same value in consecutive test cycles. The AE method was successfully applied in product certification procedure and control of electric power system. The degree and parameters of porosity of ceramics have been defined on the basis of velocity and attenuation of ultrasound. The good agreement of theoretical and experimental data was obtained. [Work supported by KBN Grant No. T 07B 03413.]

MONDAY AFTERNOON, 15 MARCH 1999   ROOM H1012, 1:55 TO 6:20 P.M.

Session 1pPPa

Psychological and Physiological Acoustics: Recent Advances in Models of Auditory Processing

William A. Yost, Cochair

Parmly Hearing Institute, Loyola University, 6525 North Sheridan Road, Chicago, Illinois 60626, USA

Torsten Dau, Cochair

AG Medizinische Physik, University of Oldenburg, D-26111 Oldenburg, Germany

Chair's Introduction—1:55

Invited Papers

2:00

1pPPa1. Modeling within- and across-channel processing of amplitude modulation. Torsten Dau (AG Medizinische Physik, Univ. of Oldenburg, Oldenburg, Germany)

Recently, a model was presented in which a modulation filterbank was introduced to analyze the envelope fluctuations at the output of each excited peripheral auditory filter [Dau *et al.*, J. Acoust. Soc. Am. **102**, 2892–2905 (1997)]. As a decision stage, an optimal detector was applied which combines all modulation-filter outputs linearly, assuming independent observations. The model accounts for temporal modulation transfer functions (TMTF) with narrow band and broadband carriers as well as for modulation masking data. The model does not cover conditions which require across-channel processing, such as modulation detection interference (MDI). In this case, the auditory system seems to combine information across frequency channels whose responses are temporally modulated. A new modeling approach is presented that integrates the modulation-filter outputs across frequency channels prior to the decision stage, while preserving the original model's ability to describe TMTFs and modulation masking. To further ensure the possibility of predicting spectral masking, a low-frequency part of the modulation spectrum, including the dc component, is processed separately within each peripheral channel. The model can account for the main features of MDI, but it does not account for comodulation masking release.

2:20

1pPPa2. Computer models of the auditory periphery. Ray Meddis (Ctr. for the Neural Basis of Hearing at Essex, Essex Univ., Colchester CO4 3SQ, UK, rmeddis@essex.ac.uk)

Enough is now known about the auditory periphery to allow the development of useful computational models. These accept an arbitrary acoustic stimulus and predict the response of auditory nerve fibers. Current models are largely based on linear filters representing the mechanical frequency selectivity of the cochlear. Nonlinear models would be more appropriate if they could be developed in a form that replicated physiological measurements. This paper describes one approach that combines a nonlinear model of frequency selectivity with a revised inner hair cell model. The nonlinear filters are evaluated using published data collected at three sites on the basilar membrane. They behave nonlinearly at frequencies close to best frequency and linearly at remote frequencies. They show appropriate impulse responses and patterns of two-tone suppression. The hair cell model incorporates a simple account of the receptor potential and retains a previously developed reservoir model of synaptic flow. An innovation includes a model of the discrete release of transmitter quanta that explicitly assumes a presynaptic origin for the stochastic nature of the auditory nerve response. The combined model gives a useful account of the differences between low-, medium-, and high-spontaneous rate fibers that is consistent with earlier physiological speculation.

1pPPa3. Modeling the effects of peripheral compression on temporal processing. Andrew J. Oxenham (Dept. of Speech-Lang. Pathol. & Audiol., Northeastern Univ., Boston, MA 02115)

Most psychoacoustic models to date have assumed that the peripheral auditory system can be treated as a bank of quasilinear bandpass filters. Physiological studies have shown that this approximation holds only for damaged cochleae: the response of the basilar membrane in a healthy cochlea to sounds around the characteristic frequency is highly nonlinear and compressive over a wide range of sound-pressure levels. Many aspects of psychoacoustic performance can be explained by taking these nonlinear characteristics into account. Short-term temporal integration in simultaneous and forward masking (where signal detectability improves with increasing signal duration), the critical masking interval (where the detectability of a brief signal temporally centered in a masker decreases with increasing masker duration), and the nonlinear growth of forward masking can be modeled with relatively simple assumptions if a physiologically reasonable nonlinearity is included. Similarly, some aspects of temporal processing in listeners with cochlear hearing impairment can be well described by making the reasonable assumption that cochlear compression is reduced or absent. Thus the altered temporal processing shown in some tasks by hearing-impaired listeners may in many cases be due to reduced cochlear compression and not to more central processing deficits. [Work supported by the NIH/NIDCD.]

1pPPa4. Recent advances in models of binaural detection and localization. H. Steven Colburn (Hearing Res. Ctr. and Biomed. Eng., Boston Univ., 44 Cummington St., Boston, MA 02215, colburn@enga.bu.edu)

There has been a significant increase in the mathematical modeling of binaural phenomena in the past couple of years by a number of research groups. Available review chapters [e.g., Colburn, in *Auditory Computation* (Springer-Verlag, New York, 1996); Stern and Trahiotis, in *Handbook of Perception* (Academic, New York, 1996)] do not include this new material. Recent advances were stimulated in part by the availability of more powerful computational tools and in part by recent data that include more complex stimulus situations such as noise waveforms with interaural time differences but no interaural intensity differences and vice versa [van de Par and Kohlrausch, J. Acoust. Soc. Am. (1998)]. This presentation reviews recent developments in binaural modeling with particular attention given to models of binaural detection and sound localization. Both pink-box models, which incorporate available information from auditory physiology, and black-box models are considered. [Work supported by NIDCD Grant No. R01 DC00100.]

1pPPa5. Temporal models of pitch processing. William A. Yost (Parmly Hearing Inst., Loyola Univ. Chicago, 6525 N. Sheridan Rd., Chicago, IL 60626)

Most models of pitch perception assume three basic stages of processing: cochlear filtering, haircell/neural transduction, and pitch extraction. In temporal models of pitch processing the pitch extraction stage is often based on autocorrelationlike processes. This talk will review our work using regular interval stimuli to study temporal models of pitch perception. Work with regular interval stimuli suggests that temporal models of pitch processing can account for the pitch and pitch strength of complex sounds with both resolved and unresolved harmonics. The pitch and pitch strength of regular interval stimuli appears to be based on the temporal properties of the waveform fine structure. Human and animal psychophysical results, physiological recordings from the cochlear nucleus, and model results will be presented. [Work supported by NIDCD and AFOSR.]

3:40–4:00 Break

Contributed Papers

1pPPa6. Cochlear wavelength representation of acoustic frequency content. Timothy A. Wilson (Dept. of Elec. Eng., Univ. of Memphis, Memphis, TN 38152-6574)

Peak-to-peak wavelengths of the sinusoidal steady-state response of a simple cochlear model (i.e., incompressible, inviscid, linear fluid in a rigid box-shaped chamber with a linear stiffness-damping-mass cochlear-partition model having exponential parameter variation with place) were examined. For a wide range of mechanical parameter values and for a wide range of cochlear places (at least those basalward of the best place, though sometimes extending through the best place toward the cutoff regime), the wavelengths were found to be (1) a decreasing, approximately linear function of the logarithm of the stimulus frequency, and (2) a decreasing, approximately linear function of cochlear place. A simple algorithm for estimating the instantaneous frequency of sinusoidal signal components of complex signals (both constant frequency and chirps), as well

as that of the center frequency of pulse- and noise-excited resonances, was developed. Estimator performance for the cases of signals plus noise and of noisy wavelength measurements will be presented.

1pPPa7. Evaluation of a physiological ear model for the simulation of nonlinear masking effects. Frank Baumgarte (Inst. fuer Theoret. Nachrichtentechnik und Informationsverarbeitung, Univ. of Hannover, Appelstr. 9A, D-30167 Hannover, Germany)

A physiological model of the human auditory system with the aim at generating masked thresholds for arbitrary sound signals was presented at the previous DEGA meeting. The main parts of the ear model are an active nonlinear cochlear model and a threshold detector which evaluates changes in specific loudness. It was shown that the ear model is able to rebuild the level dependency of spectral and temporal masking patterns,

the asymmetry of masking between noise and tone, and the different detection thresholds for amplitude and frequency modulation. This study presents new simulation results for nonlinear masking effects. These results indicate that the nonlinear ear model allows demonstration of other nonlinear psychoacoustical effects such as the additivity of masking, suppression, and distortion product detection. Since the physiological ear model represents a unified approach covering the most important masking effects of arbitrary sounds, it is suitable for applications such as perceptual audio coding. An improved coding efficiency is expected due to the more accurate masked thresholds generated by the ear model in comparison to results from psychoacoustical models commonly used in audio coding which neglect many of the nonlinear effects of auditory perception. [Work supported by DFG.]

4:40

1pPPa8. Physiological measurements and mathematical models for interaural pathways. Hans-Peter Rangol (Zoologisches Institut, Johann Wolfgang Goethe Univ., Frankfurt am Main, Germany, hprangol@gmx.de)

The middle-ear cavities of birds, reptiles, and anurans are often connected through a tube, which can act as an interaural pathway between both eardrums. Such an arrangement of acoustically effective structures is described as a pressure-difference receiver. It is generally assumed that pressure-difference receivers enhance the skills for directional hearing in animals. Sound pressures and phases were measured with probe-microfones in the interaural pathway from Zebra Finches during azimuthal rotation. The results show pressure differences between right and left middle ear, which are dependent on the azimuth and the frequency of the applied sound. A mathematical model was constructed, which gives the resulting pressure and phase of two overlaying waves that travel different distances. Assuming one wave traveling through the pathway and a second around a sphere, the resulting wave from dependency on azimuth and frequency was calculated. The pattern shows the same characteristic pressure differences as the physiological measurements. The same procedures were performed with a sphere as a hardware model for a head with an interaural pathway. The results show again the expected characteristics. The combination of physiological measurements, mathematical modeling, and testing this model with hardware models gives a meaningful explanation for the acoustical properties of middle ears as a basis for psychological acoustics.

5:00

1pPPa9. A network model for the functional architecture of the mammalian medial superior olivary nucleus. Claus Weiland (Dept. of Zoology, J. W. Goethe Univ., Frankfurt/Main, Germany)

The mammalian MSO is part of the superior olivary complex, the lowest level in the auditory pathway, which receives binaural input. The microarchitecture of the anatomical convergence of both paths is considered as fundamental for the nucleus' function in time-domain processing. This architecture is mostly, according to a proposal of Jefress [L. Jefress, J. Comp. Physiol. Psychol., 35–39 (1948)], interpreted as a network of coincidence detectors. However, the topographic organization of the afferent innervation of the MSO shows much more complexity as is assumed in this model. The connection scheme of the afferent fibers indicates even aspects of terminal divergence, which should result in broadly tuned neurons instead of sharply tuned coincidence detectors. This study proposes a network model for the processing of interaural time delays (ITD) in the MSO based on connection patterns and response properties of the MSO fusiform cells. Instead of local coincidence detection, coding of ITD's is distributed over a cluster of fusiform cells, where each individual cell's degree of stimulus resolution is considered as low. While in the Jefress model each delay line is processed as a single channel, cluster formation based on MSO's multipolar cells could be a mechanism for sampling over these different channels.

5:20

1pPPa10. Modeling the precedence effect: Mechanisms of onset enhancement in binaural lateralization models. Roberto M. Dizon and H. Steven Colburn (Hearing Res. Ctr. and Biomed. Eng. Dept., Boston Univ., 44 Cummington St., Boston, MA 02215, dizon@bu.edu)

The general behaviors of several cross-correlation-based lateralization models in response to dynamic impulsive stimuli, specifically those that elicit the precedence effect, are presented as a means of gaining insight into the contribution of individual stages of the models to precedencelike performance. In general, an enhancement of activity at onset relative to later activity in the model outputs (or equivalently, a temporary post-onset suppression in activity) is associated with the relative perceptual emphasis of early portions of stimuli as described by the precedence effect. Onset enhancement in cross-correlation models has thus far only been exhibited using dynamic mechanisms such as that used in a model by Lindemann [J. Acoust. Soc. Am. (1986)]. This work evaluates the dynamic properties of several cross-correlation models that include mechanisms of onset enhancement. These mechanisms include that of Lindemann's model, peripheral mechanisms such as adaptation, and a physiologically based inhibition mechanism adapted from the IC model of Cai, Carney, and Colburn [J. Acoust. Soc. Am. (1998)]. In this work, binaural models are created from various permutations of the mechanisms described above, and the final and intermediate model outputs in response to a few sample stimuli are shown and compared. [Work supported by ONR and NIH (NIDCD R01 DC00100).]

5:40

1pPPa11. A model of primary sensations, pitch, loudness, and timbre, of sound signals. Y. Ando (Grad. School of Sci. and Technol., Kobe Univ., Rokkodai, Nada, Kobe, 657-8501 Japan, andoy@kobe-u.ac.jp)

Primary sensations, pitch, loudness, and timbre, of a given source signal and sound field are described based on the model of the auditory–brain system [Ando, *Architectural Acoustics* (AIP/Springer-Verlag, New York, 1998)]. The model consists of the autocorrelation and interaural cross-correlation mechanisms. (Note that the power density spectrum is identical with the ACF.) In order to describe timbre, the human cerebral hemisphere specialization for the temporal and spatial factors of sound fields is taken into consideration in a similar manner as the subjective preference. Here, timbre is defined by a remaining primitive sensation that cannot be expressed by only pitch and loudness. [Work was partially supported by the Ministry of Education, Grant-in-Aid for Scientific Research (C), 9838022, 1998.]

6:00

1pPPa12. The mechanisms underlying temporal pitch perception studied with sequences of filtered clicks. Christian Bering and Christian Kaernbach (Inst. für Allgemeine Psych., Univ. Leipzig, Seeburgstr. 14-20, 04 103 Leipzig, Germany)

In order to probe the mechanisms underlying temporal pitch perception it is necessary to eliminate all spectral cues from the stimulus, i.e., to use stimuli that cannot be resolved by the cochlea. High-pass filtered clicks combined with the corresponding low-pass filtered noise prove to be especially suitable for the construction of stimuli with specific temporal properties. These stimuli allow for the testing of specific hypotheses concerning temporal processing in the auditory system. The present study compares psychophysical results, obtained with periodic and aperiodic click sequences with single-cell simulations of chopper cells in the ventral cochlear nucleus. The latter are thought to explain the observed sensitivity to temporal regularity.

Session 1pPPb

Psychological and Physiological Acoustics: Cross-Spectral Processing (Poster Session)

Torben Poulsen, Chair

Department of Acoustic Technology, Technical University of Denmark, Building 352, DK-2800 Lyngby, Denmark

Contributed Papers

All posters will be on display in the Poster Gallery from Monday to Wednesday, 15–17 March. Authors will be at their posters from 2:00 p.m. to 4:00 p.m. on Monday, 15 March.

1pPPb1. Effect of target versus distracter-tone frequency region and variance on sample discrimination of frequency differences. Donna L. Neff, Rebecca L. Wrage, and Walt Jesteadt (Boys Town Natl. Res. Hospital, 555 N. 30th St., Omaha, NE 68131, neff@boystown.org)

This experiment examined how performance in a complex frequency-discrimination task was influenced by the frequency region and relative frequency variability of target versus distracter tones. In a 2IFC sample-discrimination task, normal-hearing listeners were asked to select the interval containing target tones drawn from the higher of two overlapping Gaussian frequency distributions. The target distributions were placed in one of three frequency regions: ''low'' (500 Hz), ''middle'' (1414 Hz), or ''high'' (4000 Hz). For a particular target region (e.g., low), two distracter tones were then drawn at random and added, one from each of the Gaussian distributions at the two remaining frequency regions (e.g., middle and high). All distributions were equivariant on a logarithmic frequency scale. Based on the CoRE model [R. A. Lutfi and K. A. Doherty, J. Acoust. Soc. Am. **96**, 3443–3450 (1994)], it was predicted that tones with greater variance would have a greater influence on performance independent of frequency region. The results indicate that the influence of variance depends on the frequency region of targets versus distractors, with notable individual differences and training effects. [Work supported by NIDCD.]

1pPPb2. Detection of tones of unexpected frequency in amplitude-modulated noise. Matthew B. Fitzgerald and Beverly A. Wright (Audiol. and Hearing Sci. Prog., 2299 N. Campus Dr., Northwestern Univ., Evanston, IL 60208-3550)

The percentage of correct detections of tonal signals presented in wideband sinusoidally amplitude modulated (SAM) noise, or noise with SAM and unmodulated regions, was measured for an expected signal of 1000 Hz and unexpected signals of 200, 600, 1400, and 1800 Hz. The level of each signal was set so the signal could be detected on 85% of trials when presented by itself. When the noise was 100% SAM at 10 Hz, six of nine listeners detected at least one unexpected signal on over 70% of trials. When a 400-Hz wide region centered at 1000 Hz was unmodulated and the remaining wideband noise was SAM, or vice versa, all three listeners still detected at least one unexpected signal on over 70% of trials. With unmodulated noise, all unexpected signals were detected at chance. These results suggest most listeners monitor multiple auditory filters when detecting tonal signals in noise that is entirely or partially SAM. This listening strategy may be related to, but not responsible for, the better detectability of signals in wideband SAM than unmodulated noise, because here signal levels in SAM noise were similarly low for listeners who did or did not detect unexpected signals. [Work supported by NIH/NIDCD.]

1pPPb3. A behavioral reverse correlation technique to decipher early auditory feature coding. Christian Kaernbach (Inst. für Allgemeine Psych., Univ. Leipzig, Seeburgstr. 14-20, 04 103 Leipzig, Germany)

Certainly people do not perceive auditory stimuli as a bunch of time-varying spectral components, but what else could provide a format for early auditory feature coding? This question has been dealt with extensively in vision, revealing sensitivity to waveletlike stimuli and to simple object feature elements. Up to now there has been no attempt to decipher the alphabet of hearing at a coding level higher than that of sinusoidal phonons on the auditory nerve. The present study demonstrates a behavioral reverse correlation technique revealing which kind of features will be detected in random stimuli (noise). When a noise segment is repeatedly presented, it will elicit simple percepts, equivalent to the detection of lines, circles, or faces in random dot images. When these percepts are masked by another noise segment, in about 60% of the cases there is a clear-cut difference of masking level depending on the sign of the masker. By adding up sign-sensitive maskers in the correct polarity, one gets the ''noison'' that elicits this specific percept. The noisons presented in the talk represent only a random selection of possible auditory features, but they give an indication of the potential scope of feature specifications.

1pPPb4. Internal representation of spectral information in dependence of signal type and auditory filter bandwidth. Bernhard Laback and Werner A. Deutsch (Acoust. Res. Lab., Austrian Acad. of Sci., Liebigg. 5, 1010 Vienna, Austria)

Previous studies have shown that enhancement of the spectral background (lower level spectral components) improves intelligibility as well as listening comfort of music signals in subjects with broader than normal auditory filters. This is in contrast to reported results on speech perception in noise, yielding reduced intelligibility as a result of spectral contrast reduction. A proposed model for the internal object representation by separation of ''spectral layers'' accounts for these opposing effects. ''Spectral layers,'' corresponding to different sound sources [in music: dominant/accompanying musical voices; in speech-in-quiet: formant peaks/modulation sidebands of the peaks (transitions); in speech-in-noise: speech foreground/speech background/noise components] are represented in coherent spectral amplitudes. The audibility of ''spectral layers'' is derived from individual masking properties. Pilot experiments with speech tokens and short music fragments agree relatively well with the model predictions: The effect of either increasing or reducing the spectral contrast on discrimination highly depends on the distribution of the perceptually relevant acoustical information in the ''spectral layers'' relative to their audibility. Performance varies for different signal types (simultaneous musical voices, vowel, consonant). [Work supported by the Austrian Academy of Sciences.]

1pPPb5. Amplitude modulation discrimination interference (MDI) and speech intelligibility for normal-hearing and hearing-impaired subjects. Jan Koopman, Rene van der Horst, and Wouter A. Dreschler (Dept. of Exp. Audiol., Academic Medical Ctr., P.O. Box 22660, NL-1100 DD, Amsterdam, The Netherlands, j.koopman@amc.uva.nl)

Modulation detection interference is the decrease of sensitivity for amplitude modulation when the target is presented in the presence of modulating maskers [W. A. Yost and S. J. Sheft, J. Acoust. Soc. Am. **85**(2), 848–857 (1989)]. A typical MDI experiment was carried out in which the maskers were located at 500 and 4000 Hz and the target at 1400 Hz (covering the range of speech frequencies). The modulation depths of the maskers were held steady at 0, 0.18, or 0.30. The modulation rates were 4, 8, or 16 Hz. Also, speech intelligibility was measured by a sentence test in continuous noise. Both experiments were carried out for six normal-hearing and six hearing-impaired subjects. All signals were presented at MCL level. Both groups of subjects were most sensitive for amplitude modulations when they were presented in the presence of the nonmodulating maskers or without any masker at all; the sensitivity decreased by 8 dB for normal-hearing subjects and 6 dB for hearing-impaired subjects for the modulated masker condition. The modulation threshold also increased for increasing reference modulation depths. The amplitude modulation discrimination sensitivity showed a significant correlation with the critical signal-to-noise ratios for speech intelligibility.

1pPPb6. Comodulation masking release in bit-rate reduction systems. Martin D. Vestergaard,[a)] Karsten B. Rasmussen, and Torben Poulsen (Dept. of Acoust. Technol., Bldg. 352, Tech. Univ. of Denmark, DK-2800 Lyngby, Denmark, kbr@dat.dtu.dk)

It has been suggested that the level dependence of the upper masking slope be utilized in perceptual models in bit-rate reduction systems. However, comodulation masking release (CMR) phenomena lead to a reduction of the masking effect when a masker and a probe signal are amplitude modulated with the same frequency. In bit-rate reduction systems the masker would be the audio signal and the probe signal would represent the quantization noise. Masking curves have been determined for sinusoids and 1-Bark-wide noise maskers in order to investigate the risk of CMR, when quantizing depths are fixed in accordance with psycho-acoustical principles. Masker frequencies of 500 Hz, 1 kHz, and 2 kHz have been investigated, and the masking of pure tone probes has been determined in the first four 1/3 octaves above the masker. Modulation frequencies between 6 and 20 Hz were used with a modulation depth of 0.75. A CMR of up to 10 dB was obtained at a distance of 6 Bark above the masker. The amount of CMR was found to depend on the presentation level of the masker; a higher masker level leads to a higher CMR effect. Hence, the risk of CMR affecting the subjective performance of bit-rate reduction systems cannot be ruled out. [a)] Currently at Oticon Research Centre Eriksholm, 243 Kongevejen, DK-3070 Snekkersten, Denmark, mve@oticon.dk

1pPPb7. Comodulation masking release (CMR) for multiple temporal envelope maskers. Lee Mendoza (Dept. of Commun. Sci. and Disord., Louisiana State Univ., 163 M&DA, Baton Rouge, LA 70803, lmendoz@lsu.edu)

Thresholds for a pure-tone signal were measured in masking noise composed of multiple noise bands. In some conditions, one group of flanking bands (consisting of half of the flanking bands) was comodulated, and independent of the other group of comodulated flanking bands in

terms of temporal envelope. The center frequencies of the flanking bands were arranged such that noise bands from the two groups were spectrally interleaved. In these conditions, the band of noise centered on the signal was a combination of two noise bands, which were each comodulated with one group of flanking bands. CMR was obtained but was not of the magnitude observed when all noise bands shared a common temporal envelope. Potential cues in the multiple envelope maskers included the information in the summed representation of the flanking bands, and the degree of temporal envelope correlation between the on-signal band and the flankers. Additional stimulus conditions examined the viability of these cues. The results of these experiments support an envelope correlation explanation for the CMR observed with the multiple temporal envelope maskers. [Work supported by the Louisiana Board of Regents.]

1pPPb8. An exploration of coherence masking protection in a spectral profile discrimination task. Peter J. Bailey (Dept. of Psych., Univ. of York, York YO10 5DD, UK, pjb1@york.ac.uk)

Gordon [P. C. Gordon, Percept. Psychophys. **59**, 232–242 (1997)] has reported a form of masking protection, in which identification thresholds for noise-masked target signals are lower when a nondistinctive cosignal is present. Gordon suggested that target signals are protected from masking as a result of their perceptual coherence with the cosignal. The experiment reported here sought to show whether cosignals enhance performance in a task involving discrimination of spectral profiles. The target signals were increments in the level of components in a complex tone (harmonics $1-7$, $F0=125$ Hz), with the third harmonic incremented in one trial interval and the fifth harmonic in the other. Discrimination threshold increments were measured for four well-practiced listeners in three conditions. In one condition, the complex tone was presented alone, without a cosignal. In the two cosignal conditions, harmonics 8–40 were added, with either a vowellike or a flat spectral profile. Average threshold increments were similar in the three conditions, and no listener showed reliably smaller threshold increments when a cosignal was present. Cosignals may confer protection from the effects of noise masking, but they do not necessarily enhance performance in other auditory tasks. [Work supported by UK MRC.]

1pPPb9. Auditory profile analysis: Effects of asynchrony and ear of presentation. Nicholas I. Hill and Peter J. Bailey (Dept. of Psych., Univ. of York, York YO10 5DD, UK, nih1@york.ac.uk)

The ability of listeners to detect an increment in the level of a 1-kHz tone relative to four flanking tones (having frequencies of 0.16, 0.40, 2.50, and 6.25 kHz) was examined under four conditions of presentation. In two conditions all five tones were presented monaurally to the listener's right ear, while in the other two conditions the 1-kHz tone was presented to the left ear with the flanking tones presented to the right. For each of the two spatial configurations, the tones were either gated on and off at the same time or else the 1-kHz tone was gated on 100 ms before, and off 100 ms after the flanking tones. The flanking tones were always gated on and off simultaneously and had a duration of 200 ms. The overall level of the stimuli was randomized on each presentation. Average thresholds in the two asynchronous conditions were approximately 10-dB higher (signal re: pedestal amplitude) than those in the synchronous, monotic condition. Ear differences alone resulted in an average elevation in threshold of 3 dB. These results suggest that asynchrony is more effective than spatial cues in preventing intensity information from being combined across frequency. [Work supported by UK MRC.]

Session 1pSAa

Structural Acoustics and Vibration: Inverse Problems in Structural Acoustics I

Sean F. Wu, Cochair

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Chair's Introduction—1:55

Invited Papers

2:00

1pSAa1. An overview of the inverse problem for sound reconstruction in interior spaces. Earl G. Williams (Naval Res. Lab., Washington, DC 20375)

Calculation of the sound field between noise sources and a closed two-dimensional measurement surface is an inverse propagation problem. A powerful and popular method for the reconstruction of sound fields in this space is based on boundary element methods applied to the Helmholtz integral equation. Unfortunately, due to the numerical nature of these solutions, the physics associated with backtracking (backpropagating) the field from the measurement surface to the source surfaces is hidden. Analytic models are used to view this backpropagation as a deconvolution operation on the measured field. The nature of this deconvolving function (called the inverse propagator) is examined; it is a singular function, it is spatially local, and it *unsoothes* the measured pressure field. An understanding of the nature of the inverse propagator is critical to successful solution of the inverse problem by BEM methods (especially the regularization process), as well as the sucessful development of three-dimensional sound projection methods for sound synthesis. Furthermore, this understanding leads to new insights into the solution of the inverse problem. [Work supported by ONR and NASA.]

2:20

1pSAa2. Problem of reconstructing the full vibration field from limited data. Yuri I. Bobrovnitskii (Lab. of Structural Acoust., Mech. Eng. Res. Inst., M. Kharitonievsky 4, Moscow 101830, Russia, bobrovni@orc.ru)

The problem of expanding the vibration field from the measured points to unmeasured (inaccessible) points is posed and solved. This field reconstruction problem is one of the inversion problems of structural dynamics and acoustics. The procedure of handling it is the following: the full vibration field under study is described by a mathematical model containing a certain number of parameters (e.g., a modal model with the mode amplitudes as the model parameters); the values of these parameters are identified from the limited data; the field of the structure is then computed using this model. The main result of the present submission is that there exists the best mathematical model (containing a finite number of parameters) which gives the minimal error of describing the field in unmeasured points. Another result presented is that there also exists the optimal volume of the measurement data. The paper contains a general theory, equations relating the error to the measurement accuracy and the data volume, and some computer simulation results. Discussion of how to choose the best field mode using only the measurement data is also presented.

2:40

1pSAa3. An overview of reconstructing acoustic pressure fields using the HELS method. Sean F. Wu (Dept. of Mech. Eng., Wayne State Univ., 5050 Anthony Wayne Dr., Detroit, MI 48202)

The acoustic pressures radiated from complex vibrating structures are reconstructed by using the Helmholtz equation least-squares (HELS) method [Wang and Wu, J. Acoust. Soc. Am. **102**, 2020–2032 (1997); Wu and Yu, *ibid.* **104**, 2054–2060 (1998)]. Specific examples include an engine block and the interior space of a passenger vehicle. These structures are of arbitrary shapes and geometry, containing sharp edges and corners, and abrupt changes in surface contour. To test the robustness of the HELS method, measurements of field acoustic pressures are taken over a planar surface at a certain distance away from the structure. The reconstructed acoustic pressures, however, extend over the entire (nonplanar) surface of the structure. Note that the input data in these cases are not error free due either to measurement uncertainties or to the loss of the near-field effect. On the other hand, reconstructed acoustic pressures consist of predominantly the near-field effect. Hence the problem becomes mathematically ill-posed. To overcome this ill-posedness difficulty, an optimization scheme is developed which enables one to obtain satisfactory reconstruction results with a relatively few number of measurements in the field. The HELS method is shown to be effective in the low- to mid-frequency range, and can become a robust noise diagnostic tool for analyzing structure-borne sounds. [Work supported by NSF.]

3:00

1pSAa4. Some inverse problems in structural acoustics. D. N. Ghosh Roy (SFA, Inc., 1401 McCormick Dr., Landover, MD 20785), J. A. Warner, and L. S. Couchman (Naval Res. Lab., Washington, DC 20375-5350)

Fundamental problems of direct and inverse solutions of the multidimensional Helmholtz equation in structural acoustics are discussed. Inversion of an obstacle from the far-field pattern is compared to that of a medium and/or an inhomogeneity. An overview of various methods of solving these problems is presented. The linear techniques for a medium and/or inhomogeneity problem and some intrinsically nonlinear methods of inversion for an obstacle are described. Four nonlinear methods are compared, namely, the ansatz of potential over an internal curve, the dual space method using the Herglotz wave functions, methods based on the Rayleigh hypothesis, and the most recent method of shape differentiation combined with the Padé approximation. The efficacy and advantages of the last technique are illustrated by numerically reconstructing both penetrable and impenetrable obstacles of various shapes and orientations. The robustness of the method in regard to the initial conditions and regularization are demonstrated and its extension to elasticity and an acoustical shell structure are presented. Finally, some open questions and future directions are discussed.

3:20

1pSAa5. Identification of forces by the RIFF method. C. Pezerat and J. L. Guyader (Lab. Vibs. Acoust. de l'INSA de Lyon, 20 avenue A. Einstein, 69621 Villeurbanne Cedex, France)

The goal of this presentation is to show the application of a test bed which should permit the identification of forces created by an engine when it is not possible to measure them directly. The interest of this experimentation is to have an idea of the forces in the real situation. The mechanism consists of fixing the engine on a known plate which has characteristics related to those of the real situation. To identify forces exciting the plate, the method used, the RIFF method, is based on the computation of the force distribution, i.e., the second member of the equation of motion of the plate, from the knowledge of the flexural displacements. The main difficulty of this inverse problem comes from uncertainties in data that produce a high noise level in the result. To overcome this instability, two regularization techniques are presented. The first approach is based on singular value decomposition (SVD), consisting of eliminating the singular values of the operator applied on displacements. The second approach is based on high wave number filtering. Formulations of techniques are presented and experimental results are shown in order to validate both approaches.

3:40

1pSAa6. Source localization on a "fuzzy structure" beam using matched-field processing. David Feit and Matthew Craun (Carderock Div. Naval Surface Warfare Ctr., 9500 MacArthur Blvd., West Bethesda, MD 20817-5700)

Matched-field processing (MFP), a technique originally developed by the underwater acoustic community [Baggeroer, Kuperman, and Mikhalevsky, IEEE J. Ocean Eng. **18**, 401–424 (1993)] for source localization, is used to localize the source of excitation on a flexurally vibrating beam to which is attached a "fuzzy structure (FS)." The FS, for this study, consists of a set of flexurally vibrating strips, of randomly varying lengths, attached at uniformly spaced points along the length of the beam. MFP differs from classical inverse methods in that it does not require an inversion of the system model. Instead it correlates or "matches" observed signals with solutions of the equations of motion for a range of excitation points using a variety of linear and nonlinear methods. In this paper several different MFP algorithms are used and compared where the observed or simulated data is measured both globally, along the entire length of the beam, or locally over a limited region of the structure. Successful localization is found to be dependent on frequency. For lightly damped structures, the MFP localization results become ambiguous at the resonance frequencies of the structure. [Work supported by ILIR Program CDNSWC.]

4:00

1pSAa7. Inverse scattering of acoustic waves by two elastic shells in water. G. C. Gaunaurd (Naval Surface Warfare Ctr., Carderock Div., Code 684, West Bethesda, MD 20817-5700) and H. Huang (Consultant, Bowie, MD 20715)

The key problem of inverse scattering consists of the identification of an unknown target from its scattering pattern (form-function). For a single, simple-shaped target in an infinite medium, the problem has been solved elsewhere. There is present interest in the case of two elastic shells, with centers a distance d apart, insonified by a (distant) sonar, operating at frequency f, in a boundless sea. The determination of the scattering pattern of such a double-shell target was analyzed by this group earlier [viz., J. Acoust. Soc. Am. **98**, 2149–2156 (1995)]. That direct solution is now built upon to solve the associated inverse problem. From the form-function of the two shells—assumed equal—which is also assumed given in two arbitrary directions, it is determined: (a) that there are actually two interacting shells (i.e., spatial resolution), (b) their separation d, (c) the pair orientation angle α, (d) the shape and size of the shells, (e) their thickness, h, and (f) their material composition. This information is all extracted from features in the form-functions, using the detailed procedure explained here, which accounts for all orders of multiple scattering, since the shells could be close to each other, and thus be strongly interacting. A way to obtain/measure the two needed form-functions is using ultra-wideband (UWB) interrogating waveforms or UWB processing. [Work partially supported by the ILIR Program of NSWC-CD.]

4:20–4:40 Break

4:40

1pSAa8. An inversion approach based on multiple-aspect resonance analysis of finite cylindrical shells in water. Alessandra Tesei, Warren L. J. Fox, and Alain Maguer (SACLANT Undersea Res. Ctr., La Spezia, Italy)

Inverse scattering of fluid-loaded, elastic, thin-walled cylindrical shells is addressed by multiple-aspect resonance analysis. A set of aspect-dependent acoustic phenomena is selected from membrane wave and resonance scattering theories, which are expected to be backscattered by fluid-filled or empty thin-walled shells and to give rise to resonance phenomena in the ka range (1,50). The features selected are from spatial axial modes, and Lamb-type, Scholte–Stoneley, and shear helical waves. The last three wave families are significant only over a small range of target aspects near broadside, with the width of this range depending on shell properties. Approximate equations are formulated relating resonance behavior in the aspect-frequency domain, to target parameters such as shell outer radius, thickness, length, and material. The inverse methodology is validated on experimental data from a steel cylindrical shell with flat end-caps, filled with air or water and suspended in the water column. The target was continuously rotated on the plane of its longitudinal axis while insonified by broadband pulses. Good agreement between theory and experiment encourages the extension of the approach to more complex scatterers for classification purposes. [Work partially supported by EC (MAST DEO project).]

5:00

1pSAa9. High spatially sampled interior acoustic measurements of a turboprop aircraft for near-field acoustic holography (NAH) algorithm development. Peter C. Herdic,[a] Brian H. Houston (Naval Res. Lab., Code 7136, Washington, DC 20375), and Earl G. Williams (Naval Res. Lab., Washington, DC 20375)

Experimental results are reported of in-flight acoustic responses acquired by scanning a dense array of microphones within an aircraft interior. High data quality was obtained in several measurement sets. This database will ultimately be used for inverse and forward projections of the measured, enclosed acoustic surface to determine the velocity of the fuselage skin and the entire interior acoustic field. A discussion of the experimental design methodology will include approaches to measure the in-flight pressure field coherently during the execution of a scan, spatial sampling requirements based on measured vibration over an arc of the fuselage wall, and the effect of ambient noise on the measurement. The in-flight acoustic response shows spatial variation throughout the interior with dominant blade passage harmonics. Two ground-based (engine off) measurement scans were performed where excitation by an internal acoustic source and a point force applied to a frame assist in identification of interior acoustic modes and structural resonances. [Work supported by NASA, Langley.] [a] Also SFA, Inc.

5:20

1pSAa10. Force spectra identification by FRF matrix inversion: A sensor placement criterion. Matthias Blau (TU Dresden, Inst. fuer Technische Akustik, D-01062 Dresden, Germany)

Excitation force spectra are a key quantity in structure-borne sound characterization. In this paper, the identification of multiple broadband force spectra by FRF matrix inversion is considered. Because of the notorious sensitivity of inverse methods to small errors in measured data, such measurements must be carried out with extreme care. One major decision that could heavily affect the accuracy of the final results concerns the placement of the vibration response sensors. To date, this is often accomplished by just putting them as close as possible to the force input points, or by subjective judgment, or by considering the condition number of the FRF matrix as a criterion. In this paper, an alternative criterion is proposed which takes into account that, especially in the frequency region up to a few hundred Hz, the resulting errors are often dominated by measurement noise-induced bias on response spectra estimates.

5:40

1pSAa11. Experimental reconstruction of the vibration field in a cylindrical shell. Alexander A. Kochkin (Lab. of Structural Acoust., Mech. Eng. Res. Inst., M. Kharitonievsky 4, Moscow 101830, Russia, kochkin@hotmail.com)

The problem of reconstruction of a vibration field in an unmeasured region of a thin cylindrical shell is considered. The forced vibrating shell of finite dimension is divided into two parts: the first part (the measurement region) is accessible for measuring any vibration characteristics while the second part (the reconstruction region) is considered as ''inaccessible.'' The problem is solved using the general approach–see the presentation by Yu. I. Bobrovnitskii at this Special Session. The vibration field of the shell was modeled by a finite sum of normal waves, amplitudes of which were determined by equating the modeling values to the data. The SVD technique was used in the inverse procedure for parameter identification. Most attention was paid to finding an appropriate method for choosing the best model of the shell which minimized the reconstruction error. For the measurement error 0.08 of the experiment conducted, the reconstruction error varied from 0.2 to 0.7 as the measurement region diminished from 0.7 to 0.3.

Session 1pSAb
International Workshop on Active Noise and Vibration Control

Structural Acoustics and Vibration: Signal Processing and Controllers for Active Control

Richard J. Silcox, Cochair
Structural Acoustics Branch, NASA Langley Research Center, Hampton, Virginia 23681-2199, USA

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Invited Papers

2:00

1pSAb1. Iterative feedback control design for active vibration control.　Sandor M. Veres　(School of Electron. and Elec. Eng., Univ. of Birmingham, Edgbaston B15 2TT, UK, s.m.veres@bham.ac.uk)

Model-based robust feedback control design is widespread: a parametric model is identified with some measure of the unmodeled dynamics present in the most advanced approaches. In off-line design, often open-loop experiments are used which can fail to provide satisfactory models because the identification criterion does not serve the aims of feedback robustness. Another frequent problem is possible large bias of the parametric part because of wrong model orders. In the best case large bias comes with large bounds of the unmodeled dynamics in the H-infinity or the L-1 norms. The conclusion is that the use of a plant model is the source of considerable problems while it has so far seemed a natural approach to controller design. In this paper a new ''model-free'' approach will be outlined for designing feedback vibration controllers. This will be based on a direct search using a sequence of iterative experiments. The experiments will alternate between two types: with and without the vibration source being active. The iterations of the controller parameters allowed by the sequence of experiments will lead to a sequence of improvements of control performance. This will be verified experimentally with reduction of sound-induced low-frequency vibration of a glass plate.

2:20

1pSAb2. Predictive feedback and feedforward control for systems with unknown disturbances.　Jer-Nan Juang　(Structural Dynam. Branch, NASA Langley Res. Ctr., Hampton, VA 23681) and Kenneth Eure　(NASA Langley Res. Ctr., Hampton, VA 23681)

This paper presents the theory and implementation of a hybrid controller for general linear systems by incorporating a feedforward path in the feedback control. The generalized predictive control is extended to include a feedfoward path in the multi-input multi-output cases. There are cases in acoustic-induced vibration where the disturbance signal is not available to be used by the hybrid controller, but a disturbance model is available. In this case the disturbance model may be used in the feedback controller to enhance performance. In practice, however, neither the disturbance signal nor the disturbance model is available. This paper presents the theory of identifying and incorporating the noise model into the feedback controller. Implementations are performed on a test plant and regulation improvements over the case where no noise model is used are demonstrated.

2:40

1pSAb3. Constrained beamformer optimization for antenna arrays.　Marcus Bronzel　(Dept. of Elec. Eng., TU Dresden, Helmholtzstr. 18, 01062 Dresden, Germany, bronzel@ifn.et.tu-dresden.de) and Holger Boche　(Heinrich Hertz Inst. fuer Nachrichtentechnik, 10587 Berlin, Germany)

A common problem for hands-free voice terminals and for smart antennas in future mobile communication systems is to optimize the directional response of an array of microphones or antennas in order to enhance the signal-to-noise-and-interference ratio (SINR). Digital beamforming (DBF) algorithms are commonly used for spatial filtering, which effectively results in more benign channel characteristics with reduced interference and delay spread. A new constrained beamforming (CBF) algorithm is presented which allows optimum beam steering and interference cancellation for arbitrarily spaced directions for signals-of-interest (SOI) as well as for interfering waveforms. The proposed CBF combines the properties of the Bartlett and the Capon beamformer [L. C. Godara, ''Application of antenna arrays to mobile communications, Part II: Beam-forming and direction of arrival considerations, Proc. of the IEEE **85**, 1193–1245 (1997)] while maintaining minimum power requirements. It is shown that the result can be interpreted as the beam vector representation using special interpolating fundamental functions. [Work supported by BMBF.]

1pSAb4. Real-time identification of multiple modes using IIR filters. Tae W. Lim (Dept. of Aerosp. Eng., The Univ. of Kansas, 2004 Learned Hall, Lawrence, KS 66045, tlim@aerospace.ae.ukans.edu)

Infinite impulse response (IIR) filter design is investigated in this work for the real-time identification of modal parameters such as natural frequencies, damping ratios, and mode shapes of vibrating structures or acoustic medium. This investigation is mainly aimed at identifying multiple modes simultaneously using a new structure of IIR filters. The previous algorithm [Lim, Cabell, and Silcox, J. Vib. Acoust. **118**, 649–656 (1996)] was limited to the identification of one or two modes due to its complexity of converting identified IIR filter coefficients to modal parameters. The new IIR filter structures are designed to allow direct conversion of filter coefficients to modal parameters and to facilitate the identification of multiple modes. The new approach provides promise for a realistic application of the algorithm for narrow-band signals, which typically contain multiple frequencies. The new IIR filter structure is described in detail in this paper. Results of the multiple mode identification are presented using the new approach. [This research was supported by Structural Acoustics Branch, NASA Langley Research Center.]

1pSAb5. Predictive control for suppression of vibration induced by flow noise. Minh Q. Phan (Dept. of Mech. and Aerosp. Eng., Princeton Univ., Princeton, NJ 08544, mqphan@princeton.edu), Kenneth Eure, and Jer-Nan Juang (NASA Langley Res. Ctr., Hampton, VA 23681)

Predictive control is based on an intuitively appealing concept where the current control action is based on a prediction of the system controlled output at some time step in the future. Originated from chemical process engineering, several predictive control methods have emerged, including model algorithmic control (MAC), dynamic matrix control (DMC), extended prediction self-adaptive control (EPSAC), extended horizon adaptive control, multistep multivariable adaptive regulator (MUSMAR), and generalized predictive control (GPC). Sharing a similar philosophy, the details of these controllers are different from each other due to different choices of cost functions, constraints, and dynamic models. Almost all of them are model-based methods. A data-based treatment of predictive control will be presented, and it will be shown how it is an effective tool to suppress flow-induced vibrations. In this approach, the predictive controller gains are synthesized directly from input–output data instead of working through an intermediate identification model. Experimental results will be used to illustrate the benefit of this data-based treatment of predictive control over conventional model-based approach. A technique will be developed to avoid drifting of the controller gains identified with closed-loop data, and comparison with other conventional methods such as conditional updating, dither, and leakage will be made.

3:40–4:00   Break

Contributed Papers

1pSAb6. Multichannel signal decorrelation using a new generalization of the prediction error filter. Jonathan G. Cook and Steve J. Elliott (Inst. of Sound and Vib. Control, Signal Processing and Control Group, Highfield, Southampton SO17 1BJ, UK, jgc@isvr.soton.ac.uk)

In many multichannel adaptive active control systems the convergence speed is limited by the correlations between the reference signal channels. This paper describes a causal digital filter capable of reducing multiple channel correlated colored input data to an underlying set of uncorrelated unit variance white signals. This filter is the inverse of the spectral factorization matrix for this data set. The problem is approached in two steps. In the first step it is shown analytically that a single-point delay prediction error filter (PEF) will produce white output error data from multiple nonwhite inputs. The flaw in this process is that there is an inherent matrix of gains which scales and correlates the output channels. In the second step, a technique is presented to derive this matrix of gains from the spectral density matrix of the filter output and thus calculate the appropriate inverse filter. The output from this two-step process is uncorrelated, unit variance and white. The complete signal reduction algorithm is then demonstrated for multiple-channel simulated signals and is finally calculated and tested on correlated road noise data.

1pSAb7. Nonlinear adaptive controller for electrodynamic transducers without additional sensor. Stefan Irrgang and Wolfgang Klippel (Klippel GmbH, Aussiger Str. 3, 01277 Dresden, Germany, klippel-gmbh@t-online.de)

Nonlinearities inherent in common transducers (woofers, shakers) produce new spectral components (distortion) in the reproduced sound which affect the perceived sound quality and impair active noise and vibration control. Recent research has shown that these nonlinear mechanisms are predictable and a nonlinear control can compensate for these distortions by inverse preprocessing of the electric input signal. Adaptive schemes have been developed to adjust the control parameters to the particular transducer and to cope with parameter uncertainties due to heating and aging. The adaptive controller requires an acoustic or mechanic output signal derived from the transducer. Using an additional sensor increases the costs and is impractical under harsh environment. However, a motional signal can also be derived from the back-induced EMF by monitoring the input current of the transducer only. Following this approach a nonlinear adaptive observer is implemented in a digital control system providing a robust and cost-effective solution for audio applications and active noise control.

1pSAb8. Fast convergence algorithms for active noise control in vehicles. Luis Vicente and Enrique Masgrau (Dept. of Electron. Eng. and Commun., Univ. of Zaragoza, Zaragoza E50015, Spain, lvicente@posta.unizar.es)

When the reference signal for the FXLMS algorithm is taken from an acoustic sensor, convergence can be very slow due to great eigenvalue spread. Using a nonacoustic sensor, such as a tachometer, cancellation of narrow-band noise in the sensed fundamental frequency and harmonically related ones can be achieved very fast, although other periodic noises and the underlying broadband noise will remain. Backward prediction errors resulting at the various stages of an adaptive lattice predictor (ALP) represent a time-domain orthogonalization of the input signal. An ALP structure, with the acoustic reference as input signal, before a FXLMS, makes up the FXGAL algorithm. Due to orthogonalization, FXGAL is significantly faster compared to FXLMS with reference from a microphone. When compared to FXLMS with a tachometer signal, it is not faster but it can cancel every periodic noise, independently of the harmonical relation between them, as well as the underlying broadband noise. Depending on the relative weights of the different components (periodic harmonically

related, periodic nonharmonically related, and broadband noise) in the reference signal, the FXGAL algorithm can be an excellent alternative to FXLMS. Comparative results between FXLMS (with acoustic and nonacoustic reference) and FXGAL will be presented. [Work supported by CONSI+D-DGA.]

5:00

1pSAb9. Application of nonlinear adaptive signal processing techniques to blind source separation and interference suppression. H. Leon Sibul, J. Micheal Roan, and A. Gregory Babich (Appl. Res. Lab., Penn State Univ., P.O. Box 30, State College, PA 16804-0030)

In active noise and vibration control it is important to identify independent noise generation mechanism and sources. In all large physical systems only a mixture of noise sources can be observed. The mixture model can be either linear or nonlinear. Recently there has been an explosion of significant research literature on blind deconvolution. In blind deconvolution, the actual mixture model and source signals are not known; however, in the theoretical development it is assumed that sources are statistically independent. In practice, good results have been obtained even if sources are not independent. If the mixture model is linear, statistically independent noise sources can be separated by blind deconvolution techniques that are based on the theory of the independent component analysis (ICA). We implement ICA by a cascade of the linear adaptive algorithm that is based on singular value decomposition (SVD) and a nonlinear algorithm that is based on a radial-basis function (RBF) neural network. A RBF neural network can also approximate the inverse of the nonlinear mixture model. We use a fast RBF updating algorithm for adaptation of the nonlinear network. Theoretical results are illustrated by computer simulations. Results are compared with linear adaptive signal processing algorithms. [This research has been supported by ONR, Dr. Kam Ng, Program Officer.]

5:20

1pSAb10. Digital filters used in the feedback loop of an ANR earplug. Veronique Zimpfer, Karl Buck (French German Res. Inst., BP 34, F-68301, Saint Louis Cedex, France), and Nicole Gache (CPE Lyon, LASSSO, Villeurbanne Cedex, France)

Commercially available active hearing protectors are usually implemented as ear muffs with an analog driven feedback loop. The active attenuation bandwidth of these devices is limited to about 800 Hz. Moreover, it is not possible to adapt the analog system for specific noise events or to the user's morphology. Active earplugs allow one to extend the bandwidth of the active protection to frequencies higher than 2 kHz. A digital feedback system has been implemented for active earplugs, in order to allow an adaptation to different noises and users. As the acoustic delays are too short to obtain a causal system when using FIR filters, the filters have been implemented as IIR type filters. The paper presents numerical simulations of different possible algorithms, used for the cascaded IIR filter of the ANR system. It shows how numerical errors are propagated in cascaded Bi-Quad implementations, and how an optimum signal-to-noise

ratio may be obtained. The numerical predictions and experimental results will be presented and compared. As these filters will be used in a closed-loop feedback system, a numerical simulation of the behavior of the closed-loop system, including the electroacoustic transfer function of the ear plug, will be presented.

5:40

1pSAb11. A control friendly software to convert a low-cost DSP board into a powerful active noise controller. Antonio Minguez (Instituto de Investigación del Automovil INSIA, Universidad Politécnica de Madrid, Ctra. Valencia Km. 7, 28031 Madrid, Spain, aminguez@diac.upm.es) and Manuel Recuero (Universidad Politécnica de Madrid, 28031 Madrid, Spain)

A windows environment software has been developed in order to convert the low cost EZ-KIT Lite DSP board of Analog Devices into a powerful monochannel active noise controller. The software, developed with LabVIEW, downloads the code to the EZ-KIT Lite and permits one to modify in real time any working parameter until obtaining the optimal control performance. Among the parameters to control are the gain inputs, outputs, low-pass filters, and dc removing. There are three types of adaptive algorithms that can be used: filtered-X/U LMS, multiple adaptive notch, and a genetic algorithm. All these algorithms are implemented in such a way that it is not necessary to do any previous cancellation path estimation. Also implemented is internal generator that produces up to four simultaneous tones added with a white noise. As it is said before, any parameter of the algorithms and of the generator can also be easily changed in real time, when the cancellation is in progress. This permits one to obtain the best parameters before implementing an active noise controller in any application.

6:00

1pSAb12. Adaptive compensation of noise in a sonar system. Kai Wicker and Egidius Arens (STN Atlas Elektronik GmbH, Sebaldsbrücker Heerstr. 235, D-28305 Bremen, Germany)

The performance of a sonar system on a ship is strongly influenced or even limited by its own and other ship's noise. Its own noise can have different sound sources. The engine causes disturbing line spectra, the auxiliary equipment can radiate both harmonic signals and white or colored noise signals. Due to the viscous flow during ship movement, disturbing noise can also arise. Interaction between the hydrodynamic flow and the whole ship structure can excite the ship's own modes, which results in strong natural oscillations (flow noise-induced resonances). Additionally, other loud ships can disturb the sonar processing, too. An adaptive noise reduction method is presented. It is suited to reduce a ship's own machinery noise, and, with slight variations, also jamming targets. The output of the adaptive method can be calculated in the time domain as well as in the frequency domain. Working in the time-domain, techniques are available which reduce the numerical expense, but some reasons suggest the calculation in the frequency domain. Calculations based on both simulated data and real data recorded at sea are presented.

Session 1pSCa

Speech Communication: Speech Quality in Telecommunications

Hans-Wilhelm Gierlich, Cochair

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Mark E. Perkins, Cochair

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Chair's Introduction—1:55

Invited Papers

2:00

1pSCa1. Methods for subjective assessment of service quality: Milestones to customer-centered quality engineering. Harald Klaus (Quality and Acceptability of Tele-Services, Deutsche Telekom Berkom GmbH, Goslarer Ufer 35, 10589 Berlin, Germany)

In a liberalized market, the opinion of the customer about the quality of an entire service is a key issue for success. In the telecommunication market, technology evolves very rapidly to intelligent and integrated applications. Therefore, new views and methods for the assessment of quality aspects have to be developed in time. The first part of the presentation gives an overview of subjective assessment methods for speech as well as for audio and video applications. Furthermore, practical aspects of quality testing are discussed. In the second part, issues and trends for customer-centered quality engineering are outlined.

2:20

1pSCa2. Speech quality and the E-model. Ute Jekosch and Sebastian Möller (Inst. of Commun. Acoust., Ruhr-Universität Bochum, D-44780 Bochum, Germany, jekosch@ika-ruhr-uni-bochum.de)

The E-model is a quality prediction model for network planning purposes. Its use is going to be recommended by the ITU-T. In comparison with both listening-only and conversation tests the model often shows high predictive power. Accordingly, network planners have high confidence in its applicability. However, until now it is not yet discretely understood which aspects of quality are actually covered by E-model predictions. To name only two, uncertainties do exist, e.g., for the mode of communication (listening only or conversation) and user-related factors. In the talk, a schematic is presented which covers, besides speech-communication-related elements (one-way voice transmission quality, communication effectiveness, ease of communication) also human-factor-related and service-related elements (comfort, costs). It is shown that matters are simplified by exclusively using the MOS as the basis for a thorough verification of the model. In the course of verification, measures of user acceptance, expectation, etc., are needed in order to understand how human- and service-related features influence the users' perception of the quality of service. An approach is presented to obtain subjective judgments other than MOS on a scale which is closely related to the scale of the model, namely the transmission rating factor R.

Contributed Papers

2:40

1pSCa3. Hands-free car kits—A performance discussion of "state-of-the-art" solutions. Frank Kettler, Hans Wilhelm Gierlich, and Winfried Krebber (HEAD acoustics GmbH, Ebertstrae 30a, 52134 Herzogenrath, Germany)

The increasing demand for hands-free solutions in vehicles partly results from safety reasons. In several countries, the use of handsets is already prohibited for drivers. From the technical point of view, not only the "hands-free problem"—acoustical stability and echo—has to be solved. The environmental conditions, i.e., noise level and position of microphone and loudspeaker, are extremly critical. Technical implementations include various signal processing methods such as echo cancellers, noise reduction algorithms, and level switching devices. Typically they work in conjunction with codecs (GSM or others). Consequently, speech transmission quality is influenced in various ways. Comparative measurements carried out with eight different hands-free car kits are discussed in this presentation. The test setup guarantees reproducible conditions using the auditory part of a driving simulator, artificial head technology, and a codec simulator. The determination of important objective parameters was carried out with adapted test signals, while real speech samples were used for subjective evaluation. Measurement setup and relevant parameters concerning speech transmission quality are briefly discussed. A number of listening examples demonstrate differences between various implementations.

3:00

1pSCa4. Scenarios for economic conversation tests in telephone speech quality assessment. Stephan Wiegelmann, Sebastian Möller, and Ute Jekosch (Inst. of Commun. Acoust., Ruhr Univ. Bochum, D-44780 Bochum, Germany, moeller@ika.ruhr-uni-bochum.de)

In telephony, there is a permanent need for auditory testing of the effect of different types of impairment on speech communication quality. Though quality perception is different in listening-only and conversation situations, for economic (time and money) reasons, in most cases only listening-only tests are carried out to assess telephone speech quality. For some parameters, however, conversation tests are mandatory, as the parameter under investigation affects only the conversation situation (e.g., echo, delay). In order to overcome the limitations of conversation tests, new scenarios have been developed which allow the testing of around three times as many circuit conditions within one test session compared to conventional conversation test scenarios. The scenarios cover everyday situations of information exchange, and meet most of the basic require-

ments of quality assessment scenarios (naturalness, balance, comparability, clarity). The scenarios are available in three languages and have been successfully used by different telecommunication test laboratories. Comparison with listening-only test results shows that differences exist between the two types of quality judgments, which underlines the necessity for conversation tests in order to get more valid quality judgments. Modifications of the scenarios for mobile and Internet telephony situations are presented.

3:20

1pSCa5. Expectation in quality assessment of Internet telephony. Sebastian Möller and Joachim Riedel (Inst. of Commun. Acoust., Ruhr Univ. Bochum, D-44780 Bochum, Germany, moeller@ika.ruhr-uni-bochum.de)

Telephony from Internet terminals is getting more and more popular. Thus it is interesting to know which level of quality one user expects from a connection established from a computer terminal in relation to a normal wirebound telephone. For mobile situations, it has been shown that the lower expectation results in more favorable quality judgments in comparison with wirebound systems when assessing physically equal connections. The user's expectation includes call motivation, attitude, experience, emotions, etc., and it will have an important influence on his/her perception of and judgment on quality. A conversation test has been carried out in order to investigate the quality impact due to Internet-typical impairments (absolute delay, low-bitrate codecs), both for users of a conventional wirebound telephone and an Internet terminal. In contrast to what can be observed for mobile phones, it turns out that the expectation level of potential users may result in a disadvantage effect, i.e., quality expectation is so high that Internet terminal connections are rated worse than conventional ones for physically equal connections. The situation is different for subjects who never operated a computer terminal before. Results are discussed regarding user groups and the consequences for the assessment of Internet terminal telephony.

3:40

1pSCa6. Speech quality evaluation: Auditory and instrumental procedures. Frank Kettler, Hans Wilhelm Gierlich, and Ekkehard Diedrich (HEAD acoustics GmbH, Ebertstrae 30a, 52134 Herzogenrath, Germany and Deutsche Telekom Berkom GmbH, 10589 Berlin, Germany)

The complexity of signal processing in modern communications requires advanced test procedures. This goes both for telecommunication networks and for terminal equipment. Hands-free telephones (HFT) are a typical example of telecommunication equipment showing a nonlinear, time-variant, and speech-controlled transmission characteristic. In addition, the performance depends heavily on the environmental conditions like test rooms or ambient noise. New test procedures, both auditory and instrumental, have been developed during the last few years to determine quality aspects of HFT's. The principles of auditory tests as conversational tests, specific double-talk tests, and a new realization of listening-only tests are discussed. These procedures are used for both parameter identification and value selection. The parameters found to be auditory relevant have been identified, and advanced tests for quality parameters—determining the auditory quality—were developed.

4:00–4:20 Break

4:20

1pSCa7. Psychoacoustically motivated objective speech quality evaluation procedures, PSQM, and improvements. John G. Bereends and Andries P. Hekstra (KPN Res., P.O. Box 421, NL- 2260 AK Leidschendam, The Netherlands)

PSQM (Perceptual Speech Quality Measure), measuring speech quality objectively, has been standardized by ITU-T as recommendation P.861. PSQM characterizes the perception of the (degraded) output speech signal of the system in comparison to the (ideal) input speech. A perceptual model is used that maps input and output signals onto psychophysical

representations using psychophysical equivalents of frequency (Bark) and intensity (compressed Sone). The quality of the device under test is determined with a simple cognitive mapping from the differences in the psychophysical representation to the perceived speech quality in terms of Mean Opinion Scores (MOS). Within an ITU benchmark testing a limited set of unknown codec distortions the PSQM showed high correlations (around 0.97) between subjectively perceived and objectively measured speech quality. When applying PSQM to a wide variety of real world distortions two major limitations show up: First, dynamic time warping effects, as they will be found, e.g., in Internet telephony, cause a degradation in correlation. Second, the perceptual model that is used within the PSQM method is too simple to account for a wide variety of distortions. This presentation discusses extensions to the PSQM method that allows application to real world distortions.

4:40

1pSCa8. Noise reduction in acoustic signals using the perceptual coding and intelligent decision systems. Rafal Krolikowski and Andrzej Czyzewski (Tech. Univ. of Gdansk, WETI, Sound Eng. Dept., 80-952 Gdansk, Poland)

A new algorithm of broadband, nonstationary noise reduction was engineered employing the perceptual approach to the removal of noise from acoustic signals. It enables analysis and processing of sound according to characteristics of hearing sense, and employs a decision system to automatically adjust thresholds of masking. The presented algorithm provides an extension of perceptual coding applications, because it makes possible the reduction of noise included in source signals. Although noisy components of a signal may occur randomly, it is possible to estimate the noise distribution on the basis of signal analysis in silence passages of the transmission. Subsequently, the decision should be made as to the masking curves' parameter settings which will make the noise inaudible. That is the reason why an intelligent approach is used in modeling some interrelations between available noise patterns and the noise affecting consecutive portions of useful signals. The decision system was implemented in various ways using fuzzy reasoning, rough set approach, and modified feedforward neural networks with functional links. The proposed methods of noise reduction and obtained results of noisy speech processing are presented in the paper. [Work supported by the Committee for Scientific Research, Poland, Grant No. 8 T11D 021 12.]

5:00

1pSCa9. Application of auditory contours to speech quality measurements. Peter Daniel, Manfred Zollner (Neutrik Cortex Instruments, Erzb.-Buchberger-Allee 14, D-93051 Regensburg, Germany, daniel@neutrik-cortex.de), and Wolfgang Ellermeier (Univ. Regensburg, 93040 Regensburg, Germany)

For the development of instrumental speech quality measures it is important to model human auditory signal processing. The ITU-T standard P.861 for telephone transmission quality, however, neither takes into account the time-frequency resolution of the human ear nor its masking properties. Moreover the abstraction or contorization processes occurring in speech perception are disregarded. Terhardt proposed to use spectral pitch as the primary auditory contour. In our approach it is derived from the part-tone time-pattern (PTTP) originally conceptualized by Heinbach [Acustica **67**, 242–256 (1988)]. The PTTP is extracted from an aurally adequate spectral analysis. A new instrumental method based on differences of auditory contours will be presented. The results will be compared with other instrumental methods and with data from subjective listening tests.

1pSCa10. Application of automatic speech recognition to evaluation of speech transmission quality in analog communication systems. Wojciech Majewski and Wojciech Myslecki (Inst. of Telecommunications and Acoust., Wroclaw Univ. of Technol., Wybrzeze Wyspianskiego 27, 50-370 Wroclaw, Poland, majewski@aipsa.ita.pwr.wroc.pl)

The preliminary results of application of automatic recognition of isolated words to objective evaluation of speech transmission quality in analog telephone channels are presented. A memoryless, finite state recognition system with LPC, FFT, and BF-FFT (where the speech signal was filtered in Bark bands) parametrization was applied. In classification stage a dynamic time warping and nearest-neighbor algorithm were utilized. Nonsense word lists consisting of 100 logotoms were recorded in a studio by a professional male speaker and utilized next as a test material. Speech transmission quality was examined in laboratory models of telephone channels with frequency bands of 300–3400, 400–2500, and 100–6000 Hz for speech-to-white-noise ratios in the range of +15 to −15 dB. The results of objective measurements expressed in percent of logotoms correctly recognized by the recognition system were compared under the same transmission conditions with subjectively measured logotom intelligibility. The best agreement between subjective and objective evaluation of speech transmission quality was obtained for automatic speech recognition utilizing BF–FFT parametrization. The results of objective evaluation of speech transmission quality by means of the presented method are encouraging and the experiments will be continued for other communication channels (e.g., digital) and different distortions and disturbances.

5:40–6:00   Discussion

MONDAY AFTERNOON, 15 MARCH 1999

POSTER GALLERY, 2:00 TO 6:00 P.M.

Session 1pSCb

Speech Communication: Data Based Speech Synthesis and Analysis I (Poster Session)

Andy Harsin, Chair
Lucent Technologies/Bell Labs Advanced Technologies, 101 Crawfords Corner Road, Holmdel, New Jersey 07733, USA

Contributed Papers

All posters will be on display in the Poster Gallery from Monday to Wednesday, 15–17 March. Authors will be present at their posters on Monday, 15 March. To allow contributors an opportunity to see other posters, authors of odd-numbered papers will be at their posters from 2:00 p.m. to 4:00 p.m. and authors of even-numbered papers will be at their posters from 4:00 p.m. to 6:00 p.m.

1pSCb1. Automatic syllabic boundary extraction from connected speech. Andy Harsin (Lucent Technologies/Bell Labs Adv. Technologies, 101 Crawfords Corner Rd., 1L-513, Holmdel, NJ 07733)

A computational algorithm is presented which locates syllabic boundaries in connected speech. Filtered, digitized speech is processed in two stages. The first stage is an auditory front end which filters the digitized speech into critical bandwidths. The critical bands are amplitude-scaled according to a standard frequency/amplitude function, then low-frequency amplitude modulations in the 2–30-Hz range are emphasized according to a perceptual modulation sensitivity function. Next, the critical bands are low-pass filtered at 100 Hz and decimated at a rate of 25:1. Then they are low-pass filtered at 100 Hz, again creating acoustic envelopes of the processed speech, one for each critical band. During the second stage of processing, an autocorrelation-based algorithm is run on the envelopes. The local minima of this algorithm are pooled across the critical-band envelopes, yielding syllabic boundaries. The utterances used to develop and test this algorithm were taken from the Harvard Phonetically Balanced Sentences. Currently, the algorithm places boundaries within a few tens of milliseconds of where a phonemic syllabification would place them. Work continues on fine-tuning the algorithm. Another goal is to compare the algorithm's performance against human listeners'. [The author wishes to acknowledge the support of Lucent Technologies in conducting this project.]

1pSCb2. Issues in the design of an advanced unit selection method for natural sounding concatenative synthesis. Andrew Breen and Peter Jackson (BT Labs., Martlesham Heath, Ipswich IP5 3RE, UK)

This paper describes a method for selecting units from a database of recorded speech, for use in a concatenative speech synthesizer. The simplest approach is to store one example of every possible unit. A more powerful method is to have multiple examples of each unit. The challenge for such a method is to provide an efficient means of selecting units from a practical inventory, to give the best approximation to the desired sequence in some clearly specified way. The approach used in BT's Laureate system uses mixed N-phone units. In theory, such units could be of arbitrary size, but in practice they are constrained to a maximum of three phones. This method dynamically generates the unit sequence based on a global cost. Units are selected using purely phonologically motivated criteria, without reference to acoustic features, either desired or available within the inventory. A sophisticated method of unit selection does not, however, guarantee natural sounding synthesis. The process of signal generation plays an equally important part. The style of speech generated by such an approach is highly influenced by the underlying nature of the original recorded speech data. The paper will discuss this and other limitations of purely concatenative synthesis.

1pSCb3. System for wideband speech coding based on recursive filterbanks. Ralf Th. Pietsch and Arild Lacroix (Institut für Angewandte Physik, Johann Wolfgang Goethe Univ., D-60054 Frankfurt am Main, Germany)

Wideband speech coding becomes increasingly important because of the enhanced quality due to the extended frequency range. Possible applications are in various fields partly with fixed and partly with variable data rates: synchronous and asynchronous transmission over digital networks, mobile telephony, and multimedia systems. In this context a system is presented which is well suited for dialog communication. The signal analysis and synthesis is done by filterbanks based on recursive filters. The use of recursive filters implies short system delays in the filterbank. However, phase distortions are present and have to be considered carefully. Different methods were implemented to reduce phase distortions and to compensate group delay peaks. Coding is based on redundance and irrelevance reduction. The irrelevance reduction is implemented by the exploitation of psychoacoustical masking effects. Encoder and decoder are running on a single DSP32C signal processor in real time. The achieved data rate for good speech quality is 32 kbit/s and below.

1pSCb4. Selection of cross-fade regions for concatenation of demisyllable units. Nicholas J. Kibre (Panasonic Technologies, Inc., Speech Technol. Lab.)

This paper describes how overlap regions are identified in demisyllabic filter trajectory concatenation units used in a residually excited formant synthesis system, currently being developed at PTI/STL [Pearson, Kibre, and Niedzielski, ICSLP 1998]. This approach has been shown to produce clear and human-like synthetic speech, but as in other concatenative methods smooth transitions in cross-fade regions are essential to sound quality. This can best be obtained if a nucleus region is identified for each segment type which has consistent filter trajectories in all tokens. Database size precludes manual tuning and labeling, and this paper considers and compares two approaches to automating this task. The first of these is a rule-based approach, in which observation and phonological theory are used to formulate an ideal cross-fade region for each segment. Each token is searched for its best match to this definition, as determined according to penalty weights for different kinds of deviation from it. The second is a statistical approach utilizing HMMs to model the cross-fadable nuclear region of each segment type, as well as the adjacent transition regions. Rather than manually marking these regions in the training data, embedded training can be used to discover optimal definitions for the three phases.

1pSCb5. Blind separation of speech signals in the frequency domain. Jörn Anemüller (AG Medizinische Physik, Carl von Ossietzky Universität Oldenburg, 26111 Oldenburg, Germany)

Blind source separation algorithms try to reconstruct original signals, e.g., multiple speakers, from knowledge of their superpositions, using solely the mutual statistical independence of the source signals as criterion for separation. However, application of existing algorithms to acoustic superpositions is limited by the complex nature of room transfer functions and by the use of nonlinear computations. Expanding on our previous work, we linearize the acoustic source separation problem by moving to the frequency domain [Anemüller and Gramss, DAGA (1998)] and eliminate the need for computation of nonlinear functions by using a multiple decorrelation approach. Thus, our algorithm exploits the highly redundant structure of, e.g., speech signals in order to reduce the computational cost. Results of separation experiments are presented.

1pSCb6. A combined wideband speech and audio coder using human articulatory and auditory models. Guangyu Wang (Univ. of Kiel, Inst. of Network and System Theory, D-24143 Kiel, Germany)

The modern high-quality speech and audio coding algorithms are mostly based on human articulatory models. On the other hand, high-quality audio coders are mostly based on models of human auditory system. However, neither of these two methods can provide acceptable performance for both music and speech. In this paper a coder is proposed which operates for both wideband speech and audio signal. The structure of the coder is based on both human articulatory and auditory models. In the first part the LPC parameters are drawn using the conventional autocorrelation method in order to simulate the human articulatory system. With LPC parameters the redundancy of input signal in time domain is removed. In the second part of the coder, the LPC residual signal is further transformed into frequency domain through MLT (modulated lapped transform). The MLT coefficients are then quantized in frequency domain using the human auditory properties such as masking property. The informal listening test has shown that the proposed coder provides very good quality for wideband speech and for most music signals.

1pSCb7. Evaluation of monaural and binaural speech enhancement for robust auditory-based automatic speech recognition. Michael Kleinschmidt, Thomas Wittkop, and Birger Kollmeier (AG Medizinische Physik, Universität Oldenburg, D-26111 Oldenburg, Germany)

A major deficiency in state-of-the-art automatic speech recognition systems is the lack of robustness in additive and convolutive noise. The model of auditory perception, as developed by Dau *et al.* [J. Acoust. Soc. Am. **99**, 3615–3622 (1996)] for psychoacoustical purposes, partly overcomes these difficulties when used as a front-end for speech recognition. Especially in combination with locally-recurrent neural networks (LRNN) the model output, called ''internal representation'' had been shown to provide highly robust feature vectors [Tchorz and Kollmeier, J. Acoust. Soc. Am. (submitted)]. To further improve the performance of this auditory-based LRNN recognition system in background noise, different speech enhancement methods were examined. The minimum mean-square error (MMSE) short-term spectral amplitude estimator (STSA), as proposed by Ephraim and Malah [IEEE Trans. Acoust., Speech, Signal Process. **32**, 1109–1121 (1984)], was compared to a binaural Wiener filter [Wittkop *et al.*, this meeting], based on directional and coherence cues. Both noise reduction algorithms yield highly improved recognition rates in nonreverberant noisy conditions, while the performance in clean speech is not significantly affected. The algorithms were also evaluated in real-world reverberant conditions with speech-simulating noise and jammer speech.

1pSCb8. A text-to-speech oriented comparison of English and Polish intonation. G. Demenko (A. Mickiewicz Univ., Inst. of Linguist., Miedzychodzka 5,60-371 Poznan, Poland) and W. Jassem (Inst. of Fundamental Technol. Res., Warszawa, Poland)

Current analyses of Polish intonation in terms of $F0$ tracings indicate that it is possible to represent the melodies in the two languages within one descriptive framework, much as in the ''British'' tradition. The basic structure of an intonation phrase can be represented as ([([wPT])+[sPT]) +[NT (+NT)], where NT is the nuclear tune, sPT is the strong (accented) prenuclear tune, and wPT is the weak (unaccented) prenuclear tune. A monosyllabic sPT and a monosyllabic NT are accented, as is the first syllable of a sPT or NT of more than one syllable. The chief structural difference between E and P is that P allows mononuclear intonation phrases only, while E has a special case of a binuclear IP. If rise-fall-rise in NT position can be shown to be an independent NT in E, then this NT does not exist in P. Most of the tunes have quite similar realizations at the phonetic level. The widest differences between E and P with respect to intonation are obtained in their semantic/pragmatic meanings and in the occurrence frequencies. $F0$ traces have confirmed the linguistic structure of an intonation phrase. The importance of dynamic parameters, i.e. those

which describe the relations in the adjacent vowels/syllables describing the set of intonation structures under investigation, was tested by means of neural networks.

1pSCb9. Blind dereverberation using short-time cepstrum frame subtraction. Jean van Eeghem, M. Tohyama (Kogakuin Univ., Inst. of Comput. Acoust. and Hearing, Tokyo, Japan), and T. Koike (NTT Adv. Technol., Tokyo, Japan)

A blind dereverberation method that can be used to separate reverberant speech into an impulse response and a dry speech contribution is presented by the authors. This method is based on the assumption that the contribution of an impulse response to reverberant speech varies slowly compared to that of the dry speech. Processing the reverberant signal for short-time frames and using the special properties of the cepstrum domain, allows a recursive scheme to remove the equal impulse response contributions. Although short-time frames are processed, the effects of an amply longer impulse response can be separated from the dry speech. The assumption of a constant impulse response in only two frames and the iterative processing make the method inherently robust to changes in the impulse response. An ambiguous linear phase addition when calculating the inverse cepstrum constitutes one of the problems to limit the dereverberation performance.

1pSCb10. A robust unit selection system for speech synthesis. Alistair Conkie (AT&T Labs. Res., 180 Park Ave., Florham Park, NJ 07932)

There has been interest for many years in diphone-based speech synthesis and, recently, a rapidly increasing interest in unit selection-based synthesis (as illustrated by interest in the CHATR system). The limits of both systems are well known. While intelligibility is generally very high for diphone-based systems, the resulting signals do not sound completely natural. This happens for several reasons, amongst them the limited number of phone variants present in a typical system, and the cost of concatenating at diphone boundaries. For unit selection synthesis, typically phone-based, it is possible to produce sentences that sound surprisingly natural and intelligible from a large database. However, quality is often not consistent, and the main difficulties appear to be related to selecting acoustically appropriate units from a large database with the correct prosodic characteristics. Typically no prosody modification is done. In an effort to capture the best features of both systems a unit-selection and synthesis algorithm has been devised that allows finer control than the CHATR system (version 0.8), both by applying selective prosody modification and by exercising finer control over the units that get chosen for synthesis. Results of experiments based on this version of unit selection synthesis will be presented.

1pSCb11. Vocal-tract parameter estimation from formant patterns. Heiko Freienstein, Knut Müller, and Hans Werner Strube (Drittes Physikalisches Institut, Univ. of Göttingen, D-37073 Göttingen, Germany)

A method is proposed to estimate individual vocal-tract parameters from formant frequency patterns. Vocal-tract parameters, such as scaling factors of vocal-tract length and area were determined in reference to the given area function of an articulatory synthesizer. These individual parameters of various speakers are expected to be suitable to normalize speakers in automatic speech recognition. It has been shown by Schroeder (1967) that there is a linear relationship between relative formant frequency deviation $\delta\omega/\omega$ and a relative area perturbation $\delta A/A$ of the area function. Further, a similar relationship can be derived for length perturbation of an area function. The presented method is based on the inverse relationship: deviations of the formant frequency pattern of a speaker relative to the

reference speaker are attributed to length and area perturbations of the area function of the reference speaker. A pilot study using synthetic speech showed promising results: estimated vocal-tract area functions were in good agreement with natural area functions. In the present study the method is extended to real speech. Investigations are carried out on speech samples of a German data base (PhonDat) to find speaker-specific parameters. [Work supported by Deutsche Forschungsgemeinschaft (Str 255/7-2).]

1pSCb12. Blueprint of a biomechanical model of vocal tract structures. F. Reiner Wilhelms-Tricarico (Res. Lab. of Electron., 50 Vassar St., Cambridge, MA 02139)

A new biomechanical model of the vocal tract is under construction. Structures of the tract will be represented in a three-dimensional finite-element model. In contrast to a previous pilot project [see Wilhelms–Tricarico, J. Acoust. Soc. Am. **97**, 3085–3098 (1995)], this model will make use of tri-quadratic finite elements and uses implicit time stepping methods. The new model comprises the entire oral floor and the tongue as a system consisting of a small number of finite elements. It will incorporate modeling of collision and sliding between the tongue and the teeth and the hard palate, as well as a moving mandible and hyoid bone. Algorithmic and implementation details will be presented.

1pSCb13. Phonetic recognition of Arabic figures based on formant analysis via artificial neural network. A. I. El Mallawany (Housing & Bldg. Res. Ctr. Dokki, P.O. Box 1770, Cairo, Egypt), M. I. Abdallah (Zagazig Univ., Zagazig, Egypt), M. Abd El Gawad, and H. S. Sedik (Housing & Bldg. Res. Ctr. Dokki, Cairo, Egypt)

This paper describes a speaker recognition for dependent and independent speaker systems based on the formant analysis technique. Also, this system is used for Arabic figure recognition. Linear predictive code (LPC) is applied for extracting the formants of the Arabic digits from 0–9. After training and selecting the proper network structure, the recognition accuracy is found to be 98.8% for the dependent system while it is 78.8% for the independent case. Recognition accuracy for independent speaker is improved to 95.5% when grouping the Arabic digits into three groups.

1pSCb14. On processing speech and non-speech signals in acoustics. Rostislav Pazukhin (Inst. of Modern Philology IFO, Pedagogical College WSP, 13/15 Armii Krajowej Ave., Czenstokhova, PL 42-201, Poland)

In a nonspeech communication, the pitch, duration, formants arrangement, and other acoustic features are *bearers of coded information*. Certain meanings are assigned to them by convention. The processing of nonspeech signals, consists in distinguishing and, subsequently, interpreting them. Speech oral messages are not, by contrast, amenable to such one-average ''technology.'' Each oral utterance betrays a two-layer organization. Here, true information carriers are coded *articulary changes* within the speech channel. Such gestures may be successfully ''read'' directly in TADOMA communication (by the hand placed against the face of the speaker), or in the ''inner speech'' (through bioimpulses announcing rudimentary movements of our speech organs). Mostly, however, the invisible articulations need applying certain ''echo-effects'' to become perceivable. Usually, the ''echoes'' of the voice, noise, whistle, and light (cf. ''lip-reading'') are used. Accordingly, the acoustician's mission in speech processing ought to be reduced to restoring—from the sound characteristics of utterances—the invisible changes within the throat of the speaker (through an improved ''inverse mapping, the directional hearing methodology, etc.''?). Starting from these data, physiologists and linguists could decipher messages using ''lexicon of speech gestures.''

Session 1pSP

Signal Processing in Acoustics and Psychological and Physiological Acoustics: Auditory Displays

Elizabeth M. Wenzel, Cochair
NASA Ames Research Center, MS 262-2, Moffett Field, California 94035-1000, USA

Jens Blauert, Cochair
Communication Acoustics, Ruhr-Universität Bochum, D-44780 Bochum, Germany

Chair's Introduction—1:55

Invited Papers

2:00

1pSP1. Virtual concerts in virtual spaces—in real time. Tapio Lokki, Lauri Savioja, Jarmo Hiipakka, Rami Hanninen (Telecommunications Software and Multimedia Lab., Helsinki Univ. of Technol., P.O. Box 5400, FIN-02015 HUT, Finland, Tapio.Lokki@hut.fi), Ville Pulkki, Riitta Vaananen (Helsinki Univ. of Technol., FIN-02015 HUT, Finland), Jyri Huopaniemi (Speech and Audio Systems Lab., 00045 Nokia Group, Finland), Tommi Ilmonen, and Tapio Takala (Helsinki Univ. of Technol., FIN-02015 HUT, Finland)

The DIVA system is an experimental interactive real-time virtual environment with synchronized sound and animation components. The system provides real-time automatic character animation and visualization, dynamic behavior control of virtual actors, interaction through motion analysis, sound generation with physical models of musical instruments, and three-dimensional sound auralization. The combined effect of 3-D visual and acoustic elements creates stronger immersion than would be possible with either alone. As a demonstration, a virtual band with four artificial musicians has been implemented. The user interacts with the virtual musicians by showing the tempo with a baton, like real conductors do. The animated band follows the gestures of the conductor and another user controls the viewpoint of the audience. Due to the real-time acoustic modeling and sound rendering, both users hear the auralized music in a real concert hall.

2:20

1pSP2. Perceptual criteria for eliminating reflectors and occluders for efficient rendering of environmental sound. William L. Martens and Jens Herder (Univ. of Aizu, Tsuruga, Ikki-machi, Aizu-Wakamatsu City, Fukushima, 965-8580 Japan)

Given limited computational resources available for rendering spatial sound imagery, it is important to determine effective means for choosing what components of the rendering will provide the most audible differences in the results. Rather than begin with an analytic approach that attempts to predict audible differences on the basis of objective parameters, subjective tests were executed to determine the audible difference made by two types of sound obstruction: reflectors and occluders. Single-channel recordings of 90 short speech sounds were made in an anechoic chamber in the presence and absence of these two types of obstructions, and as the angle of those obstructions varied over a 90-deg range. These recordings were reproduced over a single loudspeaker in that anechoic chamber, and listeners were asked to rate how confident they were that the recording of each of these 90 stimuli included an obstruction. The results revealed the conditions under which these obstructions have a significant impact on the perceived spatial image. These confidence ratings were incorporated into an evaluation function used in determining which reflectors and occluders are most important for rendering.

2:40

1pSP3. Auditory displays using loudspeaker reproduction in a car cabin. Winfried Krebber and Hans W. Gierlich (HEAD acoustics GmbH, Ebertstr. 30 a, D-52134 Herzogenrath, Germany, winfried.krebber@head-acoustics.de)

Playback of binaural sounds generated by an auditory display is normally made by headphones. For automotive applications such as driving simulation or sound design, however, loudspeaker reproduction is often required. A setup using a cabin-specific four-channel loudspeaker arrangement is presented and discussed. In order to get a sufficient 3-D sound, the loudspeakers have to be equalized carefully. Best results could be achieved using individualized HRTF sets. The quality of loudspeaker reproduction in comparison to headphone reproduction was evaluated by several psychoacoustic experiments. Localization and distance perception experiments show that loudspeaker reproduction does not achieve the same quality as individualized headphone reproduction. However, for simulator applications the achieved quality is quite sufficient.

1pSP4. Spatial auditory displays for speech communications: A comparison of free-field and virtual acoustic environments.
Todd Nelson (AFRL/HECP, 2255 H St., Wright−Patterson AFB, OH 45433-7022, tnelson@al.wp.afb.mil), Mark Ericson (AFRL/HECB, Wright−Patterson AFB, OH 45433-7901), Robert Bolia (Veridian, Dayton, OH 45440), and Richard McKinley (AFRL/HECB, Wright−Patterson AFB, OH 45433-7901)

The ability of listeners to monitor the simultaneous presentation of multiple speech signals was evaluated in free-field and virtual acoustic environments. Two acoustic environment conditions (free-field and virtual) were combined factorially with two spatial conditions (spatially separated and nonspatially separated), eight simultaneous talker conditions (1, 2, 3, 4, 5, 6, 7, and 8), and two sex of critical speech signal (male and female) to provide 64 experimental conditions. A within-subjects design was used. Participants, four males and four females, were required to detect and identify the presentation of critical speech signals among a background of nonsignal speech events. Speech stimuli consisted of a call sign and a color–number combination contained within a carrier phrase (e.g., ''Ready TIGER go to WHITE ONE now.''). The USAF Air Force Research Laboratory's (AFRL) Auditory Localization Facility—a 277-speaker geodesic sphere housed within an anechoic chamber—was used for free-field presentation, and AFRL's 3-D auditory display generators were employed for virtual presentation. Results indicated that spatial separation of the speech stimuli enhanced performance efficiency of the free field and the virtual conditions. Implications for the design of spatial auditory displays to enhance communication effectiveness and situation awareness are discussed.

1pSP5. Sonification of range information for 3-D space perception.
Evangelos E. Milios, Bill Kapralos, and Sotirios Stergiopoulos (Dept. of Comput. Sci., York Univ., North York M3J 1P3, Canada, eem@cs.yorku.ca)

A device is presented that allows 3-D space perception by sonification of range information obtained via a point laser range sensor. The laser range sensor is worn by the user, who scans space by pointing the laser beam in different directions. The resulting stream of range measurements is then converted to an auditory signal whose frequency or amplitude varies with the range. This device differs from existing navigation aids for the visually impaired. Such devices use sonar ranging whose primary purpose is to detect obstacles for navigation, a task to which sonar is well suited due to its wide beam width. In contrast, the purpose of this device is to allow users to perceive the details of 3-D space that surrounds them, a task to which sonar is ill suited, due to artifacts generated by multiple reflections and to limited range. Preliminary trials demonstrate that the user is able to detect corners and depth discontinuities accurately with ease and to perceive the size of the surrounding space.

1pSP6. Auditory perception of rolling balls.
Mark Houben, Luuk Franssen, Dik Hermes (IPO, Ctr. for Res. on User-System Interaction, P.O. Box 513, NL 5600 MB Eindhoven, The Netherlands, houben@ipo.tue.nl), Armin Kohlrausch, and Berry Eggen (IPO and Philips Res. Labs., Eindhoven, The Netherlands)

Two experiments investigating the perception of recorded sounds of rolling wooden balls are reported. In the first experiment, whether subjects can identify differences in the size of rolling wooden balls, is studied. In the second experiment, the velocity of rolling balls is varied. Real recordings of wooden balls rolling over a wooden plate were presented pairwise, with a 700-ms pause in between. The stimuli had a duration of 800 ms and were presented at equal SPL. Subjects had to decide which of the two sound examples was created by the larger (first experiment) or faster ball (second experiment) (2I2AFC procedure). The results of the first experiment show that subjects are able to identify differences in the size of rolling balls, except for the stimulus pair with the smallest relative difference in diameter of 14%. The second experiment reveals that most subjects can clearly discriminate between rolling balls with different velocities, if the relative difference exceeds about 30%. However, some subjects had difficulties in labeling the sounds correctly, resulting in percentage correct responses close to 0%. It is to be expected that labeling errors will disappear when subjects receive feedback about the correctness of their responses.

Contributed Papers

1pSP7. A filtering model for efficient rendering of the spatial image of an occluded virtual sound source.
William L. Martens, Jens Herder, and Yoshiki Shiba (Univ. of Aizu, Tsuruga, Ikki-Machi, Aizu-Wakamatsu City, Fukushima, 965-8580 Japan)

Rendering realistic spatial sound imagery for complex virtual environments must take into account the effects of obstructions such as reflectors and occluders. It is relatively well understood how to calculate the acoustical consequence that would be observed at a given observation point when an acoustically opaque object occludes a sound source. But the interference patterns generated by occluders of various geometries and orientations relative to the virtual source and receiver are computationally intense if accurate results are required. In many applications, however, it is sufficient to create a spatial image that is recognizable by the human listener as the sound of an occluded source. In the interest of improving audio rendering efficiency, a simplified filtering model was developed and its audio output submitted to psychophysical evaluation. Two perceptually salient components of occluder acoustics were identified that could be directly related to the geometry and orientation of a simple occluder. Actual occluder impulse responses measured in an anechoic chamber resembled the responses of a model incorporating only a variable duration delay line and a low-pass filter with variable cutoff frequency.

1pSP8. Communication and 3-D sound: Speech intelligibility and speaker recognition.
Rob Drullman and Adelbert W. Bronkhorst (TNO Human Factors Res. Inst., Kampweg 5, 3769 DE Soesterberg, The Netherlands)

In a 3-D auditory display, sounds are presented over headphones in a way that they seem to originate from virtual sources in a space around the listener. The possible merits of such a display were investigated with respect to speech intelligibility and speaker recognition against a background of competing speech. Various conditions were investigated: speech material (words or sentences), presentation mode (monaural, binaural, or

3D), number of competing speakers (1–4), and virtual position of the speakers (in 45° steps around the frontal horizontal plane). Average results for 12 listeners show an increase of speech intelligibility for a 3-D presentation, with more than two competing speakers compared to conventional monaural or binaural presentation. The acuity to recognize a speaker is slightly better and the time required for recognition is significantly shorter for a 3-D presentation in the presence of two or three competing speakers. Although absolute localizability of a speaker is rather poor, spatial separation appears to have a significant effect on communication. For either speech intelligibility, speaker recognition, or localizability, no difference is found between the use of an individualized 3-D auditory display and a general display. [Work supported by the Royal Netherlands Navy.]

5:00

1pSP9. Simplified auralization of reflections in a virtual auditory environment. Bernd Dürrer (Inst. of Commun. Acoust., Ruhr Univ. Bochum, Geb. IC 1/132, 44780 Bochum, Germany, duerrer@ika.ruhr-uni-bochum.de)

Based on the mirror-image method, a system for generating virtual auditory environments can simulate reflections using virtual sound sources. During the auralization process, the signal of each virtual sound source has to be convolved with the appropriate pair of HRTFs. In this investigation, a simplified model based only on interaural time and intensity differences was used for the auralization of higher-order reflections: This approach greatly reduces the necessary computational effort. Stimuli using the simplified model were compared in listening tests with stimuli using convolution with HRTFs. The tests were performed with artificial-head HRTFs (17 subjects) and individual HRTFs (9 subjects). In addition to the reduction of computational costs, significant perceptual effects on the extent of the auditory event were noted which are believed to be beneficial for certain applications (e.g., telecommunications). The usability of this model was evaluated in a speech intelligibility test in a concurrent speakers scenario. [Work supported by the EU project AUDIS (Esprit No. 22352).]

5:20

1pSP10. Design criteria for auditory virtual environments. Jörg Sahrhage (Inst. of Commun. Acoust., Ruhr Universität Bochum, 44780 Bochum, Germany, sahrha@ika.ruhr-uni-bochum.de)

During the last decade a large variety of different systems and technologies for generating auditory virtual environments emerged, ranging from low-cost displays for computer games to high-end interactive and immersive systems comprising real-time head tracking and more or less authentic room simulation. Auditory virtual environment generators employing different technological approaches have been developed for various application areas like navigation aids, teleconferencing systems, virtual control rooms, integrated multi-modal VE generators, or even as tools for psychophysical research. Depending on the particular application, perceptually relevant parameters (e.g., responsiveness, smoothness, and authenticity) will be identified on the basis of psychophysical knowledge. Further, an attempt will be made to derive and quantify adequate physical design criteria. An overview of the available auditory VE generators will be followed by a careful analysis and comparison of the system's alleged capabilities and technological constraints. These will be checked against the proposed application-dependent design criteria.

5:40

1pSP11. A sound judgment depending on the urban visual setting? Stephanie Viollon and Catherine Lavandier (Univ. de Cergy, IUT Dept. Genie Civil, Rue d'Eragny, Neuville sur Oise, 95031 Cergy Pontoise Cedex, France, viollon@u-cergy.fr)

In a number of experiments carried out in the experimental psychology field, audition and vision proved to provide information which interacted each other. What about the audio–visual interactions in the city? The first aim of this research was to develop an experimental procedure involving methodological parameters best suitable to test the influence of the visual setting on the sound judgment in the complex sound environment. The second one was to apply it to various urban situations. In a series of simulation tests, participants rated some urban sound stimuli and this, under different visual conditions varying according to the degree of urbanization. An artificial audio–visual environment was specifically created. The experimental methods were based on three types, involving different structures of presentation of the audio–visual combinations. For each of them (except for the completely random one), two types of orders of presentation of both the visual and sound sequences were defined. The results for all the experiments were commented on and compared. The best suitable experimental method was validated. The present results point out that the visual influence was many sided. Some clusters of sound stimuli and sound variables were defined according to the visual influence exerted.

6:00–6:20

Discussion and demonstration sign-up

Attendees may ask questions of session authors and may sign up to attend a demonstration of DIVA (1pSP1), or to present their own sound samples, in a high quality studio environment. Nearly all sound formats up to 12 channels can be reproduced for listener groups of up to 20 persons. The studio is located nearby.

Session 1pUW

Underwater Acoustics: Wave Propagation: Session in Honor of Leonid M. Brekhovskikh I

Peter N. Mikhalevsky, Cochair
SAIC, 1710 Goodridge Drive, McLean, Virginia 22102, USA

Nikolai Dubrovsky, Cochair
N.N. Andreyev Acoustics Institute, 4 Shvernika Street, 117036 Moscow, Russia

Invited Papers

2:00

1pUW1. Acoustic thermometry of ocean climate (ATOC). Walter Munk (Scripps Inst. of Oceanogr., Univ. of California, San Diego, 9500 Gilman Dr., La Jolla, CA 92093) and ATOC consortium

The purpose of ATOC is to monitor ocean variability over scales exceeding 1 year and 5000 km, using the method of ocean acoustic tomography. ATOC takes advantage of the favorable condition in the sound channel for low-frequency propagation. By forming long-range averages, the very intense mesoscale variance is reduced by two orders of magnitude to permit the detection of the relatively weak climate signals. Results of 15 months of transmissions from a source off Monterey, California will be shown. The exploration of the oceans by acoustics dates back to the use of acoustic depth meters and the navigation of drifting buoys. With the modern development led by Brekhovskikh, the move is toward matched-field methods which attempt to interpret the entire recorded sound field in terms of the pertinent ocean parameters.

2:40

1pUW2. Leonid Brekhovskikh: The man and the scientist. Nikolai A. Dubrovsky (Andreyev Acoust. Inst., #4 Shvernika Ul., Moscow 117036, Russia, dubrov@akin.ru)

L. Brekhovskikh was one of the main founders and a first Director of the Acoustics Institute (AI) of the USSR Academy of Sciences (1954). He contributed a lot in the selection of young gifted scientists and engineers as AI staff, formation of the main branches of research in AI, construction of new buildings and experimental setups, and promoted a new approach to design and construction of the research vessels *Sergei Vavilov* and *Petr Lebedev*. The active creative work of L. Brekhovskikh in AI (1963–1980) became even more intensive after he left the director's position. It relates to new theoretical achievements, fruitful and numerous experiments in the ocean, organizing the famous Zvenigorod conferences on ocean acoustics, and publishing the fundamental treatise ''Acoustics of the Ocean.'' Brekhovskikh possesses the personal features that facilitate his achievements in science. He is a very gifted, amicable, benevolent, reserved, purposeful, concentrated, and rational person. Sport exercises and yoga practice have considerably supported his creative efforts for decades.

3:20

1pUW3. Theory of sound propagation in the ocean: A tribute to L. M. Brekhovskikh. Finn B. Jensen (SACLANTCEN, Viale S. Bartolomeo 400, 19138 La Spezia, Italy, jensen@saclantc.nato.int)

This presentation will be an account (from a personal perspective) of the impact L. M. Brekhovskikh's first (and now classic) book on *Waves in Layered Media* from 1960 has had on the development of modern-day numerical models for solving ocean acoustic problems. A whole generation of acoustic modelers, starting in the early 1970s, devoted a career to developing computer codes that would predict sound propagation through a realistic ocean environment, including refraction, scattering, and diffraction of sound within the water column itself, as well as reflection and scattering at ocean boundaries (sea surface and seabed). Not everybody realized that much of the underlying theory had been developed years before and presented in the monograph by Brekhovskikh in 1960 (English edition). In fact, much of the numerical work undertaken in the 1970s and 1980s was merely an automation—through the use of a computer—of solution procedures already outlined by Brekhovskikh. Few, if any, have contributed more to establishing a sound theoretical foundation for ocean acoustics, and L. M. Brekhovskikh's influence on the field can hardly be overestimated.

1pUW4. Development of underwater acoustics in China and L. M. Brekhovskikh's early activities in China. Dinghua Guan (Inst. of Acoust., Chinese Acad. of Sci., Beijing 100 080, PROC, zrh@canna.ioa.ac.cn)

In this paper the early activities and influence of L. M. Brekhovskikh in the development of underwater acoustics in China are reviewed. The activity of Brekhovskikh as an invited underwater acoustics expert in the formulation of the 12-year plan of S&T development of China in 1956 is described. This paper reviews the early development of underwater acoustics in China, including the visit of Chinese experts to the Institute of Acoustics in Russia, the first joint Sino–Russian experiment in the South China Sea, and the training of young Chinese scientists. The monograph ''Waves in Layered Media'' has had a large influence on Chinese underwater acoustics. This paper also reviews part of the research on underwater acoustics in China. The ray-mode theory was developed, improved, and used in propagation, reverberation, and noise field modeling in shallow water. The relationship between rays and normal modes was analyzed and proved. The influence of sea bottom in shallow water was expressed in terms of angular dependence of bottom reflection loss. Wave theory for long-range reverberation was developed. MFP, MMP, and other methods of positioning and inversion was developed. Internal waves and their influence on sound propagation were studied.

4:00–4:20   Break

4:20

1pUW5. The tangent plane approximation and related approaches in rough surface scattering theory. Alexander G. Voronovich (NOAA/ Environ. Technol. Lab., 325 Broadway, Boulder, CO 80303, agv@etl.noaa.gov)

The tangent plane approximation (TPA) was introduced into the theory of wave scattering from rough surfaces by L. M. Brekhovskikh in 1952. This method corresponds to the semi-classical (WKB) approximation and is the second ''classical'' method (after the small perturbation method introduced into this realm by Rayleigh in 1907) that is widely used for the solution of wave propagation problems. Development of this approach significantly increased the capabilities to solve practical tasks and produced a long-lasting effect on other theoretical developments. For example, the operator expansion method or the small-slope approximation developed recently could be considered as a generalization of the TPA. The first part of the talk will be devoted to a demonstration of the relationship between TPA and different ''nonclassical'' approaches developed later. In many but not all instances, TPA could be obtained as a zeroth-order iteration of an appropriate integral equation for surface sources (for this reason TPA is also called the Kirchhoff approximation). However, some other approaches to the approximate solution of this integral equation based on smoothness of the rough surface are also possible. Corresponding enhancements of the TPA will be considered also for the case of impedance surface.

4:40

1pUW6. Leonid M. Brekhovskikh and his scientific school. Yury P. Lysanov (Andreyev Acoust. Inst., 4 Shvernik Str., Moscow, 117036 Russia, bvp@asu.acoins.msk.su)

The following items will be discussed: The first meeting with L. M. Brekhovskikh (1951), the Acoustics Laboratory at the Lebedev Physics Institute, L. M. Brekhovskikh and the organization of the Acoustics Institute, a collection of young scientists and post-graduates from many Russian cities for the new Institute, a post-graduate studentship, new problems in ocean acoustics, wave scattering by rough surfaces, Brekhovskikh's approximation (the tangent plane method), the growth of international relations, the Second and Fourth International Congresses on Acoustics (USA, 1956; FRG, 1959), all-union Acoustical Conference in Moscow (1958), the great role of L. M. Brekhovskikh in educating young scientists in ocean acoustics and oceanology (lectures at the Moscow State University and the Moscow Institute of Physics and Technology), scientific seminars under the supervision of L. M. Brekhovskikh (Sukhumi, Zvenigorid, Moscow), and Brekhovskikh's books and papers and their fundamental contribution to ocean acoustics and oceanology.

5:00

1pUW7. Applications of the waveguide invariant approach. William A. Kuperman, Gerald L. D'Spain, Hee Chun Song, and Aaron M. Thode (Marine Physical Lab., Scripps Inst. of Oceanogr., La Jolla, CA 92093-0704)

The waveguide invariant approach [S. D. Chuprov, *Acoustics of the Ocean: Current Status*, edited by Brekhovskikh and Andreevoi (Nauka, Moscow, 1982), pp. 71–91; L. M. Brekhovskikh and Y. P. Lysanov, *Fundamentals of Ocean Acoustics*, 2nd ed. (Springer-Verlag, Berlin, 1991), pp. 139–145] summarizes the dependence of robust interference phenomenon on the ocean environment by a parameter denoted ''β.'' This elegant approach is based on the dependence of group speed on phase speed for general classes of environments. It has been used to describe interference patterns in the range-frequency plane for range-independent and two-dimensional, mildly range-dependent environments. The approach is extended to include the third dimension, azimuth dependence. Furthermore, an analogous derivation for describing the ambiguity structure of a broadband linear matched-field processor reveals that the trajectories of sidelobes of such a processor in the range-frequency plane are also subject to the same invariant description. A connected result is that the focus of a phase-conjugate array can be shifted away from or toward the original probe source by a process directly related to the waveguide invariant formalism. These new applications have all been experimentally verified.

1pUW8. Wave diffraction at reflection and propagation in layered media. Oleg A. Godin (School of Earth and Ocean Sci., Univ. of Victoria, P.O. Box 3055, Victoria, BC V8W 3P6, Canada, ogodin@uvic.ca)

In his theory of waves in layered media, L. M. Brekhovskikh has systematically employed integral representations of the field in terms of plane or quasi-plane waves, with all the other useful representations of the field derived from this exact solution. It is this approach that enabled him to efficiently study, and for the first time gain a deep insight into and a quantitative description of the fundamental diffraction phenomena accompanying total reflection, guided propagation, and formation of caustics in inhomogeneous media. In this presentation, L. M. Brekhovskikh's work on wave diffraction in layered media, including his concept of diffracted rays, study of ray and beam displacement at reflection, investigation of ray theory and the WKBJ approximation domains of validity, and an exact solution for a range-dependent benchmark problem, are reviewed. Recent developments in the theory of acoustic propagation in inhomogeneous media are traced back to the ideas first introduced and explored by L. M. Brekhovskikh.

1pUW9. My life, my teachers, pupils, and friends. Leonid M. Brekhovskikh (Acoust. Wave Propagation Lab., P. P. Shirshov Oceanology Inst. of the Russian Acad. of Sci., 36 Nakhimovsky Av., Moscow, 117851 Russia, leon@rav.sio.rssi.ru)

The presentation will cover the following topics: Childhood dreams: Have they become a reality? Why and how did I become an acoustician? Sixty years of creativity in the golden times of Russian science, including oceanography and ocean acoustics. Contact with which outstanding persons did influence my world outlook? What could I pass on to my pupils? What does the ''iron mode of life'' mean? What is the short- and long-term future of Russian science from my point of view?